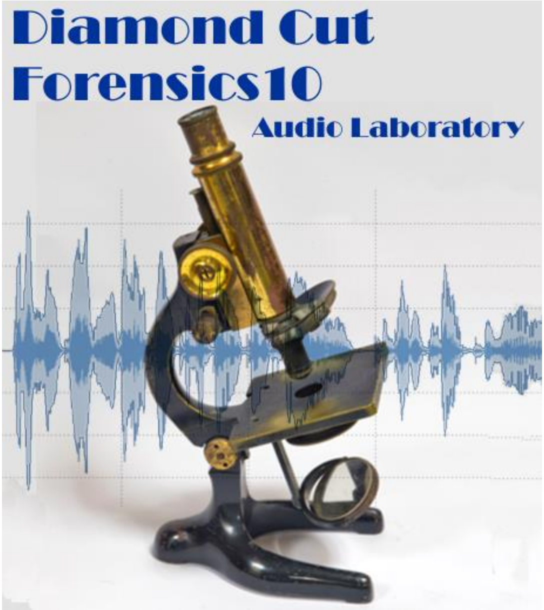

&
Diamond Cut DCart10.8x

User’s Manual

For

Diamond Cut Productions, Inc.

DCForensics10.8x Audio Laboratory

&

DCArt10.8x

Sound Restoration for Sound Projects

Made in the U.S.A.

DC Forensics Audio Laboratory v. 10.8x
&
Diamond Cut Audio Restoration Tools (DCArt) v. 10.8x

User's Manual **25th Edition**

DC Forensics Audio Laboratory v. 10.8x & DCArt v. 10.8x

DC Forensics10 Audio Laboratory, Version 10.8x and above but less than Version 11.0
DCArt10, Version 10.8x and above but less than Version 11.0

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In Fond Memory of Les Paul for his Inspiration and Friendship

Contents

Installing and/or Upgrading Information

Here is some important information to be aware of when installing your new Diamond Cut version 10.xx software (sometimes referred to as DC-Art, version 10). This applies to both the DCArt10.xx and the Forensics10.xx software programs. Be sure to un-install previous versions that you have if they are 10.5x, 10.6x before installing the new version to avoid clashes; your registration information will be maintained when updating to version 10.7.

Please allow 24 hours to receive your serial number after your purchase; we use a manual process to create those for security reasons.

Similarly, allow 24 hours to apply for registration using your serial number for your registration number to show up in your email. Again, this registration process involves manual operations by humans to avoid security issues.

Meanwhile, your software will fully function operable for 15 days after you download it from our website, allowing plenty of time for the registration process to take place.

If you are upgrading from an earlier version of a Diamond Cut Audio Restoration product such as DC7 or DC8, (or an earlier version of DC Forensics) there are some things to be particularly aware of.

1. If you've purchased a hard copy version of the product, you will find your serial number located behind the install disc which is contained in a pouch located at the back of the printed user's manual. Hard copies are available on thumb drives by special request only for an additional charge. Otherwise, your serial number will have been e-mailed to you after an electronic purchase online. It is a 25 figure code (5 groups of 5 characters). Please do not key-stroke this number into your system. Copy and paste it in to avoid registry error that are difficult to correct.

2. If you have purchased a new installation (not at upgrade), you will need to enter a valid email address* (up to 256 characters maximum), and a serial number. After that, the software will fetch your registration code via your internet connection or via email. If you do not have an internet connection, you can call us or write to us to obtain

a registration code. We will need the exact email address that you used when you installed the software and your product serial number. If either of these are not provided exactly as entered into the software, the system will not register and, will not work after the time trial period has expired. The Serial Number will have 25 figures. The registration code will have 12 figures.

3. If you are purchasing, an upgrade, the Diamond Cut Merchant system will look for a previous registration in terms of your User Name, Serial Number and/or email address during the store check-out process. The server system looks for existing DC6, DC7, DC8, or DC10 registrations. Earlier versions may not be recognized. Forensics10 upgrades will look for earlier DC Forensics versions in your account. If it finds one, it will extract the required information and use it for the electronic registration of your upgraded product which should occur within 24 hours. The registration code should not be keystroked into your system; please copy and paste it into the registration field of the software to avoid errors.

4. After product installation has been completed, please check for software updates occasionally at www.diamondcut.com because we are in a continual process of creating product improvements and bug fixes. These updates are free to registered Diamond Cut customers.

5. Every so often, it is a good idea to check for additional product upgrades to keep your software current in terms of bug fixes and feature enhancements. Look under the Help menu for details pertaining to your software update status and version number.

6. You do not need to un-install DC7 or DC8 before installing DCArt10. The same applies to an upgrade from an earlier version of DC Forensics to DC Forensics Audio Laboratory, v.10.x. As a matter of fact, it may be of advantage to leave your earlier version of DC-Art (Diamond Cut Audio Restoration Tools) installed if they contain important presets that you have created over time or if you want to maintain your existing DC Tune Database and playlists.

7. However, you do need to un-install previous versions within a product family before updating. For example, if you are updating your software from DCArt10, version 10.x to version 10.y, we recommend

un-installing v. 10.x first. Upgrades from any version 10x product to 10.7x require that the earlier version must be uninstalled first.

8. If you have personal favorite presets that you had created using an earlier version of DC-Art (like DC8.5), you can use the Preset Manager (found under the Edit Menu) in DCArt10 to import them into the new preset directory created by DCArt10 as long as the earlier version is still installed.

9. The first time that you open your new DCArt10 DC Tune Library, the system will ask you if you want to import your old DC Tune Library Database from an earlier directory into your DCArt10 directory. If you choose to do this, it will also transfer your old playlists into the DCArt10 Directory.

10. It is a good idea to record your User Name, Serial Number and Registration code for safe keeping in case your hard drive fails. One good place to record this is in the “Notes” section located at the back of the printed users guide. You will need that information to restore your system in the event of a system failure. To locate that information, go to your Diamond Cut Help\About DCArt --- menu. All of the salient Diamond Cut Software registration information can be found there including your software revision number. The most common support call that we receive occurs after a hard drive crash. That question is “what was my User Name, Serial Number and Registration Code”. Due to security concerns, we have to research the request and then go into our archive to find that information. This process delays the user’s restoration of their computer system. Thus, the delay can be avoided by recording that information (or printing a screen shot) immediately after your initial Diamond Cut Software installation. With that information at hand, you can re-install your Diamond Cut software without contacting the company (you can find the software program on our website at www.diamondcut.com). Of course, if that information has been lost, feel free to contact us, but give us 24 hours to do the necessary research and to get back to you with your Diamond Cut license information. The product serial number, user name, registration code and e-mail address are not kept on the Diamond Cut Productions servers indefinitely. It is the customer’s responsibility to preserve that information in their own archive.

Section 1 - Welcome & the Product Basics

Thank you for purchasing another fine product from Diamond Cut Productions, Inc. We strive to provide you with audio DSP software that pushes the envelope in terms of features, but leave enough money in your wallet to afford the envelope. We will do our best to not only provide you with the most effective DSP audio software available anywhere, but also complement it with excellent support materials. This manual is your window into the product. Please use it. A great supplemental source of information can be found at the Diamond Cut Productions forum located at:

http://www.diamondcut.com/vforum/

It is a free service and has a large body of information (roughly 28,500 posts coupled with a search engine) built up over a long period of time by users of the various Diamond Cut products. Though we will support this product to the best of our ability via telephone, email, the forum and our web site, chances are, everything that you need to navigate DCArt10 and Forensics10 are right here. We've done our best to make this manual and the other supporting documents as user friendly and intuitive as possible…give it a try… "Reading is fundamental to understanding the product".

History of the Product

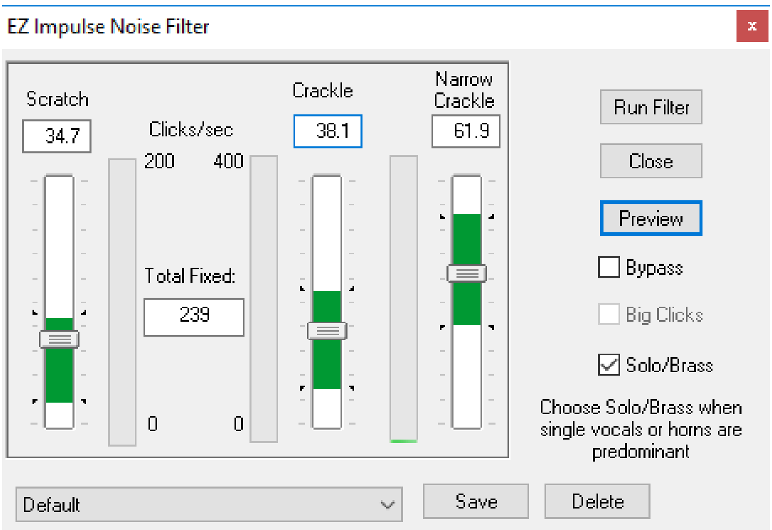
Diamond Cut Audio Restoration Software was originally written by two engineers in their spare time to facilitate the very specific needs that arose in their restoration of the Edison Lateral Collection of Test Pressing Recordings, which is located at the Edison National Historic Site in West Orange, New Jersey. Rick Carlson and Craig Maier developed and improved this program over a 25 year period of time. When they made it available to the general public many years ago, it was with the idea in mind that if it solved some audio restoration problems for them in dealing with the Edison archive, it might also be of use to others confronted with similar audio signal processing problems. Since then, it has grown into the most used audio restoration, enhancement and forensics audio authentication product on the planet, and beyond.

How to Get Started
This manual is nice and thick but it's not a novel and we don't expect you to read it cover to cover. However, we'd like you to get started by carefully reading Section 1. It covers how to install the software, hook up your external sound system, and test your installation. Section 1 also gives you some great hands-on tutorials that will take you step-by-step through some real restorations. If you go through that section of the manual, you'll be fully oriented and ready to use the program! The remaining sections are great reference materials should you run into any questions for which you need more detailed answers. Additional tutorials can be found in Section 3.

Important Note: This manual supports two different products. As many of the features for both DCArt10.8x and Forensics Audio Laboratory v.10.8x are similar, we have streamlined this manual to support both products. As you browse the Table of Contents, you'll notice numerous entries marked with the sub-title of "*Forensics Only*". These features are only supported in the DC Forensics Audio Laboratory v. 10.8x.

About the Product

DCArt10.8x and DC Forensics Audio Laboratory v.10.8x contain a comprehensive set of tools designed for audio restoration, enhancement audio archiving, audio signal analysis and audio surveillance. This extensive toolkit will allow the user to remove extraneous noise and also enhance the sound from any audio source without degrading the content contained on the original. Recognizing that there is a tradeoff between the degree of noise removed from a source and the fidelity, transient and frequency response maintained, we have sought to provide the highest level of user control while maintaining ease of use over the variables that affect the audio restoration and enhancement process. Thus, the slider controls have a very wide range of adjustability in order to cover the needs of any audio restoration or enhancement situation. Special "Green Zones" are provided on many of the slider which identify the range (or span) of the slider controls which have the highest probability of producing nominally good results. Some filters or effects have multiple modes of operation. In some cases, the Green Zones will change depending on your chosen mode of operation. However, there will be situations in which you will find yourself operating these controls beyond those zones. Look at it as a manifestation of the 80 – 20 rule. 80% of the time, you will find your desired setting inside the Green Zone, but 20% of the time, it may fall below or above that zone. Operating outside of the Green Zones is okay if you are satisfied with the results produced. So, the Green Zones are a great place to start with when tackling a project using any particular filter or effect. Another place to look for a good starting point are within the descriptive factory presets presented for each Diamond Cut Productions filter or effect.



A Typical Filter showing "Green Zones" for Slider Controls

Professional Audio Engineers, Radio and Television Engineers, Forensic Scientists, Audio Archivists, Ham Radio operators, Recording Studios and Audiophile hobbyists use our products for re-mastering, editing, noise removal, signal enhancement and audio signal analysis and/or synthesis. It is also used by many local, state, and federal government agencies including both military intelligence operations and forensics law enforcement applications, particularly in surveillance situations. Private sector forensics audio engineers use it for post processing and authentication of forensic recordings of phone calls and surveillance tapes. Radio and television networks have found the LIVE (real-time signal processing) feature of particular value in their real-time applications. Additionally, a number of Engineering Colleges and Universities use this software as part of the laboratory portion of some coursework pertaining to applied DSP techniques and Audio Forensics Science.

Important Note: Most of the algorithms used in x and DC Forensics Audio Laboratory v10.8x use double precision floating point DCArt10.8 internal math as opposed to fixed precision integer math in order to minimize the possibility of introducing digital noise or distortion into your .wav files. The tradeoff associated with using this method is the time required to process a file being slightly longer than the fixed precision integer method.

About the Version

DCArt10 (v.10.8x) and DC Forensics Audio Laboratory (v.10.8x) have been in the works for about a year. We've added a ton of novel new features. If you've been using our products and are only new to DCArt10.8x or DC Forensics Audio Laboratory v.10.8x, use the list below as your learning guide. It includes every new feature to the product so that you can quickly access all of the newest info without rehashing the stuff you've already read. Here is a list of the features making their debut in Version(s) 10.8x:

DCArt10 & DC Forensics v. 10.7x New Features:

- 32 New Paragraphic EQ Factory Presets Added
- 16 Multifilter Presets Added
- 37 Assorted presets added to other filters
- 5 Echo / Reverberation Reduction Presets added to the Dynamics Processor (Expander Mode)
- Magnetic Tape Recorder Simulator Preset added to the Multifilter
- Channel Balance Meter added to the Virtual Phono Preamp (VPP)
- 'Silk' mode added to the Virtual Valve Amplifier
- Sharp Cutoff Mode added to the Overtone Synthesizer

- 7 random noise generators added directly to the Make Waves Generator
- Make Waves Generator now produces it's signals in real-time via Preview function.
- AAC (Advanced Audio Coding) compressed audio file support added for file extensions such as .aac, .m4a, etc.
- Added docking windows for History Dialog box, VU Meter and File Information dialog boxes
- Added File Viewer Docking window to access file system more readily
- Spectrogram has new "Fast Tracking" with the Time display when zooming in and out of a sector of the file.
- Added Hr. (Hour) indicator to record time display
- Changed the software application "look" to follow various flavors of Windows.
- Channel Phase vs. Time Display added to Forensics Version
- Comparative Histogram allows comparison of statistical distribution of two file portions with each other.
- Customizable Application Look.
- Customizable Toolbar(s) facility by users.
- Customizable Keyboard Accelerators.
- Direct Access to the Adaptive Frequency Domain Filter (AFDF) provided.
- File Statistics Feature (calculated on highlighted area of a file)
- Forensics Audio Whisper Enhancer Filter added (including 15 factory presets)
- Forensics Auto "Remove Silence" function for easier transcription of long surveillance files.
- Forensics Histogram vs Time display added for analyzing entire files in a histogram domain.
- Green Zones (Sweet Spots) added to most slider controls
- Improved the audio quality of the Stretch and Squish (pitch and tempo change) system.
- Mono (L-R) for vertical cuts added to Virtual VPP
- Paste Interpolate Icon button for easy waveform interpolation when working with tablet computers.
- Large Icons user selectable option.
- Low Frequency Shelf option added to the Paragraphic EQ.
- One-Click Switch between Fast & Classic Edit Mode.

- Over 2,000 Descriptive Factory Presets are now provided for ease in "getting started" with any given function.
- Sampling Rate support extended from 192 kHz up to as high as 210 kHz providing up to 100 kHz Bandwidth capability.
- Set New Start Time for File feature added.
- Spectral Frequency Tracking to aid in Audio Authentication
- Subsonic Explorer feature to help find sub-audible events
- Tasks Pane feature guides you to the correct filter
- Updated GUI Appearance
- VST Hosting (VST Plug-in Support) Added
- Waveform Overview Added
- Wind Noise Filter Added
- Wizard added (Record Restoration)
- Interactive Update Monitor under the Help Menu
- High Probability "Green Zones" added to many slider controls
- 1 Years of Free Product Support commencing on the date of purchase - (longer contracts are available at our website)
- Dynamic Bass Processor (Dyna Bass)
- Stout Bass added to the Dynamic Bass Processor
- CNF Presets completely re-created
- Harmonic Reject Filter Presets completely re-created
- Record Restoration Wizard added
- Improved Voice ID Formants discrimination
- Expanded Video-Audio Extraction Capability added
- Improved the performance of the Sub-Harmonic Synthesizer via a new selection box called "Sharp Cutoff".

About the Manual

With more than 700 pages of information passed on to us by our scientists and engineers, we tried to format a manual that was basic enough for a first time, non-experienced user, but filled with the kind of detail and reference material that any "propeller head" would love. We've also condensed a good deal of the material…all that means to you is that if you have an interest in a specific item such as the Punch and Crunch Filter, you simply look it up in the Table of Contents or Index. You'll not only find all the information describing the tool, but also any tutorial information on that specific tool all in the same location. Our Tutorial section now contains only multiple tool procedures. As mentioned earlier, this manual supports both DCArt10 and DC Forensics10, so be aware that features native only to the Forensics version will be marked accordingly. Information contained in the product's help file will sometimes supersede information contained in the printed User's Guide (or the user printable .pdf), because the software (and thus its help file) is updated more frequently than the printed material.

Overview

The Diamond Cut Software comes in two versions. The DCForensics version (a professional version) is the superset platform of our audio software product family. The DCArt (a commercial version) is a subset of the DCForensics platform. You can find 15 day, fully functional demo versions of these products at the following links:

Forensics Version (DCForensics10.8x):
https://www.diamondcut.com/st3/product/dcforensics10/

Commercial Version (DCArt10.8x):
https://www.diamondcut.com/st3/product/dcart10/

The demo links are shown in Red typeset within the product description area. These demo version can be turned into permanent versions if you decide that you like them by purchasing serial numbers for the desired product on those same websites.

If you learn the basic concepts of the operation of either of these two programs, you will be in good shape to operate either one on a basic level. This "Getting Started Guide" presented here is intended to teach the basic program/process/flow and principles of operation that you will be using in your restoration/enhancement/analysis work.

This Getting Started Guide document does not replace the complete Diamond Cut Software (700 page) user's guide which can be found under the Diamond Cut Productions program group of your product.

Note: Everything you need to know to install the software, hook up your hardware and begin to use the program will be found in this "getting started" document. This is ***must*** reading. We think that you'll find it easy and fun!

In this guide, we'll tell you everything you need to know in order to install, configure, check out and start using the software. Read through this section carefully and perform the steps indicated and you'll have a perfectly working real time audio restoration workstation!

Step One – Make Sure System Meets Our Requirements

Most machines today are more than capable of handling extensive audio signal processing techniques. Hard drives have also grown to a point, where you really have no limits on what you can do…except perhaps time. As we've not yet developed time manipulation outside of the computer, let's just deal with what you'll need in the way of a PC. Here are the minimum System Requirements:

- Intel Pentium 4 or better (previewing large numbers of filters strung together in the Multi-Filter may require higher power processors.) Faster CPU clock rates are better. Core i5 or better is recommended.
- 16 bit (or better) Stereo Sound Card with line level inputs. Real time feed through requires a full duplex sound card that can record and play at the same time.* Note: Sound card must use Windows DirectX or WDM drivers.
- 1024 Mbytes of memory (RAM) for XP, 2GB for Vista or Windows 7, Windows 8**, or Windows 10.
- Windows XP with SP3 upgrade, *** Windows Vista, Windows 7 or Windows 8 (32 or 64 bit), and Windows 10 (Windows 10 is the preferred Operating System for this software)
- Audio Source Material
- A high quality Audio Delivery System (to interface whatever source recordings you have into your computer system)
- A Hard Drive with enough space to accommodate your .wav files. A formula is provided to calculate the space

requirement but you should have about 2.5 Giga bytes free to make a full audio CD. (The program itself only requires about 45 to 70 Mbytes of space, depending on the version.)

- Mouse, Keyboard, and Color Monitor
- A High Quality Set of Headphones (required for Forensics Audio work)

* You now possess the state-of-the-art in technology for audio restoration, enhancement, audio measurement and forensics file surveillance and authentication. If you haven't yet replaced the sound card that came with your computer (which is probably the lowest quality audio component you own), consider purchasing a high-performance unit. Your audio restoration results will only be as good as the weakest link in your audio chain allows. Maybe that's a good item to add to your "wish list".
** If your computer has more than the minimum required to run your OS adequately, further increases in the quantity of RAM will not appreciably speed up the DCArt10 / DC Forensics10 algorithms since they are almost totally limited by the processor. Upgrading to a faster processor <u>will</u> allow the filters to run faster.
Most DCArt10/DC Forensics10 algorithms use a single thread and will not be appreciably sped up by multi-core or multiprocessor CPUs.

***This product uses DirectX and requires DirectX 9.0 or higher.
Some older computers may require additional component installations.

Step Two - Installation of the Product

Most users find this extremely easy.

1. Download the demo software program installer from our website https://www.diamondcut.com/st3/product-category/software/
2. Install the program. After the demo time period runs out, you can choose to purchase the software. The demo version will convert to a fully registered license if you decide to make the purchase at the end to the demo period. Each license is for one user on up to two computers.

The software will be installed in a folder under “programs” called *Diamond Cut Productions*. Several demo .wav and/or .mp3 files are also supplied as part of the demonstration tutorials found in both the Help file and PDF version of the manual.
During the installation process, you will be asked to approve the license agreement, and select an install location. After a few seconds of installation time, you’re ready to launch the program by clicking on the DCArt10 or Forensics10 Icons, depending on which program you have purchased.

When DCArt10 or Forensics10 is first run you must choose your User Name and/or email address and enter it into the software. You also will need to enter the Serial Number that was supplied to you when you purchased the product. Enter the serial number exactly as shown using only the numbers 0-9 and letters A-Z.

The combination of your email and serial number is your unique key to DCArt10 or DC Forensics10; no other personal information is used.
After you have entered your name and Serial Number, you must then register the program with Diamond Cut Productions. The registration process can be done automatically over the internet or via phone or email.

If you choose automatic registration, DCArt10/DC Forensics10.xx will attempt to contact the servers at Diamond Cut Productions and automatically register your product. The registration process initiates our server to create a registration code and emails that code back to you. You will receive an email with the registration code; use this code in your software. Save it for your records, should you ever need to re-install the product.

If you register by phone or email, you will receive a registration code from Diamond Cut Productions that you must enter into the program to complete the registration. We strongly encourage either automatic or email registration. .

Note 1: Diamond Cut Productions is the ONLY authorized source of registration codes, regardless of where you purchased your product.

Note2: DCArt10 and DC Forensics10 require a User Name or email address, Serial Number and Registration Code from Diamond Cut Productions, Inc. After you receive your Serial Number and have entered a User Name or email address, you will receive a registration code via email. Record those data somewhere for future reference. The most common support question is “my hard-drive crashed and I just replaced it. I have my serial number but I do not remember my user name. Can you retrieve my user name from your server database for me?” Of course, we can usually do that, but it will cause you an unnecessary delay. Once you have a successful installation, you can print out these data from the Help/About Diamond Cut Audio Restoration Tools, XX.YY menu.

Installing the software on a non-internet computer (Lab Computer Installation)

Installation on ‘”internet-isolated” computer (DAWs), often found in Forensics laboratories involves a different procedure to maintain security. It involves the use of a proxy computer. Since your license permits two computers for one user, here is how to install the software on a computer that does not touch the internet. Install the software and register it on a non-lab computer (your administrative computer for example). This is your proxy computer. Test the software and make sure it is registered as indicated under the “Help About Diamond Cut Restoration Tools”. Copy and Paste the Serial Number and Registration Codes into your Word Processor. Place that document containing those two numbers onto a USB Memory stick. Then, make a copy of the .msi installer onto a USB memory stick (thumb drive). Bring the USB memory stick and move the .msi installer into a directory under the lab computer’s C drive. Install the program. Next, bring up the serial number and registration code on the lab computers word processor. Copy the Serial Number and paste it into the Serial Number field of the software. Next, copy the Registration Code and paste it into the registration field of the software. We recommend that after the process has been completed that you delete the Forensics software from the proxy computer.

Step Three - Configure Your DCArt10/DC Forensics10 Program

DCArt10/Forensics10 does not require much configuration, but it does require that your sound card be installed and working properly. If you have more than one sound card in your system, make sure the one you wish to use has been selected in the Sound Card Selection screen (in the Edit/Preferences/Soundcard Menu).

Next, check the Temp File Path under the Edit/Preferences/General Menu. DCArt10/DC Forensics10 automatically assigns temporary file names for files that are being processed. You should set this temporary drive path to the disk drive that you wish to use for audio editing. This is usually the drive with the most free disk space. Keep in mind that high quality (44.1 kHz – 16 bit) Stereo recording consumes 10.5 Mbyte of disk space per minute.

Generally, this is where we get calls from customers who experience one form of problem or another. This is usually because most people don't record onto their computer's hard drive very often and they may not have their system set up for this type of application. We're happy to help, but before you call us… please read the next few paragraphs:

1. Make sure you have the output of your audio source plugged directly to the "Line" input of your sound card. (Don't use the Microphone (Mic) Input of your sound card for line level signals because it can cause clipping distortion.)
2. If you have a turntable, make sure that you are using either a stereo preamp or the "Tape Out" on your Stereo system in between the sound card and the turntable. Your turntable cannot generate enough signal level on its own to make a good line level recording. You'll need the additional amplification to boost your turntables signal to be compatible with the "Line" input of your sound card.
3. If you try to play or record and you get the message "Cannot play the specified format…etc." This is a message generated by Windows…not our product. It is telling you that DCArt10/DC Forensics10 is trying to play a .wav file through a device that is not able to do this. This could be a result of Windows having a modem listed as the primary device for playing .wav files…or some other non .wav device. Check in

your Start Menu/Control panel settings to make sure that your sound card is listed as the primary playback device. Also, make sure DCArt10/DC Forensics10 is set correctly for your sound card under the Edit/Preferences/Soundcard menu.

4. If you can hear audio playing through the speakers but the record VU meters are not jumping, then check your sound card's mixer. Many times, the recording mixer is turned down or muted. Just click on the mixer (in many cases it's a little yellow speaker icon located in your Windows Tray beside your clock), and go to Options/Properties and look for "Recording". Click on "Recording", then click "OK" to select your input mixer so you can turn up or select your Line Level inputs.

   Important Note – also make sure your "Line" input is selected in this mixer – not your "Mic" input (unless you are attempting microphone based recording).

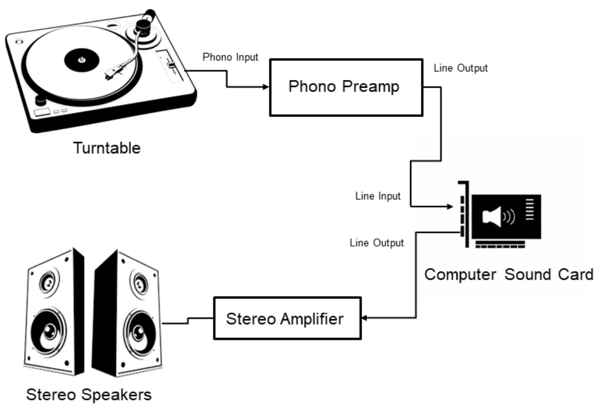
We'll cover more in the Troubleshooting section of the manual, but these are the highlights.

Step Four – Connect Your Computer to Your Audio System

Your sound card has both an input and an output. The output is used when you play an audio file and the input is used when you record audio into the computer. Refer to your sound card manual or the little icons on your soundcard to identify which jack is for output (playing back audio) and which is for input (recording audio). Often, a small speaker or headphone icon represents the output jack.

We first want to connect speakers to the output of your sound card so we can hear audio. You probably already have speakers connected. If you do, then fine. Just leave them alone. If not, you'll need to connect the output of your sound card to either an amplifier or amplified speakers having line level inputs (-10 dBv).

Note: Your sound card records and plays audio at a certain established Voltage level. This standard level is referred to as "Line Level". Just about all devices play or record at this exact audio level so they're all compatible with each other. CD players, Tape Recorders, Amplifiers, Receivers, VCRs, DAT machines, Mini discs and most other devices all play or record this standard "Line Level" signal amplitude. This means you can plug your tape player output or other line level device directly into the input of your sound card since they both use the same signal level (amplitude).



Wiring Things Up!

Next, we need to connect a signal into the input of your sound card so that we can record this audio onto your computer's hard drive. If you are using a line level device, just plug it into the sound card "Line" level input using whatever adapter cable mates your player device (i.e., tape recorder) to the input jack on your sound card.

<u>Critical Note on Turntables</u> - There are two common playback devices that are not line level and which can't be plugged directly into your sound card line level input. The first is a turntable with a magnetic phono cartridge. A magnetic phono cartridge produces a signal that is much lower in amplitude than line level sensitivity. Your sound card cannot use this signal as it is; it must be amplified up to the standard line level before it is applied to the line input of your sound card.

If your turntable has a magnetic phono cartridge, you must use what is called a magnetic phono preamplifier as the first element of the signal path. If you have a stereo system with an input jack on the back-labeled “Phono”, you probably already have a magnetic phono preamp. It’s in your stereo system. Just leave your turntable connected to the “Phono-Input” of your stereo and connect a Line Level output of your stereo to the sound card input. Many times, this line level output is marked as “Tape Out”. Now, any audio (including your turntable) you can hear from your stereo system will be available to your computer for recording. Another option to consider is the use of a “Flat Phono Preamplifier”. Flat preamplifier front ends, when used in conjunction with the Diamond Cut Virtual Phono Preamplifier (VPP™) provides a high level of EQ curve flexibility as well improved sonic performance of your transfers. For more information on this technique, you can download an Application Note from our website located at:

http://www.diamondcut.com

Navigate to the Application Note section of the site and look for AN 8 titled “The New Way of Recording LPs”.

Note: A more direct link to this Application Note is:

http://www.diamondcut.com/AppNotes/TheNewWay.html

If you don’t have a stereo system to connect your turntable to, or it’s in another room, you’ll need a stand-alone preamplifier. These are not expensive and you can purchase one at:

https://www.diamondcut.com/st3/product/flat-response-phono-preamp/

Again, this device takes the tiny signal from your turntable and amplifies it so that it’s now a standard Line Level signal that can be plugged directly into your sound card. An application note describing its use can be found here:

https://www.diamondcut.com/AppNotes/AN-13%20DCP-47K-F%20Flat%20Preamp%20Operating%20Instructions.pdf

<u>Note 1:</u> Low cost turntables utilizing Ceramic Phono Cartridges do not require a special pre-amplifier because their output signal levels are already at approximately line level.

<u>Note 2:</u> Be sure to connect the turntable-grounding wire (strap) to the chassis ground of your Phono Preamplifier or to the computer chassis itself to minimize ground loop induced hum.
Another device that does not develop a standard Line Level signal is a microphone. If you want to record from a "mic", just plug it into the jack on your sound card that is labeled "Microphone". If you need to use this mic jack, again, refer to your sound card manual for specifics.

<u>Note 3:</u> Some laptops have only two analog audio jacks. One is always an earphone / audio output jack while the other is often labeled "mic". Many laptop computers allow you to convert the functionality of this "mic" input over to that of a "line" input via the sound system chipsets driver routine control panel. Please refer to the manufacturer of the laptop for details on this mic/line level switchover functionality.

In most cases, you'll simply be taking the output of your source to the input of your sound card. This is pretty easy to visualize since the computer is really acting like a familiar tape recorder with an input and an output. Many audiophiles have very sophisticated and complicated audio systems that allow many different types of hookups. Therefore, there are many alternative methods for connecting your computer to a sound system in order to be able to use DCArt10/DC Forensics10. Here are several methods:

(NOTE: Typical users will not have to use these or other complicated hookups – just supply an audio signal to the input of the sound card and listen to it on the output.)

<u>Method #1:</u> **Using a home stereo tape-monitoring loop**

1. Connect a stereophonic magnetic phono pickup system to an audio pre-amplifier with magnetic phono equalization inputs.
2. Connect your line level sound card input to one of the pre-amplifier's tape recording outputs.

3. Connect your line level sound card output to one of the pre-amplifiers tape monitoring inputs.

<u>Method #2:</u> **Using a DAT with digital (S/PDIF) inputs and outputs**

1. Connect a stereophonic magnetic phono pickup system to an audio pre-amplifier with magnetic phono equalization inputs.
2. Connect the DAT machine analog output to a tape monitoring input on the pre-amplifier.
3. Connect the DAT machine analog input to a tape output on the pre-amplifier.
4. Connect the Digital Output of the DAT machine to the Digital Input of a "Digital-Only" sound card in your computer.
5. Connect the Digital Input of the DAT machine to the Digital Output of the "Digital-Only" sound card.

<u>Method #3:</u> **Using a mixing board and Analog Sound card**

1. Connect a stereophonic magnetic phono pickup system to a magnetic audio pre-amplifier (these are available without all of the bells and whistles associated with a full-blown home audio pre-amplifier).
2. Connect the Outputs of the magnetic pre-amplifier to two of the line level inputs on your mixing board (one input for each channel).
3. Connect the line level outputs of the analog sound card to another pair of line level inputs on your mixing board.
4. Connect your tape recorder (DAT or Reel-to-Reel or whatever) line level outputs to another pair of line level inputs on your mixing board.
5. Connect any other input devices you may require into the remaining inputs of your mixing board.
6. Connect the Main mixer output to your power amplification system.
7. Connect the Monitor Outputs from your mixer to the line level input of your sound card.
8. Connect the tape recorder line level input to the Stereophonic Headphones output jack on your mixing board.

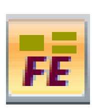
Warning: Method #3 is the most versatile method for setting up a small sound restoration lab. However, because it is so versatile, feedback loops are easily created which can produce very annoying and potentially dangerous signal levels (to your ears, power amplifier and loudspeakers). So you must be careful not to allow the output of a device to feedback into the same device when operating the mixing board. Always check twice before raising a slider control on your mixing board utilizing this method.

Step Five – Turn Screen Saver and Background Tasks Off

In some cases when recording, if your screen saver or other background computer tasks automatically comes on, it may interrupt your recording or add glitches that you didn't want. It is always better to turn these functions off before recording. Though this is less of a problem with faster computers, most users will feel more secure if they turn off automatic backups, screen savers, etc. This allows the computer resources to fully dedicate themselves to recording clean audio.

Step Six - Choose your Operating Mode

The software can be operated in either of two basic operational modes. Choose your operating mode based on your personal preference. Beginners will likely find that the Fast-Edit mode is easier to get oriented with since it works more like traditional audio editing programs. More advanced users will likely switch to the Classic mode since this offers a fully optimized restoration environment. You can quickly switch between these two modes of operation via the "One Click Edit Mode Switch" icon. Just click on the icon and it will toggle between the two alternative editing modes.



"One Click Edit Mode Switch" Icon

Fast-Edit Mode

The **Fast-Edit** (single file editing mode) mode operates much like a word processor where all editing is done on one file. The original file is not modified until a "Save" is performed. Fast-Edit mode maintains a separate history file representing the editing sequence and offers almost unlimited Undo capability. The advantage of the Fast Edit's Single file editing mode is its "greased lightning" speed, leaving you with more time for your domestic chores. Where editing examples are used in this manual in order to highlight the use of an editing or filtering sequence, the **Classic Edit** (Source and Destination) mode is utilized.

In this **Fast-Edit** mode, you preview the processing results, and if not satisfactory, you can use the "Undo" feature found in the Edit menu. Also, you can highlight a particular step in the **Fast-Edit** history in order to quickly jump back to a previous editing state. The editing processes will be temporarily undone back to the selected point in the Edit History Monitor. If you want to permanently go back to a previous editing state you can simply double click on the last edit you want to delete in the **Fast-Edit** Window (display). All editing done after that point will be removed and you can continue your editing session. The Delete function can also be found by clicking with the right mouse button. Fast Edit temp files are maintained in the same directory as the source file and include elements of the original file name for ease of identification. The **Fast-Edit** history file uses the extension of .ses. Temporary files created by your Diamond Cut Software precede the extension with **dctmp**xx. They are stored in the TempFiles directory and can have various file extensions such as .wav, .pkf, or .ses.

Classic (Edit) Mode

DCArt10/DC Forensics10 **Classic (Edit)** mode usually operates on files in a non-destructive manner. The Source and Destination mode (the Classic technique) involves the use of a "Source and Destination" set of files. When a file is processed with a DCArt10/DC Forensics10 filter or effect, the software reads the Source file, modifies it with the selected filter or effect, which then writes it to the Destination file. The main workspace of DCArt10/DC Forensics10 always has a Source and a Destination file in the **Classic (Edit)** mode. This mode of operation has a few important benefits:

1. The original source file is not modified, leaving it available for instant comparisons with the processed version.
2. The original material can always be recovered if the results of processing are found to be unsatisfactory.
3. Selected sections of the file can be reprocessed using different filter parameters or different filters entirely (refer to “Sync Files” mode).
4. Every filter that is run yields a new file which can have yet another filter run thereon. All of these intermediate files are always available so that the users can instantly go back one or more steps in the restoration process. These files are all stored in the .wav format.

The Source and Destination Workspace

When you open a .wav file in DCArt10/DC Forensics **Classic-Edit** Mode, two workspaces will appear. The top one, called the Source Workspace will display an envelope consisting of the program peaks of the .wav file just opened. If you are using Peak files (a user preference), the entire waveform should be visible in the window. If you have peak files turned off, then the amount of the .wav file that will be displayed is determined by your preferences "display limit" settings. Both settings can be found in the Edit/ Preferences/General section of the Edit Menu. The display will consist of a black signal on a yellow background (depending on the user’s preference settings).

The Destination Workspace just below the Source workspace will contain no waveform information initially, and will contain a gray background color. Both of these two workspaces display amplitude on the Y-Axis (vertical) and time on the X-Axis (horizontal). When you initially open a file, the entire file is displayed, and is periodically represented by a sample of the peak of the waveform envelope. When you Zoom-in on a portion of the waveform, at some value of magnification, you will begin to see continuous waveforms, rather than impulse representations of your .wav file signal. For more information regarding Zooming-In on a .wav file or Zooming-Out on a .wav file, please refer to the sections entitled "Zooming-In & Zooming-Out on portions of a .wav file". Please note that the active workspace is always shown in yellow (the software default setting).

If you are working with stereo .wav files, the workspace will display a pair of waveforms. The top waveform in either of the workspaces represents the left channel while the bottom waveform represents the right channel. If the signal is monophonic, only one waveform will be seen in the workspace(s).

At the top of the DCArt10/Forensics10 screen is a Title bar, which contains the name of the opened Source .wav file. At the bottom of the DCArt10/Forensics10 screen and on the right side, you will see five little boxes. The first shows the Mode in which the file was recorded or processed ("Stereo" or "Mono") followed by the Sample Rate that was used to create the file and then the Bit Depth. The fourth box give the total running time of the Source .wav file and the final box shows the space remaining on the hard drive being used.

After a .wav file has been processed by one of the functions under the Filter command, the output of that file will be sampled just like the Source file and displayed in the Destination workspace just below the Source workspace. It will become highlighted in yellow just following the completion of a processing session.

At the bottom of each of the two workspaces, you will see several time displays. Each display is indicated in Minutes: Seconds: Milliseconds. The time display on the left side of the workspaces indicates the starting time of the portion of the .wav file being displayed in the particular workspace. The time display on the right side of the two workspaces indicates the ending time of the displayed portion of the highlighted .wav file. When a file is initially opened, the display on the left will indicate 00: 00: 00. The right display will indicate the total time duration of the opened .wav file. If you use the Zoom function, the left display will now display the start time of the highlighted Zoom-In portion of the .wav file. The right time display will indicate the end-time of the highlighted Zoomed-In portion of the .wav file. The total time duration of the Zoomed-In highlighted portion of the .wav file will be displayed on the status bar located below the workspaces.

At the right hand side of each of the two workspaces, you will see two vertically oriented slider controls next to one another. These are useful for viewing details in a selected portion of a waveform which has been

Zoomed-In on. For example, there may be a small transient that you want to see in more detail that is riding on top of a much larger waveform. The control on the farthest right is the “Display Gain” control. Using your mouse, you can move this control up and down in order to change the display gain. Moving it downwards will increase the gain of the time display. This will cause the waveform to appear larger on your screen (you can also adjust the Display Gain using your mouse scroll wheel). However, it may get so large as to move the portion of the waveform in which you are interested off of the top or the bottom of the display screen. The control just to the left of the gain control is the “Offset” control; this is used to move the entire portion of the waveform in which you are interested back into view. You should experiment with these controls a few times to get a feel for how they behave, and then you will begin to understand their usefulness.

At the bottom of each workspace, you will see the “Time Axis Scroll-bar”. This control is also operated by the left mouse button, and is used to move the “Play Pointer” to various locations within the display workspace. Sometimes, there can be a few seconds of delay when using this slider, so be patient as it performs the calculations to keep up with your commands. When you are Zoomed-In on a portion of a file, the slider control can be used to move the display start-point within the highlighted field using either the slider with your mouse, or by using the arrow controls which are located at each end of the slider. The Time Axis Scroll-bar position is always relative to the entire file length, no matter how zoomed-in on a particular waveform you may be. Clicking on the right arrow button will move the waveform to the left of the workspace 1/10th of the overall display length and clicking on the left-hand arrow button will do the same thing, only moving the file in the opposite direction. If you click on the Scroll-bar (not the slider control itself), the waveform will move one full frame to the left each time you click.

Note 1: The Time Axis Scroll-bar is inactive when you are fully zoomed-out.

Note 2: If you are not using Peak Files, then the software only reads the first few Megabytes of a .wav file for the initial display. No .wav file processing operations are adversely affected by this action. Portions of your .wav file not shown on the display can still be played, filtered and operated on just as if they were displayed. To set the size of the waveform that will be displayed, use the Preferences dialog box (found under the edit menu) and increase the "Display Length Limit" to the size of the file you wish to be displayed. Keep in mind that the larger the display size, the longer it will take to initially open a .wav file. We recommend the use of Peak Files for faster future displays of the .wav file data.

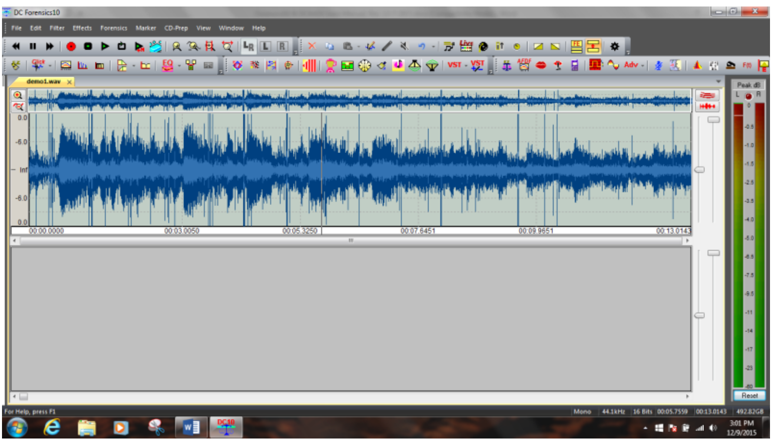
Note 3: The Destination Workspace can be converted into an Audio Spectrogram time synced to the Source Workspace. Zooming in and out of either time or spectrogram view remain synchronized. Please refer to the appropriate section of the manual for details on the operation of the Audio Spectrograph/Spectrogram.

Note 4: How to Choose Your Mode – To switch between **Classic-Editing** and **Fast-Editing** mode is fairly simple. If you start with one and decide you want to try the other one, you can simply go to the Edit Menu and click on Preferences and the General Tab. There you will see a check box that enables Fast Edit Mode. You then need to exit the program and re-run it for the changes to take place or you need to close any open file and then re-open it for the new mode to be invoked. Alternatively, you can simply click on the "One Click Switch" icon which toggles back and forth between the two editing modes.

Step Seven – Testing Your System

By now you should have installed the software and connected your speakers to the sound card output and some audio playback device (such as a tape player) to the sound card input. It's time to make sure your efforts have borne fruit.

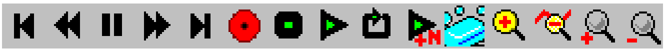
The first thing we're going to do is run the DCArt10/DC Forensics10 program. Do this by double clicking the new DCArt10 or the DC Forensics10 icon on your desktop. Put away the tips screen and click on File/Open Source. You'll now see a familiar Windows file selector box. Navigate to C:/program files/Diamond Cut Productions/DCArt10/Wavefiles (or wherever else you chose to install the program). If you are using Forensics10, the demo files are located as shortcuts in the Start menu. Alternatively, simply go to the File menu (left-most menu column) and use the Open Demo Wavefiles feature. Select the file called "Getting Started Demo" (demo1.wav) can be found under the Open Demo Wavefiles). This is a standard .wav file and was included with the program. We'll actually clean up this file in the next section, but for now we just want to play it. After the file is selected, it will open in the program and you'll see the graphic drawing of the audio waveform on your display. It will look something like this:



A Monophonic File is opened

The small band of waveforms above the main waveform is your waveform overview display. It will show you where you are zoomed-in (on the main file) with respect to the overall file via a highlight. Note the zoom-in and zoom-out icons near the left-hand side of the waveform overview display. You can drag the highlighted area left and right and that action will be reflected in the primary waveform display below.

Notice all the nice icons at the top of the display. If you move your mouse over any one of them and leave it there for a second, a little box will open telling you what that icon does. To play our file, we need to find the little icon that looks like a right pointing triangle. This is the standard symbol for "play" that you'll find on any tape deck. Click on this icon and you should hear audio and see the cursor move.



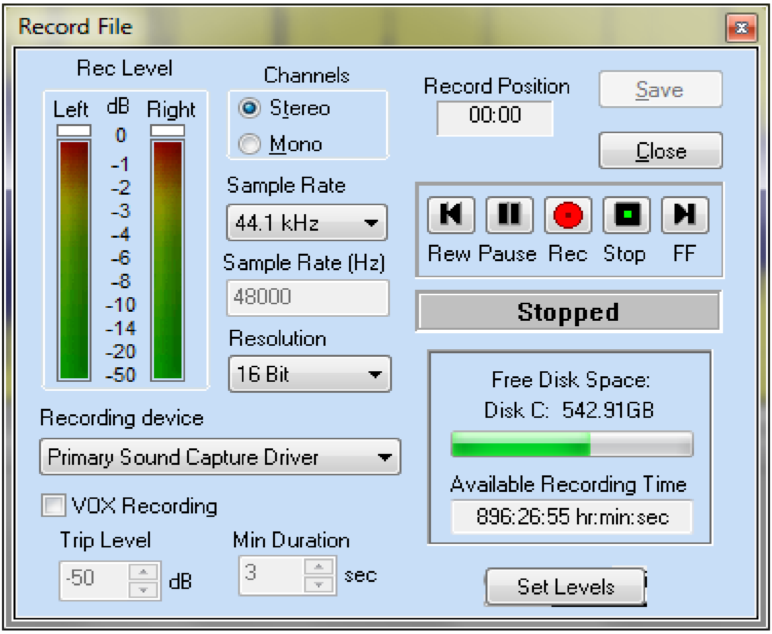
The Diamond Cut Play Toolbar

<u>Note</u>: If you aren't getting audio at this point, check out our Troubleshooting steps found later in the user's manual.

Let the audio play to the end of the file and it will stop automatically and reset itself so you can play it again. Let's play it again using a keyboard shortcut. Shortcuts are simply keys on your keyboard that perform a function without having to use the mouse. Power users love shortcuts and the Play shortcut is the easiest of all. Just hit the spacebar now and the audio starts playing. Hit the spacebar again to stop the audio from playing. Easy, isn't it? Many beginners like to spend hours doing this, but let's move on to test the recording capability of your system.

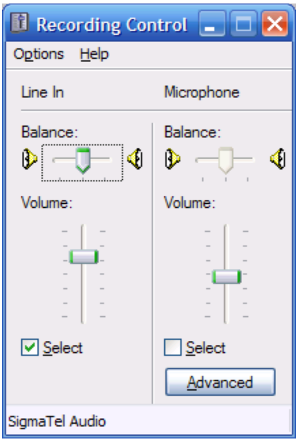
For our recording test, we're going to assume you have a tape recorder connected to the input of your sound card. A CD player or even a turntable with a preamp uses exactly the same process, so just make sure you're hooked up correctly.

To record, start your tape recorder playing a pre-recorded tape (or start playing an LP, etc.). You should hear the audio coming from your speakers. Now click the red Record button on the toolbar in DCArt10 / DC Forensics. A nice record screen comes up which looks like this:



The DC Forensics10 Record Window

This screen will allow you to set the recording parameters and make sure the recording is happening correctly. Let's start by making sure we are set to record a stereo file at 44.1 kHz, 16 bit. Just use the drop down boxes to select these parameters. Note the Recording Device box. Your sound card should be listed here. If not, drop the box down and choose the sound card that has the audio being fed to it. If you click on the "Set Levels" button, something like the following drop-down will appear which will allow you access to the Windows Mixer controls.



Windows Sound Card Mixer Control

Now click the Pause button in the Record Window. This puts the program into what is called Record/Pause mode. The VU meters on the left should now start dancing as the program is actually seeing the audio from your tape recorder.

If those meters are dancing, adjust the input level until they are reasonable (not too high or clipping and not too low) and you are now done with testing your installation. Click “Stop”, and then click “Close”. Answer “yes” when the program asks if you want to discard this recording. You now have a perfectly working audio restoration workstation and you can go on to the next section where we’ll teach you to use this powerful tool.

If you found that your meters didn’t jump when you hit the Pause button, the problem is very likely caused by your sound card being set to the Mic input and not the Line Level input. Full troubleshooting help is included later in this manual, but this common problem is easily and permanently solved by double clicking the little yellow speaker icon in your Window tray – just to the left of the clock on the bottom of your screen. Now choose Options and then Properties. Click the record button and then click OK. Now click the check box under Line Input and your meters will start to jump in DCArt10/DC Forensics10 indicating that all is now well. You just turned on the Line Level input – which is where our audio is, after all.

Now that your system has been installed and checked, it's time to get to the software and perform an actual restoration project.

Step By Step Guides

(Getting Started Guide)

In this section, we're going to take you by the hand and lead you through some complete examples of typical restoration projects. You'll remove noise and "sweeten" the audio. When you are done, you'll be licensed to drive the product yourself without training wheels.

This entire training process will take you about 15 minutes and will give you all the basics you need to know to use the program. Please follow along step by step thru the first three examples – Super Easy, Easy and Advanced Record Restoration. If you are a Forensics audio user, please go thru all three Step By Step Guides. In these Guides, we'll tell you exactly what to do and even explain why it is you are doing these things. It's important that you follow along exactly as one step builds upon the next.

Note: The actions in this section of the user's guide that you are expected to take are highlighted in ***Bold Italics***. Let's get started!

Super Easy Record Restoration - - - Step By Step Guide

If you are new to audio restoration and feel a bit intimidated with the large number of filters and tools in DCArt10, then EZ Clean™ is for you. Our first guide is designed to get you going almost instantly. The EZ-Clean™ filter will remove clicks, pops, hiss and other surface noise from music recording almost instantly. Here we go.

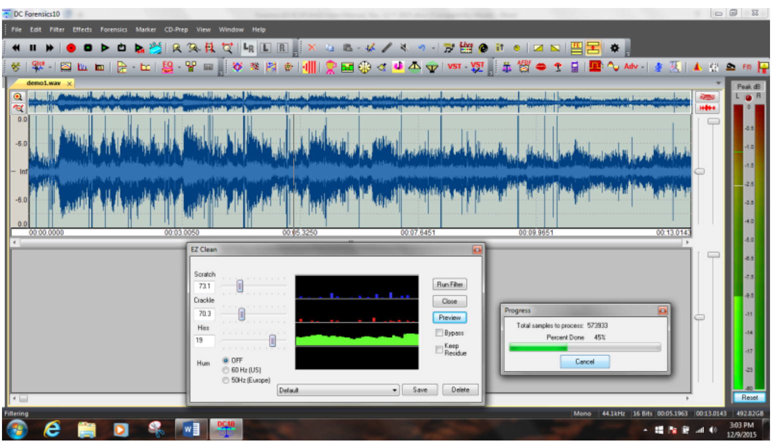
First, we need to run the program. If it's not already up, just ***double click the icon on your desktop*** to start it. DCArt10/DC Forensics10 starts with a tip screen, which offers quick suggestions as to how to use the program. You can disable this by clicking on the check box in this screen.

In this guide, we're going to use the Classic Source/Destination mode to run you thru the software. You can choose this mode by ***clicking on Edit/Preferences*** in the program. Make sure the check box that is labeled Enable Fast Edit is <u>unchecked</u>. ***If you just unchecked it, close and restart the program*** to get going in Classic mode.

Next, ***open the Demo1 .wav file*** the way we did in the testing section above.

Listen to this file by either clicking the Play button or hitting the spacebar – it's full of clicks, hiss, low frequency noise, etc. It's a mess**.** ***Stop the playback*** when you're done listening.

And, now let's get to the fun part. ***Click the Filter Menu and choose EZ-Clean***. The filter looks like this:



EZ Clean™ Is Almost Too Easy

Notice that there are only three sliders, one for scratches or clicks, one for crackle (or small clicks) and another for hiss or other continuous type noises. We are simply going to listen to the audio and move these three sliders as we listen.

Move each slider to approximately the settings shown above. We don't need to be perfectly accurate, just set them similar to what you see. When you move these sliders to the left, the filtering becomes more aggressive. ***Now click the Preview button***. You'll start to hear the audio. To stop a Preview, just click the preview button again and it will stop in a toggle switch sort of action.

Now, in Preview mode, listen for a second or two**.** ***Now click the checkbox labeled Bypass***. This "bypasses" the filter and stops the filtering. You are now hearing the original music without the filters in place. Note the large amount of clicks and hiss.

Uncheck the bypass box to start filtering again. What a relief! That's much better isn't it? But, you can do better yet. ***Move the Hiss slider a bit farther to the left*** until you remove more of the random noise. Remember, moving the sliders to the left, makes them filter more, so just ***slide them until you are happy with the result.*** If you move the controls too far to the left, you may introduce digital artifacts and/or distortion into the signal. You must strike a balance between noise reduction and artifact/distortion creation with these tools.

Now note the Hum filter on the EZ-Clean screen. If you live in the US, you'll check the 60 Hz box to remove power line hum. If you live in Europe, you'll check the 50 Hz box. There are many other tools to remove larger amounts of hum in DCArt10, but this one is quick and easy.

Now, ***stop the preview by clicking on the Cancel Button in the <u>Progress- box</u>***. ***Click on "Run Filter"*** and you now have a destination file that is fully cleaned! Could this possibly be any easier?

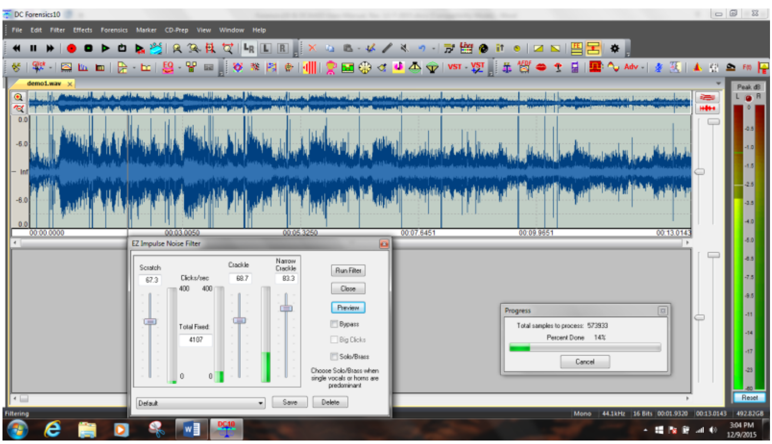
Please continue on with the Easy Record Restoration guide below. It starts your education on how to use individual tools and goes into more depth on the overall concept of the Diamond Cut software.

Easy Record Restoration Step By Step Guide

Since EZ-Clean™ is so easy to use, you don't really get a feel for the overall software program and its depth of capability. This guide assumes you will use individual tools and will perform a restoration in a series of steps.

You will be working with DCArt10/DC Forensics10 by choosing filters to apply to your audio .wav file. Some of the filters remove noise and others enhance the audio though they are all referred to in this guide as filters. You choose a filter by identifying the type of noise you want to remove and then selecting the filter that removes that type of noise. Make sense so far, right? In just about every case, when clicks and pops are present, we want to remove them first. Trust us, this is the right first step whenever you get ticks, clicks and pops on records or other recordings (called impulse noise).

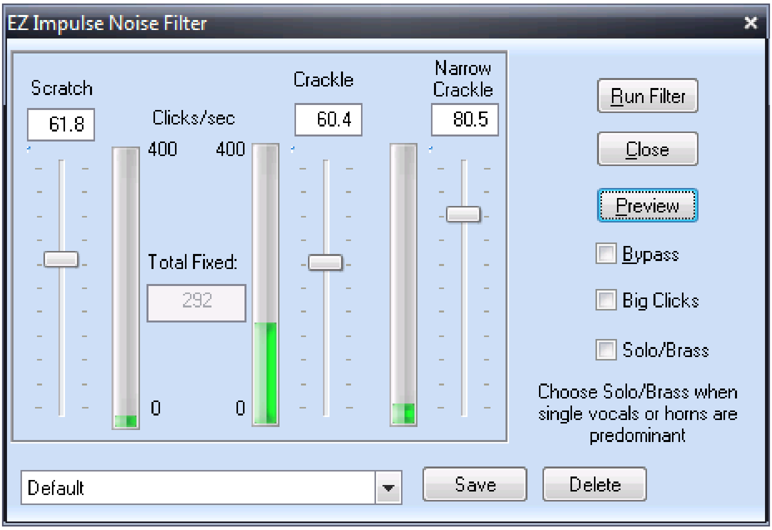
To remove the clicks, we'll choose the EZ-Impulse™ filter. Clicks are short noise impulses so it makes sense to use this filter. To choose the filter, ***click on the Filter menu item and choose EZ-Impulse Noise***. You'll see the filter open like this:



The EZ-Impulse Noise™ Filter Appears

Note that the filter takes up only a portion of the screen. You can still see part of the waveform and all the menus and icons. You can move this filter window around on your screen and position it wherever you want. Just ***grab the title bar at the top of the filter window and drag it around***. Try it now.

Now let's focus on the filter itself. Just about EVERY filter in DCArt10/DC Forensics10 will work the way this one does. Once you learn this particular filter, you'll be able to use all of them! Because of this, we're going to cover this filter in great detail. Be sure and follow along.



The EZ-Impulse Noise™ Filter

In the filter window, you will see several features. First, you'll notice sliders that control various filter parameters. These sliders can be adjusted <u>while you listen to the audio</u> so you will instantly hear the result of any changes you make. Our sliders in this example are labeled Scratch, Crackle and Narrow Crackle in this filter.

Next, you'll have Radio buttons that control other aspects of the filters. Again, you can usually change them while listening, so you'll hear the results instantly. The Radio buttons in the EZ-Impulse filter are Bypass, Big Clicks and Solo/Brass. The EZ-Impulse filter allows you to change the Bypass and Solo/Brass buttons while previewing, but not the "Big Clicks" button. You can only change that one prior to Previewing.
In every filter or effect, you will have a Preview button. This is the most important button here. This button will start the audio playing while the filter is processing it. You will hear the results of the filter instantly. This makes for easy adjustment of the filters. The Preview

button allows you to adjust any/all filters before actually running the filter on the file (the “Run Filter” button).
Also in every filter, you will have a “Bypass” checkbox. This takes the filter in and out of the audio stream instantly. When you listen to a filter being applied by the Preview button, you may want to be able to compare the processed audio to the original audio. Clicking this checkbox will bypass the filter and you’ll be hearing the original audio. Un-checking it will instantly put the filter back into the signal path. This way, you can “fine tune” even subtle nuances in the audio signal with DCArt10/DC Forensics10.

Lastly, every filter will have controls for Presets. A preset is a saved group of settings for this filter. Go ahead and ***drop down the Preset box now*** – it’s the white box at the bottom of the filter. ***Click on some of the presets*** and watch the sliders as they move to good starting points for common tasks. You can tell a lot by looking at the name of the presets. ***Now select the preset labeled Default***. This one is already set up with good settings for our Demo1.wav file. Every filter will also have a Save and Delete button that allows you to save your own presets under any name you want – and delete them too!

You’re probably ready to try this filter by now, but there is one more thing that is common to each filter that you should know. ***Hit F1 now on your keyboard***. Notice that our online Help comes up with information on this specific filter. This context sensitive help is available for each and every filter.

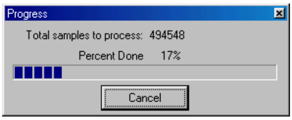
Put the Help screen away and let’s clean some audio. If you’ve already been listening to this filter, then shame on you for jumping ahead. Simon has not said click the Preview button yet. As punishment, please go back to page one of this manual and start reading again. We’ll wait for you here.

Welcome back. Now ***let’s get started by clicking the button labeled Preview***. You will hear the audio as it is being filtered. You will still hear the low frequency rumble and the hiss, but the clicks should be gone. ***Let it preview all the way to the end of the file.*** Notice that once it reaches the end, it will automatically start over at the beginning. This is called Looping and is automatic when you are previewing with a filter (though it can be turned off in the Edit/Preferences screen).

Let's just confirm that the clicks are gone. To do this, ***check the Bypass box*** in the filter while it is still previewing. Now the filter is bypassed and the clicks will once again be audible. Listen for a while and then ***uncheck the Bypass box***. Now the filter is again doing its job and the clicks are no longer heard.

It's time to learn how to adjust a filter. While you are previewing, ***move the three sliders to the bottom***. This makes the filter less aggressive and it will filter less. Notice that the clicks return as the sliders are moved down. ***Moving the sliders upwards*** results in more of the clicks being removed from the file. All of the filters work this way – you just adjust them while you listen. As you might expect, if you move them too far up you'll make the filter(s) too aggressive and you'll get distorted, stuttering or otherwise bad audio – just move them up enough to get the desired result. ***Set them all to 50***. The "Scratch" control is adjusted by the user to attenuate large impulses. The "Crackle" control is for medium sized crackle and the "Narrow Crackle" control is used to attenuate smaller impulses. "Solo/Brass" should be turned on when dealing with up-front vocals or brass instruments like trumpets or trombones.

Now ***click on Cancel*** in the Progress window seen here:



Monitor your Progress

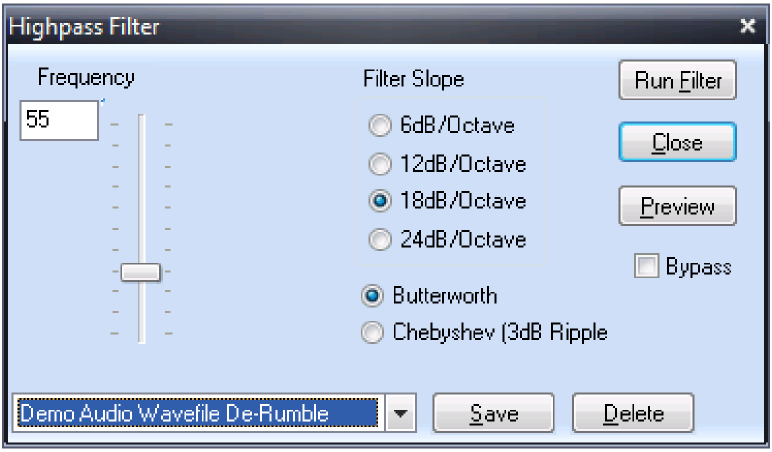
This stops the preview from playing. By Previewing, we have adjusted the filter and confirmed that it is doing its job. The next step is to ***click on the "Run Filter"*** button. This takes the filter just as you have set it and applies it to the demo1.wav file and creates a new file in the lower window. This new file has been run through the filter and has the clicks removed. At this point, ***click on Close in the EZ-Impulse filter*** window since we're now done with it.
Look at the two waveforms. The top one is called the Source. This is where we normally work on a file, preview filters, etc. The bottom is called the Destination and is the result of our filtering efforts. You can

play either one by clicking in the respective window which will highlight that window. Then, click the “Play” button (or the spacebar) to hear either the source file or the destination file. You have not changed your original (source) file at all – rather we’ve created a new cleaned up version (the one found in the destination window).
Now it’s time to remove that rumble sound, but how can we do that if we work on the Source window and we really want to remove the rumble from the semi-clean file in the Destination window? The answer is a little command under the File menu called Make Destination the Source. This moves the file in the bottom window up to the top where we can work on it. ***Click on File/Make Destination the Source now***.

A “Save As” box will pop up and suggest a new name for this file; just ***click on Save*** to accept it.

Note: DCArt10 & DCForensics10 will automatically assign sequential names to new files. While you’re new to the program, always just accept its recommendation as to file names.

Now we are ready to remove that low frequency rumble. To do that, we’ll choose the High Pass filter (an IIR type). ***Click on Filter/High Pass now***. (To get there, go to the Filter Menu\LP BP HP Filters and then select the “High Pass” filter.) A high pass filter will remove all frequencies below a certain value and allow all higher frequencies to pass. Drop the Preset box and ***select the preset called “Demo audio Wave file de-rumble***”. It will look something like this:

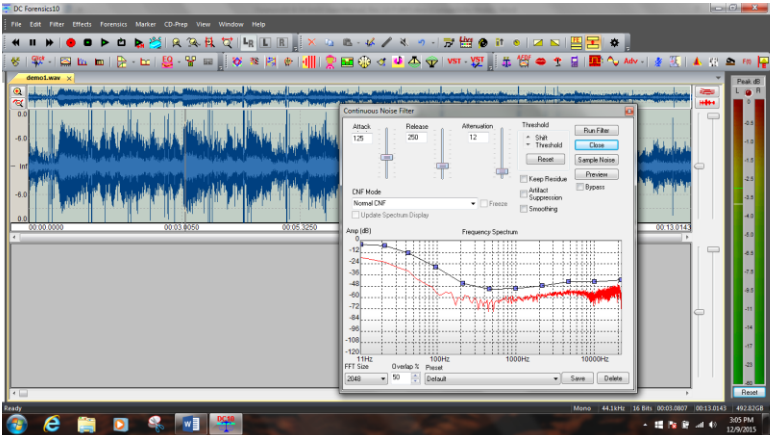


The Highpass Filter

We'll start to go a bit quicker now since you already know what most of the buttons and controls do on this filter. To find out specific info on this filter, don't forget you can call for help by pressing the F1 key. We are going to free you now to preview and play with this filter on your own, but when you're done, return it to these settings by once again clicking on the demo Wave file preset**.** ***Run the filter*** when you are ready.

You now have removed two of the annoying noise types in this file. First, we rid ourselves of the clicks and pops, and then we removed the rumble. Now it's time to get rid of that loud hiss sound. First, remember to ***move our Destination file up to the Source window***.

Now ***click on the Filter menu and select the Continuous Noise Filter***. This filter is perfect for reducing hiss and other types of continuous noise. Also, ***click on the View menu and make sure you have the Time Display checked at this time***. The Time Display box shows you various timing calculations with the program and will come in handy, as you will see. Your screen should look something like this:

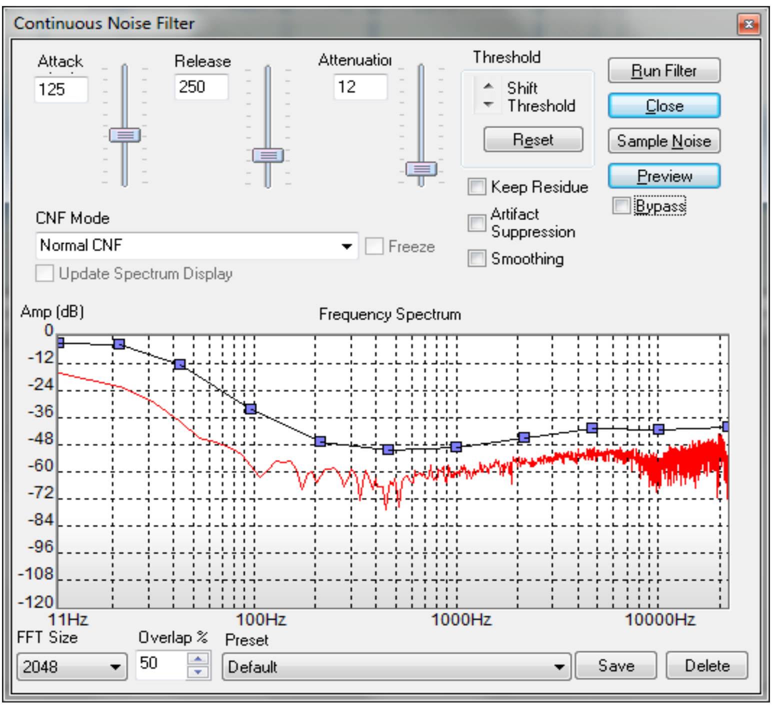


The Continuous Noise Filter is Awake

The Continuous Noise filter is one of the coolest filters in the whole program so follow along carefully here. This filter will remove just about any continuous noise in a program, but it needs you to give it a sample of this noise. Once it is able to examine the noise, it will be able to seek it out and attenuate it. ***Play the file now using the Play button at the top*** and listen carefully to the first couple of seconds. You will hear an area right at the beginning which contains only the noise and without any of the good audio. That's a great spot from which to grab a sample of the noise. To do this, we need to click and drag with our mouse to select this area of the file. The narrow area near the left-hand margin in the illustration above is the area we want to select. Just left click about ½ inch from the left edge and, while holding down the left mouse button, drag all the way left. Let the mouse button go and you will see that area highlighted in bright yellow. This is the area you have selected. Move the mouse pointer over either edge of the selected area and the pointer will turn into a Left/Right indicator. ***Click and drag as necessary to select an area from the beginning of the file that is about .5 seconds long***. ***Use the "Span=" display*** in the timer window to confirm that your selection is around ½ second long.

Now hit the spacebar. You'll play only the selected area. It'll be quick. This allows you to audition a selected area to make sure you are really working on the correct area (does not contain silence or desired signal). You should only hear the hissing noise. Again, your selected area should only include noise and no music.

Now, it's time for the fun part. You will like this. ***Hit the Sample Noise button in the Filter window***. The filter will analyze the noise sample and then display the frequency characteristics of the noise in red. The blue line is the filter that has been designed to attenuate this noise. Notice how they track with each other. Yours should look something like this:



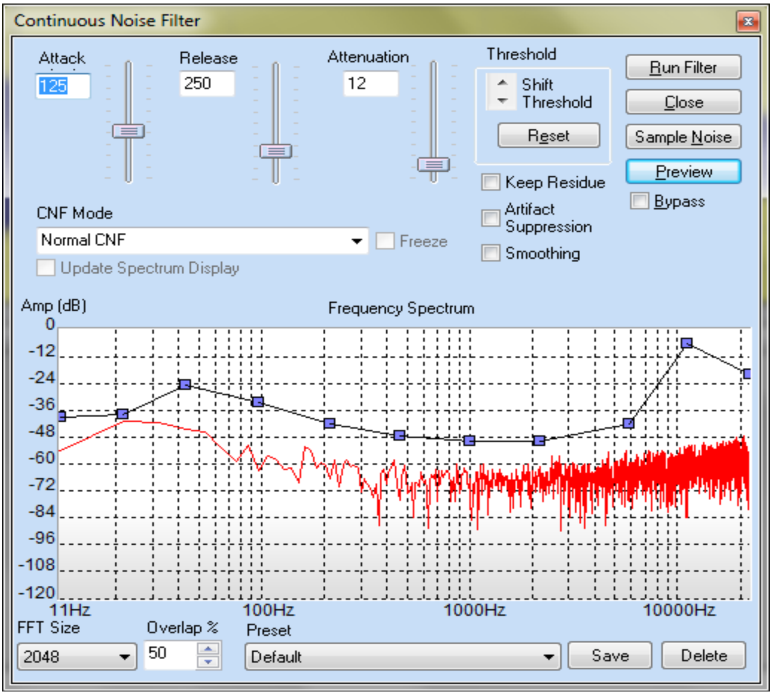
The Continuous Noise Filter Does its Job

If you have jumped ahead again and clicked on Preview, there will be no lunch for you today. First, we've got to explain those little blue dots on the filter line. Notice how the blue line is above the red noise sample? The higher the blue filter line, the more filtering occurs at that frequency. The blue dots are called Control Points and allow you to adjust the amount of filtering for frequency components in the vicinity of the control point manually. We're going to do that in a second, but realize that the Help file contains the complete information on this filter.

Before we click on Preview, we need to again select the whole file so we'll hear the whole thing and not just our ½ second noise sample that is now selected. To select the entire file, just ***double click anywhere in the waveform display area***. Notice how the entire thing gets highlighted in yellow. ***Now, go ahead and click on Preview***.

Listen to the audio. The clicks are gone. The rumble is gone and now the hiss is gone too! Well, not quite. You can still hear a bit of hiss can't you? ***While Previewing, click the bypass button a few times*** to take the filter in an out of the signal path. Yes, the hiss is reduced, but there is still some of it there. Let's adjust this filter to get rid of all of the hiss.

Make sure your bypass button is NOT checked and make sure you are previewing the audio. Now grab the 2nd control point from the right side and move it up a bit as shown below.



Tackling the remaining Hiss

It's like magic, isn't it? The hiss almost completely goes away because you told the filter to be a bit more aggressive on the high frequencies. Remember, moving the blue line upwards makes it filter more. Now ***stop the preview, click on Run filter, and close the Continuous Noise filter.*** If you like that filter setting, you can click on the "Save" button and then name it as a preset so that it can be recalled later.

You now have a Destination file that is completely restored. To finish, ***click on File/Make Destination the Source, accept the file name and you're done***. This file is the completed version and all others can be deleted or saved if you want each processing step before and after the saved file. You can now exit the program or close the files using the commands under the File Menu.

What you've learned in this section

In this quick Step by Step guide, you've learned how to launch the program, how to select filters, how to preview them, how to bypass them, how to load and save presets, what are Source and Destination files, how to apply multiple filters to a single file, how various filters work and much more. Not bad for a few minutes! Next, we'll give you a more advanced Step by Step procedure in which you get to use the most powerful feature of DCArt10/DC Forensics10, the Multi-Filter.

Advanced Record Restoration Step-By-Step Guide

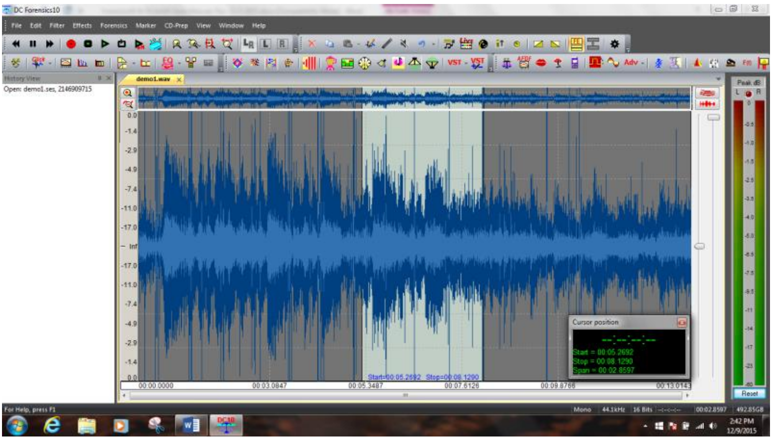
You learned a lot in the last Step-by-Step Guide, so let's continue by restoring this same file using a completely different approach. This will show you the flexibility and power of DCArt10/DC Forensics10 and will introduce you to our powerful Multi-Filter. Also, we will perform this advanced restoration in a Step-by-Step fashion using our new Fast Edit Single Screen mode. We'll assume you've done the Basic Guide above and are therefore comfortable with the basic functions of the program so we will take this at a little faster pace. As before, the actions you are expected to perform will be in ***Bold*** type. Let's get started!

First, we want to make sure that you are in Fast Edit mode. To do this, ***click on Edit/Preferences*** and ***make sure the box labeled Enable Fast Edit is checked (under the "General" tab)***. If it's not already checked, ***put a check mark in the box by clicking it***. Now ***close the program and then restart it***. The operational mode change will take effect when you start the program.

Remember, Fast Edit mode is a single screen mode of operation with no Source or Destination file window. This mode of operation has the advantage of allowing almost instant cuts, pastes and copies on even very large files. It also may be more familiar to users of traditional audio editing programs. You can change from Fast Edit to Classic Mode anytime you want.

Next, we want to ***enable the Fast Edit History box by clicking on View and then checking "Fast Edit History"***. This will open your History box. This box will show you everything that you do with this file and in the order that you do it. It also allows you to return to an earlier state of the file quickly and easily.

Now open the Demo1.wav file in the same way we did before. ***Select an area near the center of the file using a mouse click and drag***. Your screen should now look similar to this:



A Highlighted Section in Fast-Edit Mode

Let's cut out the highlighted area. ***Just click on Edit and then Cut***. Boom! It's gone! Fast Edit mode is called that because it's *FAST*. You will now have two entries in your History window – the top one is for the original file as opened, and the second one is for the version you now see in front of you which is somewhat shorter.

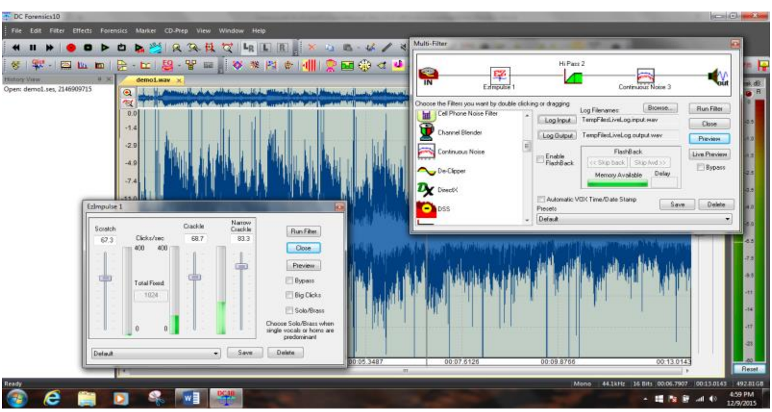
Click on the <u>first</u> entry in the Fast-Edit History window now. The original file instantly comes back! Did we say this thing was fast? Everything you do will show up in the history window and you can bounce around your various versions of a file at will. ***Click on the first and then second entries a few times*** to confirm this and then ***right click on the second entry and choose "Delete"***. You just deleted the cut-up version and we're back to the original file.

Now ***click on the Filter Menu and choose Multi-Filter***. In order to select the entire file for processing, ***double-click anywhere in the waveform display*** or left-click and drag to select a segment of the file. Notice how DCArt10/DC Forensics10 allows you to select an area or the entire file while the filter windows are open. This is a very handy feature.

The Multi-Filter is one of the most powerful features contained within the DCArt10/DC Forensics10 software products. It allows you to string together many filters and then to preview or apply them sequentially or all at once to your file. This saves a lot of time and makes it possible for users to create some incredibly sophisticated filter setups.

Notice that the Multi-Filter has an input on the left side and an output on the right side. Between the input and output is a signal path. You can drag any filter (or multiple instances of the same filter) that you want into this signal path. At this time, ***remove any filters that are in the signal path by clicking on them and dragging them out of the signal path***.

Remember when we went through the Basic Step by Step guide (above), we used three filters: the EZ-Impulse™ Noise filter for clicks, the High Pass filter for rumble and the Continuous Noise filter for the hiss. ***Drag these three filters into the signal path*** with the EZ Impulse filter on the left side followed by the High Pass filter in the middle and then the Continuous Noise filter on the rightmost side. Now ***double click on the EZ-Impulse filter icon*** in the signal path of the Multi-Filter. Your screen should look something like this:



Working with the Multi-Filter

To open the individual settings for any filter in the Multi-Filter signal path, just double click on that filter and create your settings. For the Expert Impulse filter, ***drop the preset box and choose "Demo Audio Wave file De-click"***. You could preview and adjust while listening here, of course, but we've already got some good settings available as a preset. ***Close the Expert Impulse filter window***.

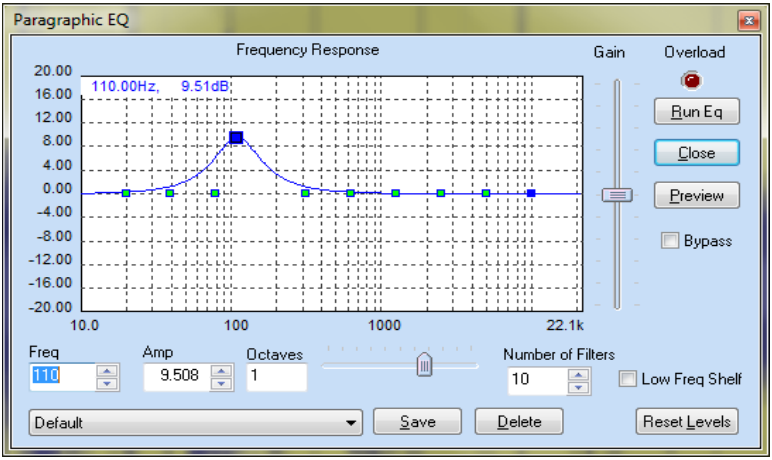
Now ***Double click on the High pass filter and select the "Demo Audio Wave file De-Rumble" preset. Close the High Pass filter window when done.***

Lastly, ***open the Continuous Noise filter and choose the "Demo Audio Wave file DeNoise" preset. Close the window.*** We used these presets only because they are available. You can always choose any preset or you can set the filters yourself by just previewing as we did in the Basic Step by Step guide.

<u>Note:</u> Each individual filter can be previewed even in the Multi-Filter.

Now we have three filters strung together. ***Click the Preview button in the Multi-Filter window***. You will hear the audio being filtered by all three of these filters. ***Click the Bypass button a few times*** to marvel at what is happening here. Feel free to ***giggle*** and then ***cancel the preview***.

Now we're going to add a fourth filter. ***Drag the Paragraphic EQ icon into the signal path*** and put it last in line (on the rightmost side). ***Double click the Paragraphic EQ icon to open its adjustment window***. ***Now again click on the <u>Multi-Filter</u> Preview button***. You may have to move the Paragraphic filter window out of the way to access this button. You are now hearing the audio filtered thru 4 filters. ***Make sure the Bypass button is not checked***. While the preview is happening, find the blue control point in the Paragraphic EQ that is close to 110 Hz. You can see this on the horizontal scale on the bottom of the EQ adjustment area. ***Move this up*** to increase the bass frequencies in our old recording. The more you move it up, the more low frequencies you get. It should look something like this:



Using the Paragraphic EQ

You should notice that you are applying all four filters at once and you are adjusting one of them. The results of your adjustments are instantly heard while all the filters are being applied. ***Cancel the preview, run the filter, and "Save Source As" the file***. Fast Edit mode requires that you save files when you are done.

What you've learned in this Advanced Step-by-Step Guide - - -

You now know how to choose and use Fast Edit Mode, how to bring up and use the Multi-Filter, and how to Preview and adjust filters (or effects) while using the Multi-Filter.

Forensic Audio Step-by-Step Guide (DC Forensics10.xx Only)

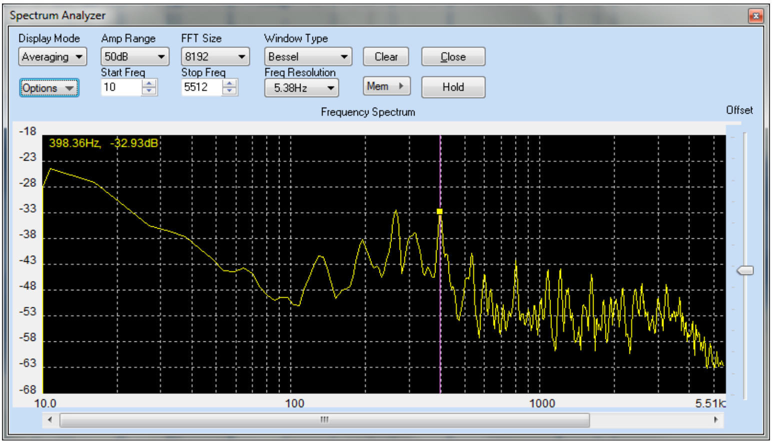
If you are a new user of DC Forensics10, please go through the Basic and Advanced record restoration guides first. We know you won't be restoring records with DC Forensics10, but these guides provide an excellent introduction to the overall flow of the Diamond Cut software programs. We'll assume you have that basic knowledge in preparation for this Forensic Guide tutorial. As before, the specific steps for you to perform are in ***Bold***.

To get started, ***run DC Forensics10 and put it in Classic (Source/Destination)*** mode. ***Now open the file*** called Forensicdemo.wav. Forensics version demo files can be found on the Start Menu.

Play this file. Like a lot of Forensic audio files, the signal to noise ratio is very poor (a low ratio). The noise is much louder than the voice and this makes it difficult to understand what is being said. The speech is masked by a loud buzz and has a high frequency and a low frequency noise throughout the piece. As always we ***start by listening so that we can identify the noise components that make up our problem***.

It's easy to hear in this file that we will need multiple filters to clean this thing up. Therefore, let's use the Multi-Filter. ***Click on Filter/Multi-Filter*** and bring it up and ***clear out any filters that are in the signal path***. The file contains a loud Buzz that almost overwhelms the good audio. To remove the hum and buzz, we will use the Harmonic Reject Filter. It's called this since hum and buzz are rich in harmonics and this filter can remove them. We'll need one of these filters for sure but don't put it in the Multi-Filter yet.

Next, ***listen again***. There are other noises in this piece that you probably can't easily identify by listening. However, we don't only have to listen to the audio, we can also see it. ***Click on View and then choose the Spectrum Analyzer***. ***Set the Spectrum Analyzer to a resolution of 5.38 Hz and 8192 FFT size***. Now ***click on Play*** on the toolbar and you'll see the audio as well as hear it. After listening for a few seconds***, hit the spacebar to stop the playback***. The Spectrum Analyzer continues to show you the frequency spectrum of the audio. Notice that there are some "spikes" of audio in here. One, for example, is at around 950 Hz. You can click on any spot in the display and the Spectrum Analyzer will show you the frequency and amplitude at that spot in the upper left corner of the display. It should look a lot like this:



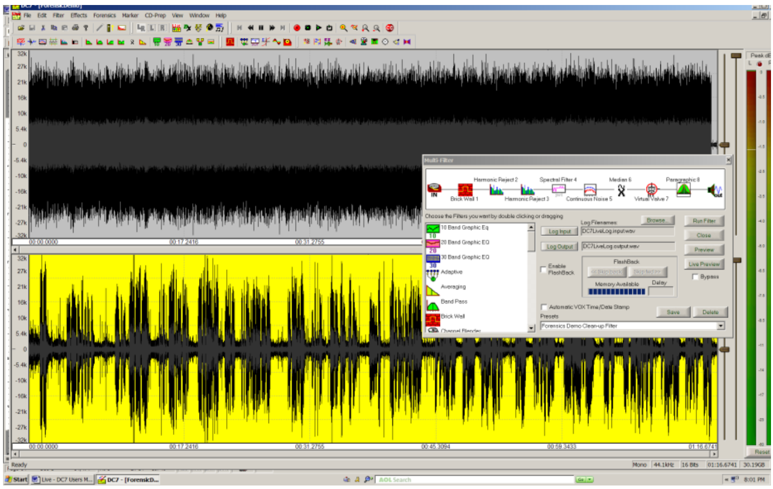
Using the Spectrum Analyzer

If you look closely, you can see that there are actually four spikes and some random noise above 12,000 Hz. We can remove specific frequency ranges with a Notch filter and we can remove that high frequency noise with our Continuous Noise filter or a Low Pass Filter. So now we know all the basics to attack this problem – we'll need a Harmonic Reject filter, 4 Notch filters and a Continuous Noise filter. You can see how we used the tools to come up with a general plan of attack. Now, ***close the Spectrum Analyzer***.

We have already created a Multi-Filter preset designed to clean this file**. *Select the Multi-Filter preset called "Forensic Demo Clean Up Filter".*** Notice that our basic filter is not quite the same as we had set-up in the earlier analysis. We know you want to hear this, so we will explain each component of this Multi-Filter preset after we listen to it in action.

__**Note:**__ This is a computationally challenging filter. Even a pretty fast computer might stutter or pause while trying to do all these calculations in real time. If this happens just stop the preview. ***Hit the Multi-Filter Preview button*** and listen. ***Click bypass in and out a few times*** to get the full effect. ***Cancel the Preview*** when you're ready.

Now ***click on Run Filter and listen to the resultant file*** in the Destination window. This is almost as impressive; check out the before display in the Source window and compare it to the after waveform display in the Destination window. It should look like this:



The Power of the Multi-Filter is Obvious

Notice how the total jumble of noise in the source window has changed to a nice speech pattern in the destination window. You could even make this a bit more "Hi-Fi" if you wanted, but the goal of most Forensic restoration and enhancement projects is to make speech understandable so that it can be transcribed.

Now we will explain the filters we used in this preset. ***Double click each filter*** as we explain it and examine the settings (each of these filters has its own tutorial in Section 2 of this Manual). ***Close each filter*** when you finish looking at the settings.

<u>Brick Wall Filter</u> – this is set to reject all low frequency sound below 135 Hz and all sound above 11 kHz. Remember the noise in the Spectrum Analyzer at 12 kHz and above? We took it out with this filter.

<u>Harmonic Reject Filters</u> – both are set to 60 Hz, which is a common power line induced noise frequency (50 Hz in many countries outside

the US). Why do we use two of them? Because multiple filters of this type will deepen the attenuation of hum and buzz noise. It's like one worker digging a hole two feet deep and then another also digging a hole two feet deep, but the second worker starts in the hole created by the first. You end up with approximately a four foot deep hole.

Spectral Filter – Remember we thought we might need 4 notch filters? We could have done that, but our Spectral Filter allows us to set up as many as 32,000 notch filters at once! Additionally, this filter allows us to either boost or cut any frequency range we want including ones like these that are not related to each other harmonically. This is just a more efficient way to handle several unrelated noises.

Continuous Noise Filter – This filter is set up to aggressively filter just about all noise frequencies within the speech range. Remember, this doesn't take out all sound, but targets noise while leaving behind good audio. It's a good filter to add to the mix.

Median Filter – This was chosen to add intelligibility to the voice. It is good for enhancing muffled speech.

Virtual Valve Amplifier - This will add harmonics to the voice, increasing intelligibility.

Paragraphic EQ – This is set to raise the volume of the speech frequencies.

So you see, we really did follow our initial path by using Harmonic filters, Notch filters and Continuous Noise filters. We ended up adding some additional tools that further enhanced our audio. You will likely do the same thing as you perform your own restorations. This process of first planning a course of action and then making it better is a common one that you will use often.
Close all filters, Make Destination the Source (saving it under a new name), and exit the program.

Live-Mode Step by Step Guide

This guide will introduce you to another major feature of the program. As before, follow along with the included demo file and perform the actions printed in ***Bold*** type. We assume you have mastered the other concepts we introduced in the Basic and Advanced audio restoration Step by Step guides. Please make sure you have completed those earlier guides before attempting this one even if you are a Forensics User. It teaches you the basic flow of the software program.

Live-mode is a powerful feature of DCArt10 / DCForensics10 that allows you to remove noise and enhance audio as it happens. You do not have to record the audio to the hard disk before applying filters. Live Mode is used by TV and Radio broadcast engineers to clean up noisy live feeds for real-time transmission. Forensics users use it to clean up real-time surveillance audio streams. It is also useful for cleaning up radio reception for Hams, SWLs, DXers, and other Forensic applications. Audiophiles sometimes use Live mode to replace the preamplifier and tone controls in their audio system. If you have your computer connected to your home stereo, you can use Live-mode to actually restore your records while you play them. It can also be used “on site” by Forensic users who may need to listen to (and record) surveillance audio as it happens.

To use Live-mode, ***make sure you have your speakers connected to the sound card output and an audio signal connected to the line level input***. This audio signal source can be from a soundboard, mixer, radio receiver (or tuner), turntable (with preamp) or any other signal. For this Step-by-Step guide, we’re going to assume you are using a radio to provide a signal to the program. In this case, you would ***connect the headphone or other audio output of your radio or tuner to the Line Input of your sound card***.

Turn on the radio and you should hear audio from the speakers connected to your computer. This step depends somewhat on the brand and model of sound card that you have, but the great majority of sound cards will simply take whatever audio appears on its input and sends it directly to the output of the sound card. If you don’t hear anything, just keep going in this Guide, as we’ll check everything out a bit later.

Now ***turn off your Radio*** and ***launch the program*** if you haven't already and then ***open the file called "Radiodemo.wav".*** It's in the same directory as the other demo files. Before we do this live, we're going to experiment a bit with a recorded bit of audio.
Once the file is loaded, ***listen to it***. You can hear that there is A LOT of noise in this file. It is understandable, but would not be fun to listen to due to all the noise. Once the file has finished playing, **open the *Multi-Filter*** (under the Filter Menu) ***and choose the preset called "Live Mode Demo Cleanup – SW Radio"***. This preset provides some useful tools for this demonstration file. This preset will also be a good starting point for your own efforts as you use the program in Live mode.

Double click the Continuous Noise Filter to open its adjustment window. You can look at the settings for all the tools in the Multi-Filter preset and you should be able to understand what they are doing based on your learning so far. Remember, you can always open a specific filter and then hit F1 to find out about it. Your screen should look something like this:

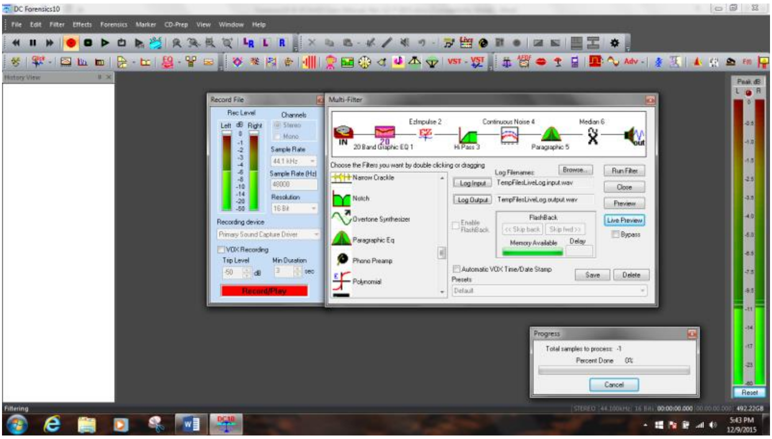
Multiple Screens Shown Working Together

We want to bring your attention specifically to the Continuous Noise filter here because we are dealing with a concept that you might not have considered – How do I use this filter in Live mode when I can't take a sample of the noise? Remember, the audio is happening in real time and is typically not recorded to the disk at all in Live-mode. It'll therefore be impossible under these conditions to actually take a noise sample. Our preset setting here shows how you can deal with this. Notice that the filter is set to provide a fairly large amount of attenuation to the low frequencies and the higher frequencies (remember, the higher the blue square controls are set, the more noise is removed at that frequency). With radio signals, we will seldom care about very low or very high frequencies anyway, so this setting gets rid of a lot of noise without unduly bothering our good signal. The controls between 300 Hz and 3,000 Hz (which are speech frequencies) were then simply adjusted for good results. Go ahead and ***hit the Multi-Filter Preview button now***. ***Click Bypass in and out*** to hear the improvement. (Alternatively, change the CNF mode of the Continuous Noise Filter to Auto Spectrum CNF or AFDF and the filter will find its own noise fingerprint on-the-fly.)

Notice the slider labeled Attenuation. This control determines the overall amount of filtering that is going on for the entire filter. If you move this control upwards, it will filter more overall noise. ***While Previewing, adjust the attenuation slider to find the point you like best***. You'll find that this very weak signal is now quite understandable – if you speak German. The other filters are also adjustable, of course. You'll quickly find that there is a tradeoff here. The more aggressive you make the Continuous Noise filter, the more digital artifacts you will induce. Just adjust it to please your ears.

Now ***stop the Preview*** and let's go into Live-mode. ***Turn on your radio*** so that you can hear the signal in your computer speakers.
Note: If you don't hear anything, continue on anyway as some sound cards don't monitor incoming audio by default. ***Double click the leftmost icon in the Multi-Filter*** so that you can adjust your audio settings. Do that and then ***click on the button labeled "Live Preview"*** and the program will immediately start providing filtered audio to your speakers. You should certainly be hearing audio now.

In the Live-mode, your screen should look something like this figure:



Live-Mode is engaged

Important -- Do you hear two versions of the audio – one a bit later in time than the other - like an echo effect? If so, you need to turn down the input of your sound card. If you have a standard sound card, just double click the speaker icon in your tray and choose "Options" then "Properties". Now click on "Recording" and then click "OK". Now move the slider labeled Line Input or Line down to the bottom. The echo should go away! If you have a more sophisticated sound card, refer to its documentation to find out how to turn off monitoring or turn down the Line Input control.

If you have the DC Forensics10 version of the program, you cannot only process audio in real time, but you can record it as well. Just click the Log to Disk button. Additionally, you can also select VOX recording (the program starts and stops recording automatically with the signal that comes in) and have the program mark the segments with the date and time of their occurrence.

Here's a tip. You can now add or change the filters as you please. A good one for radio work is the Dynamic Processor using the ALC mode. This can help a lot with atmospheric fading in a SW (**S**hort **W**ave) signal as it will even out the audio level in real time.

What you have learned:

In this Step-by-Step Guide, you've learned how to use the Multi-Filter in Live mode to process audio as it happens. You've learned how to adjust your sound card for Live-mode – remember, however, that some low quality sound cards cannot record and play at the same time and therefore can't be used in Live Mode. You've also learned how some filters can be used to make radio reception much better.

The Wizard

(For Record Restoration)

Introduction to the Record Restoration Wizard

A 'Wizard" (sometimes referred to as a "System Navigator") for record restoration can be found under the Task Pane section of your software (found on the left side of the program screen). It consists of a logical process matrix so it can be used to guide you through the various steps of a record restoration. Each step in the process will ask you some questions, the answers to which will bring up the appropriate filter (or filter chain). This will be your starting point for each step and each step must be performed in order. Often, you will find that the starting point is the finishing point with no need for adjustment of the filter parameters; that is for you to decide based on your specific situation and/or taste by previewing the filter via a decent quality audio system.

To begin, set up your turntable properly with a stereo magnetic phono cartridge equipped with the proper stylus for the record to be transferred. (A stylus chart can be found in the Charts, Graphs, and other Info section of this documentation.) Make sure that the speed is correctly set (which can be verified by one of the strobe discs provided in the software). Feed the turntable output cables into either an RIAA or a Flat phono preamplifier magnetic phono input. Connect the output of the phono preamp into the line level input of your soundcard. (Make sure that your turntable ground-strap is connected to the preamp chassis to minimize ground-loop hum in the transfer.) Transfer your record using the Diamond Cut record function using at least 44.1 kHz sampling rate and at least 16 bits of resolution. Some files may benefit in improved impulse filter performance by using 48 kHz or higher sampling rates with the law of diminishing returns kicking in at values above 96 kHz. After you have “Previewed” and adjusted the parameters to your personal liking for each step (in the process), you should “Run” the filter, thereby getting your file ready for the next step by way of the Wizard.

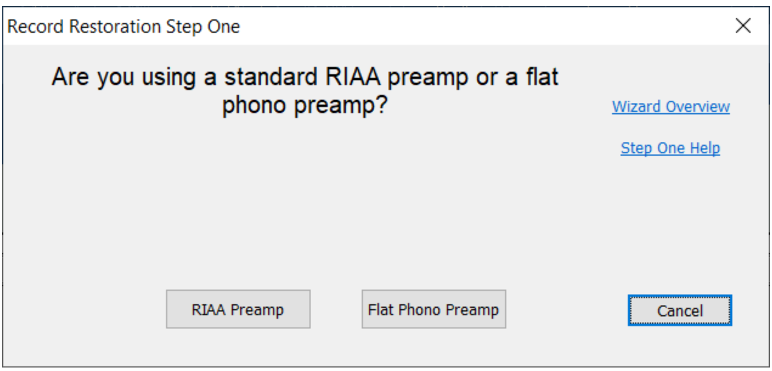
Step 1: Record EQ Optimization & Impulse Noise Reduction

The first step in the record restoration process aggregates several processes into one action. Each step in the record restoration process includes a description of the type of record to be restored. Step 1 will also provide the proper EQ for your record type. Step 1 will then de-click it (reduce impulse noise) via various factory presets. It facilitates both RIAA and Flat preamp transfers, which is your choice to select. When an RIAA preamp is used during the record transfer process (the most common type and found on most home audio receivers), it will re-EQ your file if necessary and then create the proper EQ for your record. Similarly, if a Flat preamp is used, it will apply the proper record EQ to your file (flat preamps are usually small stand-alone devices). Step 1 will also provide the proper bandwidth for your material preparing it for the next step (Step 2 which will involve reduction of continuous noise via the CNF). The bandwidth preparation process reduces the rumble associated with your record type as well as out-of-band very high frequency noise. It is important to note that distortion can occur due to excessive volume settings in the Virtual Phono Preamp sub-filter. If this occurs, just click on the VPP (the first filter in the line-up) and adjust the volume control downwards until the distortion disappears while previewing the multifilter.

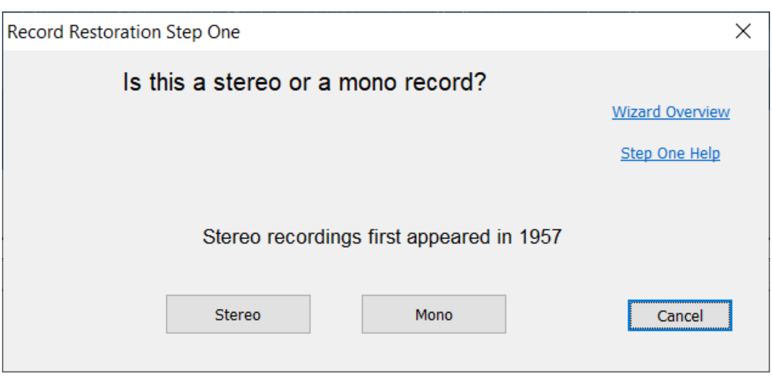
Wizard Use Example via one of the Demo Wavefiles:

Now, let’s go through an example of the use of Step 1 of the record restoration wizard. We will be using one of the demo wavefiles found under the /file/open demo wavefiles menu string. This example is representative of one of many different types of record restoration situations that can exist; your situation will vary depending on your source material and the hardware you used to make the analog to digital transfer. Click on this file to open it in the Source Window.

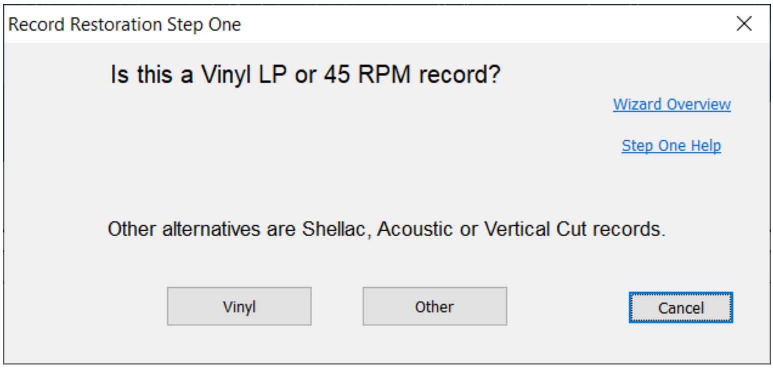
Navigate down to the file which is called “Preamp Vertical Cut Demo” and click on it. The file will appear in the software time display work area. Now, locate the task panes on the left side of the program. Under the task panes, you will see a section called “Restoration Tasks”. Navigate down to the item called “Step 1 and click on it. The following dialog boxes will appear you will need to answer the questions appropriately for your record restoration situation. Note that you can jump over to the helpfile for more detailed information by way of the blue colored links located on the left side of each box.



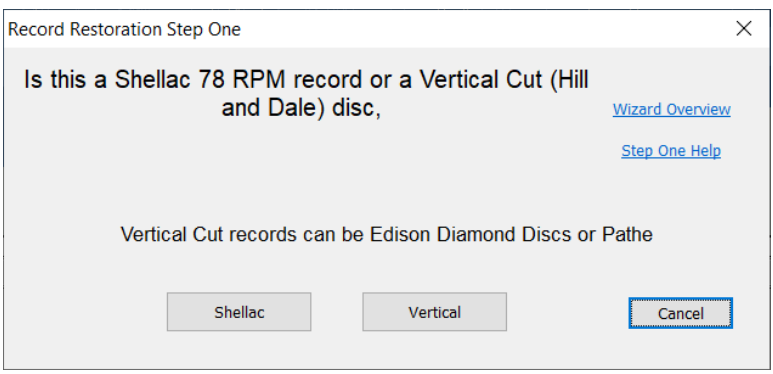
Choose Flat Phono Preamp.



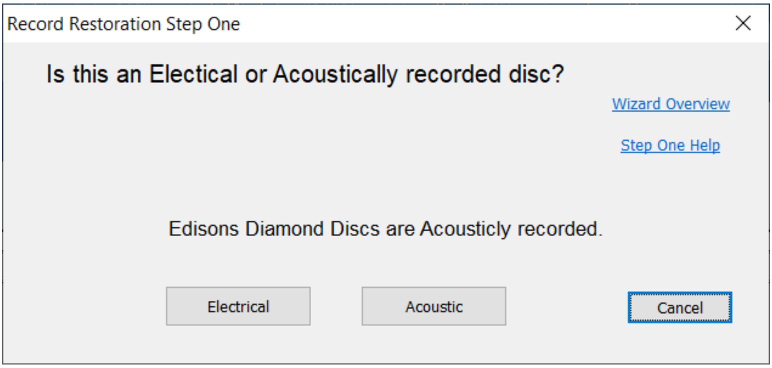
Choose Mono.



Choose Other.



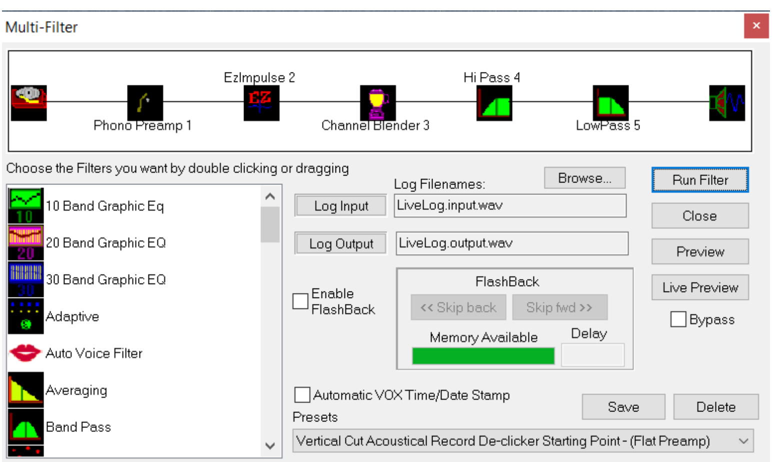
Choose Vertical.



Choose Acoustic.

Example: Details regarding Step 1

This demo wavefile was transferred using a flat phono preamp (via a stereo phono cartridge. After all the questions are answered, a multifilter line-up will appear and will look like this:



Example Resultant Filter setup for Step 1

Note that it includes multiple filters in a chain consisting of the following (in order). Each sub-filter has been tuned by the factory to provide you will a good starting point for your project; you can tweak any of the parameters within any of the 5 sub-filters if required or desired.

Phono Preamp 1: This sub filter resolves the proper signal vector from the 2-channel {stereo} phono cartridge and also provides the proper equalization for the type of record being restored through a process called “re-equalization”.
Note: If you are working with early LPs, and you want to tweak the sound, you can chose between a wide varieties of alternate EQ curves as found under the presets area of the filter.

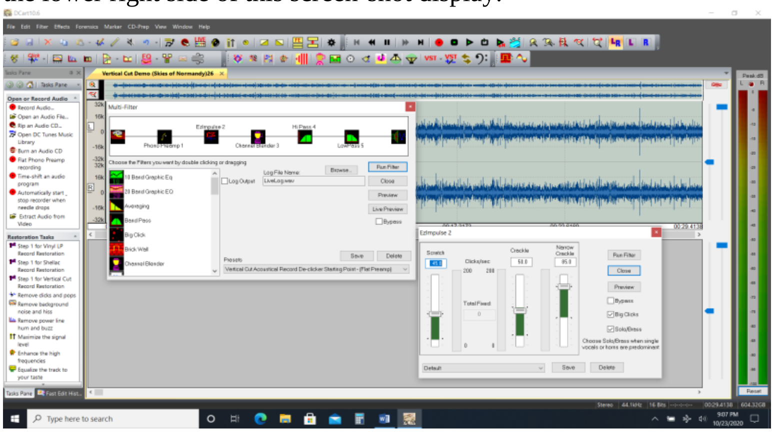
EZ Impulse Filter 2: This sub-system reduces impulse noise like ticks, clicks & pops.
Note: If you want to tweak the system, the Scratch controls the effect on loud scratches. The Crackle controls the effect on Crackle sounds and the Narrow Crackle controls the effect on very small ticks.

Channel Blender 3: It blends the 2 channels together appropriately based on record type.
Note: Early Stereo LPs often applied too much separation producing a sort of “ping-pong” effect between the two channels. The channel blender can be used to produce a more natural blend of the channels by reducing the channel separation.

Hi Pass 4: This sub-filter educes turntable or master cutting lathe rumble.
Note: You can tweak the frequency and/or slope of this sub-filter to yield the optimal balance between the bass notes and turntable rumble. This is best tuned with an audio system that has a sub-woofer.

Low Pass 5: The Low Pass Filter reduces out of band high frequency noise.
Note: This can be used to attenuate the high frequency noise heard on 78s that is beyond the capability of the media; adjust for the best balance between the upper registers of the audio spectrum and the random hiss noise.

Preview the multi-filter setup to hear if it is producing the results that you desire. Often, it will produce decent results with no further sub-filter adjustments. However, you can click on any of the sub-filters and tweak the system to your taste. Probably the sub-filter of greatest impact on the sound quality will be the EZ Impulse filter. If you click on the EZ Impulse sub-filter, it will present itself to you as follows in the lower right side of this screen-shot display:



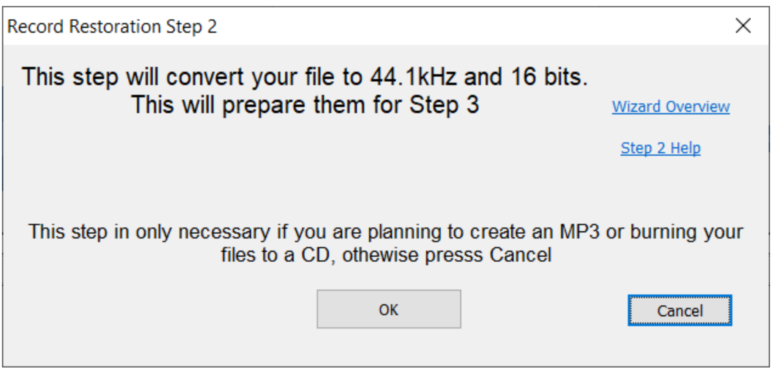
The Record Restoration Wizard with the EZ Impulse Filter Open

Please refer to the section of this User's Manual that describes in detail the specifics pertaining to this and all of the other sub-filters used in this filter line-up. After you are satisfied with the sound as heard in the Mulltifilter "Preview" mode, click on the "Run Filter" button (found in the upper right corner of the Multifilter). Your computer will then process your file. If there are any errant impulses in the file, you can highlight them with your mouse and then hit the "I" key which will produce a manual interpolation at that location.

If you are using the Diamond Cut Productions default editing mode (Fast Edit Mode) the changes will be applied to the opened file directly (which is un-doable if necessary). If you are using the Diamond Cut Productions Classic Edit Mode (Source and Destination workspaces) your processed results will end up in the destination window. You can use the File menu item called "Make Destination the Source" or the "Save File As" commands to preserve the processed results. In either case, you are now ready to proceed to Step 2 in the record restoration process.

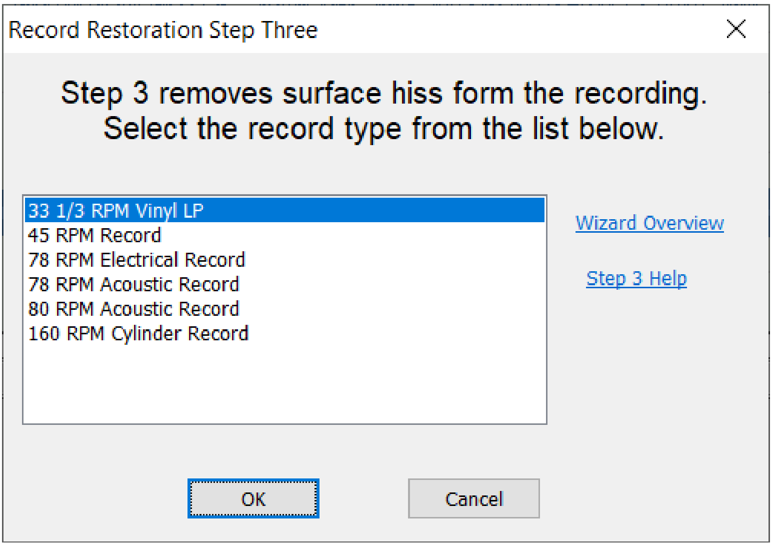
Step 2 – File Format Optimization

Step 2 will convert the sample rate and/or bit depth (resolution) to 44.1 kHz and 16 bits if you plan to create a CD or an .mp3 as your final product. You can chose not to convert it if you plan to leave your restoration in its native .wav format. Just check OK if that is your preference (no MP3 or CD anticipated).



Record Restoration Step 2 Dialog Box

Step 3 – Hiss & Continuous Noise Reduction



Step 3 will attenuate continuous noises that remain in your file via the Continuous Noise Filter (CNF). Numerous noise fingerprints (or noise signatures) are pre-programmed into the system based on record type. The Wizard will ask you a series of questions which will then bring up the appropriate CNF settings with a good starting point set of noise contours and filter parameters. Preview the file using this filter that was the result of the previous restoration steps. You can tweak the effect of the CNF per your taste, with the attenuation control having the greatest impact on your results. Higher settings of attenuation produce greater degrees of noise reduction, but when set too high will introduce artifacts into your file. Experiment with this control to obtain the best balance between noise reduction and artifact production. When you are satisfied with the CNF settings using "Preview" mode, run the filter. Depending on how you answer the questions for Step 3, one of the following presets will be selected from the CNF Presets listing:

33 RPM LP Starting Point
45 RPM Starting Point
78 RPM Acoustical Starting Point
78 RPM Electrical Starting Point
80 RPM Vertical Cut Starting Point
160 RPM Vertical Cut Cylinder Starting Point
*General Purpose (Flatline Response) Starting Point

Note: Alternatively, these presets can be accessed directly via the CNF presets selection field.

*Note: This preset is not available via the Wizard, but can be found directly in the CNF itself. It is good for unusual record types, like acetates.

Steps N on forward – Add your touch and Post-Processing

Steps N on forward involve further file restoration/enhancement and are where your artistic talent and personal taste come into play. At this point, you can enhance the file to your taste using things like any of the various graphic or paragraphic equalizers. You can add tube sound with various tube types and/or amplifier configurations using the Virtual Valve Amplifier. The Virtual Valve Amplifier also includes a harmonic exciter mode. You can add sub-harmonic with the sub-harmonic synthesizer or overtones with the overtone synthesizer. The Dynamic Noise filter can be used to dynamically enhance to top-end and the Dynamic processor or the Dynamics Processor and the Punch and Crunch processor can be used to compress or expand the dynamic range of your file. You can add reverb and/or echo or time offset between the channels. You can even make a stereo file out of a mono file via various pseudo stereo techniques that the software provides you with. And the list goes on and on and the possibilities are almost without limit. It is recommended that you consult with the sections of the manual pertaining to any of these special effects by searching the helpfile “search for” feature.

After you have added your personal touch to the recording, use the features found in the CD Prep menu to normalize the gain of your file to take optimal advantage of the dynamic range of digital recording systems. The edit/mute feature can be useful to create total silence between tracks). Then, you can find and mark silent passages. Quantize for CD audio is a useful feature to invoke at this point if you are making a CD where one track runs into the next (gapless recording). Next, break the larger file into pieces (tracks). A post processing step could include “fade-in” and “fade-out” of each track (found under the edit menu).

It is good practice to set up an archive database for the file at this point in the process and to save it in its native format for future reference.

Then, you are ready to create your final product, be it a CD, and .mp3 or any other of numerous file formats that are supported by your Diamond Cut Productions software by way of the file/save as function.

Which Tool Do I Use?

Probably the most frequently asked question we get each day is “How do I reduce the level of a certain type of noise?” Certainly, the more you use this product, the more proficient you’ll become at picking the right tool to match the noise you’re trying to remove. But for now, if you’re new to the game, we’ve included this handy chart that may help you get a jump on finding the tools that match your problem areas.

Filter Finder

Sound Restoration Category	**Sound Defect or Task to be Performed**	**Filter Type**
Early Acoustical Cylinders and Discs	"Pops"	Impulse Noise (EZ-Impulse™ or Expert Impulse Filter)
	"Crackle"	Average, Median or Continuous Noise Filter
	"Distortion"	Low Pass Filter or De-Esser in the Dynamics Processor
	"Hiss"	Dynamic Noise or Continuous Noise

		Filter
	"Rumble"	Highpass or Continuous Noise Filter
	"Thin Sound"	Graphic Equalizer
	"Reverse Skip"	Cut
	"Forward Skip"	Copy and Paste Insert
	Skip / Miss-tracking	Speed Change Filter / Fractional Speed Mastering
	Cracked Record	Big Click Filter
	"Thumps"	Selective use of the High Pass Filter (set to 6 dB/Octave)
	"Off Pitch"	Speed Change Filter
LPs & 45 RPM Records	"Ticks"	Impulse Noise or EZ Impulse™
	"Pops"	Impulse Noise or EZ Impulse™
	"Distortion"	Low Pass Filter or CNF in Artifact Suppression Mode
	"Rumble"	High Pass Filter
	"Shrill"	Graphic Equalizer
	"Reverse Skip"	Cut
	"Forward Skip"	Copy & Paste Insert
	Noise between Cuts or Tracks	Dynamics Processor – Expander / Gate
	Muddy Bass on Stereo Records	Channel Blender
	Stereo "Ping-pong" effect	Channel Blender
	"Off Pitch"	Speed Change Filter
Magnetic Tape Recording	"Hiss"	Dynamic Noise Filter or Continuous Noise Filter
	"Highs Loss"	Time Offset (azimuth correction)
	"Smeared" Stereo Image	Time Offset (azimuth correction)
	Clipping Distortion	De-Clipper
	"Off Pitch"	Speed Change Filter
AM Radio, Short Wave or Transceiver Radio	"Hiss"	Dynamic Noise Filter or Continuous Noise Filter
	"Static"	Impulse Noise (EZ-Impulse™ or Expert Impulse Filter)
	Line Frequency "Buzz"	Harmonic Reject Filter
	Volume Fading	Dynamics Processor ALC
	Heterodyning "Whistle"	Notch Filter or Multiple Notch Filters
AM Broadcast	"Whistle"	Notch Filter (Europe - 9 kHz) (US - 10 kHz)
	Line Frequency Buzz	Harmonic Reject Filter
	Volume Fading	Dynamics Processor ALC
FM Stereo Broadcast	Multi-path Distortion	Dynamics Processor / De-Esser or Channel Blender

	Multiplex Noise	Channel Blender
	Ignition Noise / Static	Impulse Noise Filter
	"Feedback"	Notch Filter
	"Hum"	Notch Filter
	"Mic 'P' Pop"	Hi-Pass Filter
	"Dead"	Dynamic Noise Filter (spectral enhancement mode)
	‘Digital Sound”	Virtual Valve Amplifier / Tube Warmth
	Street Noise	CNF Spectral Subtraction
	HVAC Noise	CNF Spectral Subtraction
	Hum	Notch Filter
	Buzz	Harmonic Reject Filter or EZ Impulse
	Sibilant “Ess”	De-Esser
	Digital Clipping	Limiter Preset in the Dynamics Processor
Telephone Conversation	"Intelligibility"	Bandpass Filter
	“Noisy - Random Out of Band”	Brick Wall Filter in Bandpass mode or Band Pass Filter in Chebyshev Mode
	"Noisy – In Band"	Continuous Noise Filter
	"Muffled or Garbled"	Median Filter or Spectral Inverse Filter
	Variation in loudness between parties (near party/ far party gain compensation)	Dynamics processor / Compressor / ALC Punch and Crunch in ALC mode
	Cell Phone Noise Interference	Cell Phone Noise Filter
Surveillance Recording	Cancellation of Radio / or TV using a reference track	File Conversions (Left – Right Mode) Adaptive Filter in Reference Mode Dynamic Spectral Subtraction filter (DSS)
	Automatic Voice Activated Recording	Multi-Filter operating in VOX mode
	Echo Reduction	Dynamics Processor in Expander Mode (use Echo or Reverb Presets)
	Reverberation Reduction	Dynamics Processor in Expander Mode (use Echo or Reverb Presets)
	Automatic Time Activated Recording	Timer Recording Function
	Noise isolation & identification Poor Speech Articulation	Slot Filter Median Filter w/ Weighting
	Varying Random Noise Profile	Adaptive Filter
	Varying Coherent Noise Profile like music playing behind speech	Continuous Noise Filter in DSS Mode with reference track

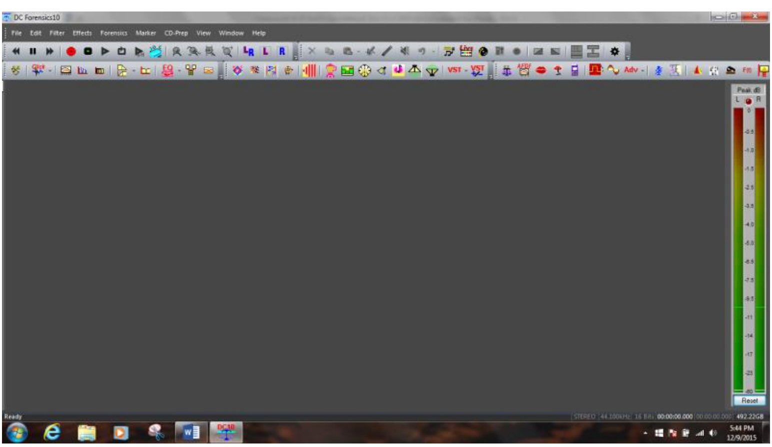
	DTMF Tone Amplification	DTMF Presets found in the Paragraphic EQ
	DTMF Tone Identification	Spectrum Analyzer
	Voice Masking	Speed Change + Stretch & Squish Filters
	Voice Disguising	Voice Garbler
Forensics Audio Analysis & Enhancement	Gunshot Ballistic Timing	Markers and Time Display / Spectrum Analyzer
	Gunshot Ballistic Identification Matching	Time Display and Spectrogram
	Voice Identification	Spectrogram or Voice ID
	Analog Tape Deck Azimuth Correction	Time Offset Adjustment in the File Converter
	Person Talking too fast for Transcription Poor Intelligibility	Stretch & Squish EZ Forensics Spectral (Inverse) Filter
	Voice Pitch too High	Speed Change Filter
	Tape Authenticity	Spectrum Analyzer
	Reference Channel Acoustical Compensation	Bandpass Filter plus 20 Band Equalizer plus Reverb
	Poor Intelligibility	20 Band Graphic EQ
	Adaptive Noise Rejection	AFDF in the Continuous Noise Filter Adaptive Filter under Forensics Menu (TDAF)
	Very weak party or sounds in the background	Whisper Enhancer (Forensics Menu – Forensics Version only)
	Narrowband Noise Rejection De-Muffling	30 Band EQ Spectral Inverse Filter or Overtone Synthesizer
Optical Movie Soundtracks	"Pops"	Impulse Noise
	"Crackle"	Median or Crackle Filters
	"Thuds"	Highpass Filter
	"Hollow"	Graphic Equalizer
	“Film Flicker”	Harmonic Reject Filter
	Reverberation	Dynamics Processor in Expander mode (use the Echo or Reverb presets)
Television / Video	Vertical Sync Pulse Bleed-through buzz	Harmonic Reject Filter – 30 Hz United States 25 Hz Europe
	Horizontal Sync Pulse Bleed-through	Low Pass Filter

Any Sound Source	Mike “P” Pop	Highpass Filter selectively applied
	Low Gain or Volume	Gain Change (under Edit menu)
	Acoustical Feedback Clipping Distortion	Notch or Harmonic Reject Filter De-Clipping Filter or Manual De-Clipping Process or Lowpass Filter selectively applied or Impulse Filter
	De-Ess (excessive sibilance of the pronunciation of the letter “S.”)	Lowpass Filter selectively applied, or use the Dynamic Processor/ De-Esser
	Pitch incorrect	Change Speed Filter
	Wind Noise	Wind Noise Filter
	Whisper among louder noise	Whisper Enhancer
	Line Frequency “Buzz”	Harmonic Reject Filter or Narrow Crackle Filter
	Excessive Dynamic Range	Dynamics Processor or Punch & Crunch
	Lacking Dynamic Range	Dynamics Processor or Punch & Crunch
	Top Octave missing	Virtual Valve Amplifier / Harmonic Exciter
	Top Octave Weak	Dynamic Noise Filter Enhancer Mode
	Bottom Octaves Missing	Sub-harmonic Synthesizer
	Recording lacks “warmth” Top Octave Missing	Virtual Valve Amplifier Overtone Synthesizer
	Too much Reverb	Continuous Noise filter/Channel Blender (Blend to Mono)
	Weak Vocal	Gain Change selectively applied
	No Ambience	Reverb
	Digital Clipping	De-Clipper
	Analog Clipping	De-Clipper
	Harmonic Distortion	Polynomial Filter / De-Esser
	Inter-modulation Distortion	Polynomial Filter
	Multi-Tone Whistle	Spectral Filter or Multiple Notch Filters
	Signal Over modulates	Limiter Presets in Dynamic Processor
	Noise Between Tracks or Rides	Noise Gate Presets in Dynamics Processor
	Automatic Noise Reduction	EZ Clean™ Filter
	Sterile Bass Sound	VVA Fat Bass
	Medium Resolution Frequency Contouring	20-Band Graphic EQ
	High Resolution Frequency Contouring	30-Band Graphic EQ
	Normalize Loudness Between	Auto Leveling in the Batch File

	Multiple Wave files	Processor
	EQ Matching	Spectral Filter operating in Spectral Difference Mode
	Occasional Spurious Noises	Manual Interpolate using the “I” key or “O” key
	Line Frequency “Buzz”	Harmonic Reject Filter or Impulse Filter
	Turntable Rumble	Dynamic Bass Processor
	HVAC Random Noise	Dynamic Bass Processor
	Bass Enhancement	Dynamic Bass Processor

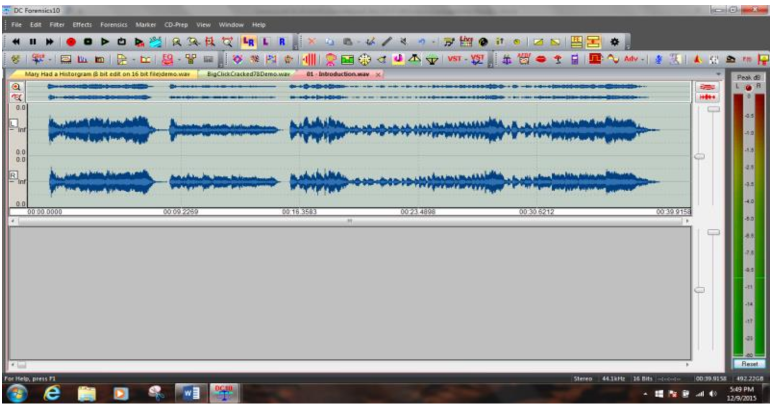
Section 2 – The System and Its Operation

In this section we’ll discuss everything you’ll see when you open DCArt10/DC Forensics10 for the first time. These items are arranged in their Menu order…meaning that we start at the left, open the first menu and proceed from there. We’ll also discuss the various ways to access these menu items. In DCArt10/DC Forensics10, many items in the Menu bar can be accessed via Hot Keys and icons on the various Tool Bars.



DCArt10/DC Forensics10 before the Source File is opened

Important Note: If you see a Tool or a Menu item listed here but don't see it in the program, keep in mind that many items are grayed out or not available until an audio file has either been recorded or opened within the program (.wav, .mp3, .wma, etc).



Audio file is opened in Classic Edit Mode

The File Menu

Open Source

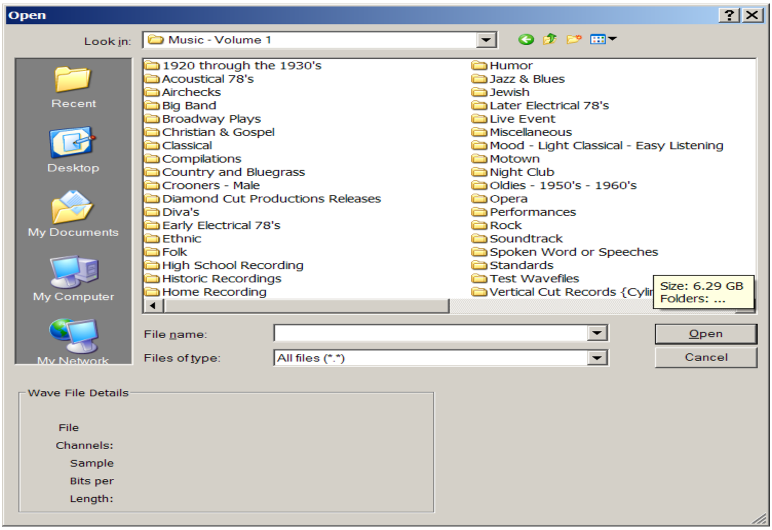
Data in the following file formats can be opened:

- Wave Files (*.wav) †
- BWF (Broadcast *.wav)
- Vorbis (Ogg Vorbis) (Lossy Compression) (*.ogg, *.oga)
- WMA Compressed Audio (*.wma) {does not support files encoded with DRM (Digital Rights Management)}
- MP3 Compressed Files (*.mp3)
- FLAC Files (*.flac) (Non-Lossy Compression)
- AIFF Files (*.aif, *.aiff)
- AAC Files (*.aac, *.m4a, etc)

- Video Files (*.avi, *.asf, *.mpg, *.mpeg)
- All Files (*.*)

† The following compressed Wave File formats are also supported:

- A-Law
- Mu-Law
- ADPCM
- GSM 6.10



Open Dialog Box

This command opens the desired audio file on which you will perform the DCArt10/DC Forensics10 processes. The file will be displayed as a periodic sampling of the file's peak amplitude envelope vs. time in the Source graphical workspace. Please note that the software allows you to load .wav, MP3, .wma, .aif and .aiff files as well as A-Law and Mu-Law compressed files and automatically converts them to .wav files upon opening. If you click once on a prospective file, the Wave File details will be displayed (File Size, Channels, Sample Rate, Bits per Sample, Time Length of File). This function can also be activated by using the file folder icon in the File Tool bar in the upper left hand corner of the workspace or by using the **Ctrl + O** hot key.

Variable Bit Rate (VBR) MP3 File Support

MP3 files of Constant Bit Rate (CBR) and Variable Bit Rate (VBR) are supported automatically within DCArt10/DC Forensics10. They simply open with no additional input from the user.

Convert Various Formats to .wav (Tutorial)

DCArt10/DC Forensics10 provides you with a means for converting various file formats into .wav extension files for editing. You can perform your signal processing after the conversion to .wav and then re-convert back to your favorite format with an appropriate encoder if necessary. The MP3 decoder supports Blade, Lame, Fraunhofer, and many other formats. Also, it can handle formats with or without an ID3 tag and with either fixed or variable bit rates. The process of converting from an MP3 (*.mp3), AIFF or A-Law and Mu-Law to wave (*.wav) is extremely simple:

1. Click on the File menu with the left mouse button.
2. Click on the “Open Source”.
3. Under the “Files of Type” selector box, find your file type and click on it.
4. Use the “Look In” box to find your file, select it and Click on “Open”.
5. The file conversion will begin.
6. After a period of time, a waveform will appear in the Source Workspace. This is a 16 bit, converted .wav file representation of the compressed file. The original file determines the sample rate and number of channels.
7. It will have the same name as the original file except it will have the .wav extension.
8. If there is already an existing .wav with the same name, a number will be added to the end of the name to distinguish it.

Note 1: No actual editing takes place on the original compressed file…only the subsequent .wav file conversion.

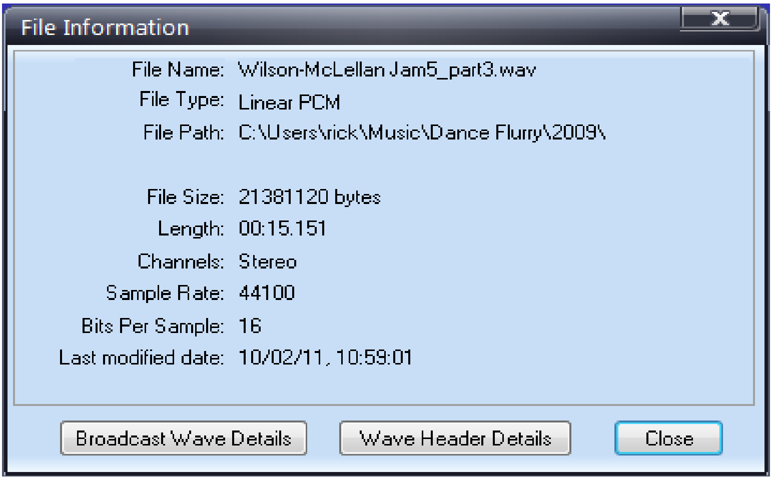
Note 2: Drag and Drop is supported for this feature.

Large File Conversion to .wav (Expanded File Conversion)

Files in the Diamond Cut native format (.wav) are limited in size to 2.14 Gbytes which is a Windows operating system constraint. At a 44.1 kHz sample rate with 2 channels (stereo) and 16 bit resolution, this represents a time limit of roughly 3 Hours and 22 Minutes. If you are attempting to open a lossy (compressed) file (such as an .mp3*) that requires more than the 2.14 GByte .wav file size limit, your Diamond Cut software will deal with the situation automatically by creating multiple files. When this occurs, the first file created will be in the standard .wav format. Subsequent files will be of the Broadcast Wave Format (BWF) which is an extension of the .wav format. These expanded files are deposited in the same folder that held the original source file and they will have a sample rate of 44.1 kHz with a resolution of 16 bits. They will be named the same as the original file but having a _partXX added to the file name, where XX starts at 2 and continues up from that number. The Broadcast Wave format contains a field in its header which includes a starting time. When opening these files, they will have a start time that is correct with respect to the original lossy file; the subsequent Broadcast Wave files will not start at a time value of 00:00:00. If you want to edit time values that you have already created, you can view the time offset value in the Broadcast Wave header. To see that menu, it can be found at:

View->File Info

Next, choose the "Broadcast Wave Detail" button. You can change the header to adjust the offset time value of these files as needed. Note that this value is given (within the Broadcast Wave Dialog Box) in samples wherein the value is essentially the sample rate (44,100 samples per second) times the number of channels (1 for mono or 2 for stereo) times 2 bytes per word (for 16 bit files). If you change the header start time value, you must then press the "Update File" button for it to stick. To write the wave header information back onto the file, you will have to close and then re-open the file to get the software to recognize the new time values.



The File Information Dialog Box

*Note: Only the .mp3 lossy format is supported by the Diamond Cut Expanded File Conversion feature at this time.

Drag and Drop File Support

DCArt10/DC Forensics10 fully supports drag and drop file opening. You can drag a file onto the open program window or you can drag a sound file icon right on top of the DCArt10/DC Forensics10 Icon on your desktop.

Video/Audio Extraction System

Extract audio from Video and Edit Audio with Diamond Cut Software

Your Diamond Cut Audio Restoration software can be quite useful for repairing flaws on the audio tracks of movies or videos. The Forensics version can also be very useful for audio track noise reduction, enhancement, signal analysis, and authentication. DCArt10/DC Forensics10 provides the ability to extract the audio portion of a video having the following file wrappers (depending on your operating system):

3GP, Advanced streaming Format (ASF), Audio Data Transport Stream (ADTS), AVI, MPEG-4, Synchronized Accessible Media Interchange (SAMI)

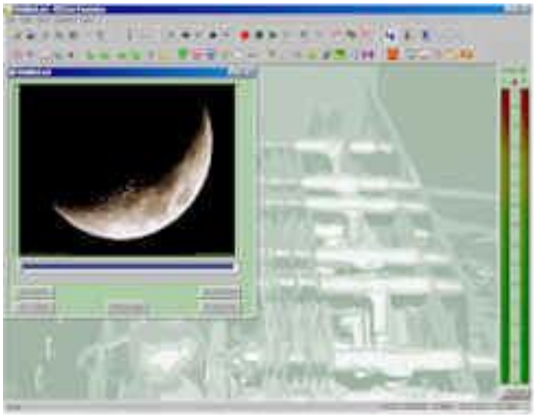
The following file extensions are supported depending on your operating system and the video file format presented to the Diamond Cut Software:

*.avi *.aac *.adts *.asf *.mpg *.m4a *.mpeg *.mp4 *.mov
*.3GP *.wma *.wmv *.sami, *.smi

Video File Container Chart

Format	**File Extensions**	**Media Source**
3GP	.3g2, .3gp, .3gp2, .3gpp	MPEG-4 File Source
Advanced Streaming Format (ASF)	.asf, .wma, .wmv	ASF Media Source
Audio Data Transport Stream (ADTS).	.aac, .adts	ADTS File Source
AVI	.avi	AVI File Source
MPEG-4	.m4a, .m4v, .mov, .mp4	MPEG-4 File Source
Synchronized Accessible Media Interchange (SAMI)	.sami, .smi	SAMI Media Source

This audio extraction system converts the audio on these video formats into the .wav format. The "Extract Audio from Video" System is found under the File Menu. To use it, just click on "Extract Audio from Video" / "Video Files" and then navigate to your video file of interest. The extracted audio will appear in your time domain window of the software program after a processing time period.



A Video Extraction Window

To hear the audio, click the Play button found on the Diamond Cut software toolbar (just like you would to play any audio file). You will hear the audio via your soundcard and audio system. To stop or play the extracted audio, use the same buttons that you would use on regular audio files. The scrollbar at the bottom of the display window indicates where you are in the process of the playing of the video relative to its total length. If you select the drop down "Time Display" found under the View Menu, you can get more specific timing information regarding your Video file. There are several important controls below the scroll-bar, which are as follows:

- **Set Start Point -** Slide the progress bar pointer to the desired Start Point and click on this button. A marker will be dropped at the corresponding position.

- **Go To Start** - This moves the play pointer to the designated start position of the video.

- **Set End Point –** Slide the progress bar pointer to the desired End Point and click on this button. A marker will be dropped in the corresponding position.

- **Go to the End –** This moves the play pointer to the designated end position of the video.

- **Extract Audio –** This button is used to extract the audio from the video between your Start and End markers.

- **Progress Bar –** This is found on the lower status bar and indicates the progress of the Audio Extraction process.

- **Cancel Button –** This button is located directly to the right of the Progress Bar. Depressing it will cease the audio extraction process.

To operate the Video/Audio Extraction system, simply browse to the video file of interest and click on it. Set the desired start and end points with the play pointer. Click on the "Extract Audio" button. The progress bar will start to increase its value. When completed, the converted .wav file will be displayed in the Source Display window. *(Some of these features are only supported on some video formats.)*

<u>Note 1:</u> The following keyboard keys perform some special functions when operating the Video/Audio Extraction system:

- *Home – Go to the start of the file*
- *End – Go to the end of the file*
- *Right arrow – Go forward 1 second*
- *Left arrow – Go back 1 second*
- *Shift + Right arrow – Go forward 1 frame*
- *Shift + Left arrow – Go back 1 frame*

<u>Note 2:</u> The Maximize button on the Video/Audio Extraction system is useful for increasing the size of the video frame so that it can be viewed in better detail. This is the square button in the upper right hand corner of the features display.

Save Source

This function saves the file to its current directory with its current name, etc. This menu item is only active in Fast Edit mode. In Classic Edit mode, files are automatically saved as you work on them.

Save Source As

"Save Source As" saves your Source file as it is, or assigns it a new name, sample rate, dithering technique, create .mp3s, .wma (or other audio file formats). It can also be used to assign your file to a new directory location.

Your data can be saved in any of the following file formats:

- Wave Files (*.wav)
- BWF (Broadcast .wav)
- Vorbis (Ogg Vorbis) (Lossy Compression (*.ogg, *.oga)†
- WMA (compressed audio) Files
- MP3 Files (*.mp3) (provided you have a third party encoder installed)
- AIFF Files (*.aif)
- FLAC Files (*.flac) (Non-Lossy Compression)
- Compressed Formats (depending on what other Codecs are available in your system)
- AAC Files (*m4a, *.aac, etc.)
- Text CSV Files שׁ {Comma Separated Values} (*.txt)
- All Files (*.*)

שׁ Note: An Example of a set of CSVs (.csv) for a stereo file could appear like this consisting only of the amplitude values of a file:

- 0.115784,0.115784
- 0.115784,0.122375
- 0.122375,0.122375
- 0.122375,0.126953
- 0.126953,0.126953
- 0.126953,0.126984
- 0.126984,0.126984

Note: A dialog box will appear when you attempt to save your file in a compressed audio format such as .mp3 or .wma. From there, you can choose from a variety of compressed audio parameters.

†Note: For more information pertaining to Ogg Vorbis Lossy Compression support, please refer to the Ogg Vorbis Tutorial Section of this User's Manual.

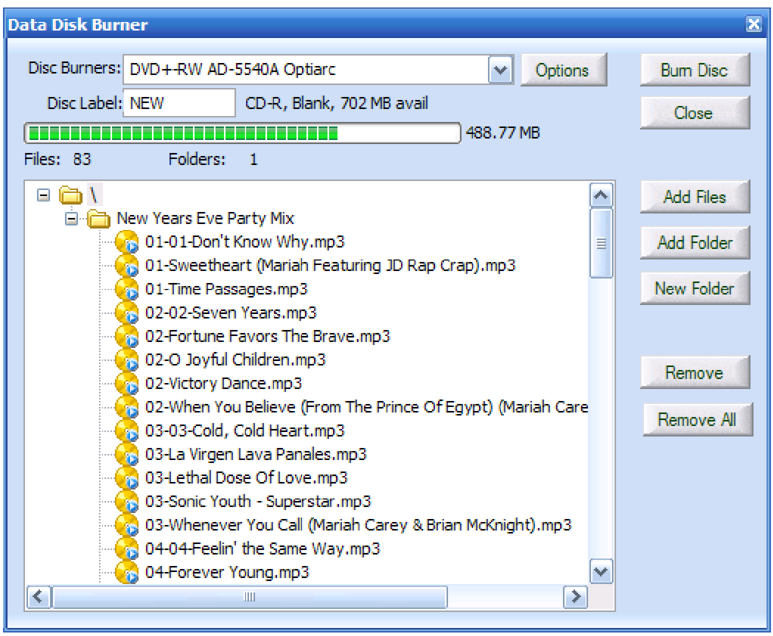
Demo Files

Quick access to your factory supplied demo files is available through this feature. Their file names are descriptive in nature, informing you what feature they are intended to be used for.

Data Disc Burner

Overview

The Diamond Cut Data Disc Burner facilitates optical media storage using formats other than just Red Book CD Audio†. You can archive data on optical media such as CD-Rs, CD-RW, DVD+/-R, and DVD+/-RW provided that your optical drive supports burning onto those formats. CD-Rs are capable of storing around 700 Mbytes of data while DVDs are capable of storing roughly 4,400 Mbytes (your mileage may vary). It works with most PC format data extensions including .wav, .mp3, .wma, .jpg, .bmp and so forth. The Diamond Cut data disc burner has three points of user access. You can access it via the Data Disc icon on the toolbar, or you can access it from either the File or the CD Prep menu under the rubric of "Burn a Data Disc". The Data Disc Burner supports complex hierarchical file tree structures having multiple directories and sub-directories. Disc layouts can be created either by browsing and adding or removing files directly from the Data Disc Burner dialog box or by dragging and dropping them into the Data Disc structure field directly from Windows Explorer. The file systems created by the Diamond Cut Disc Burner are ISO 9660 level 2 compliant. This system, however limits the number of characters assigned to a given file to 180 and the file extension limited to 3 with up to 8 directory levels. The Burner provides you with the option to choose the Joliet file system which provides for file names having up to 64 unicode characters (128 bytes) in length and directory hierarchies greater than 8 levels. Both Disc at Once (DAO) and Track at Once (TAO) modes of burning are supported, providing that your disc drive hardware supports them. Track at once mode provides the optional ability to keep sessions open after creating a CD session so that the disc can be added to at a later time.



The Data Disc Burner

The Data Disc Burner Controls

The following is a description of the various Data Disc Burner controls:

1. Add Files – Click on this button to add a file or multiple files to your disc layout from the Windows file system. First click on the location in the tree under which you desire the new files to appear. Then click on the Add Files button, browse to the file(s) of interest and then highlight them. Next, either double click on the file or click on "open". The file or files will then appear in your disc layout field.

2. Add Folder – Click on this button to add an entire folder to your disc layout from the Windows file system. First click on the location in the tree under which you desire the new folder to appear. Then, click on the Add Folder button, browse to the folder of interest and then highlight it. To move the file into your layout, click on the "OK" button. The folder will then appear in your disc layout field.

3. New Folder – This allows you to insert a folder into your new file structure. To do so, first click on the location in the tree under which you desire this new folder to appear. Then, click on the New Folder button. A dialog box will appear and you can then "Enter a Name for the New Folder". When you are satisfied with the name, click on "OK" and the new file will appear in the designated location in your disc layout. Files and other folders can then be added beneath that level in your disc layout.

4. Remove – This will allow you to remove a file or directory. Highlight the item in your layout that you wish to remove and then click on the Remove button. A dialog box will appear and query you to determine if you are really sure that you want the item removed. Click on "Yes" or "No".

5. Remove All – This allows you to remove the entire disc layout from your file tree except for the optical drive root directory. A dialog box will appear and query you to determine if you are really sure that you want to remove the entire disc layout. Click on "Yes" or "No".

6. Disc Burners Selector – Scroll to the Disc Burner that you want to use for your project.

7. Data Burners Options – Choose the setup that fits your needs:

A. File System Type

I. ISO 9660 (level 2)

II. Joliet File System (Long File Names)

B. Writing Mode

I. Track at Once (TAO)

a. Keep Session Open checkbox*

II. Disc At Once (DAO)**

C. Other Optional Attributes

I. Sort Disc Layout - When this is checked, the disc layout is placed in alpha-numeric order.

II. Enable Optimal Power Calibration - When this is checked, an extra step is added to the process to optimize the burning power of the laser diode. This creates a more consistent burn at the cost of longer disc burn time.

*Note: “Keep Session Open” is only available in TAO Mode and for CD Burning.

**Note: “Disk At Once” is typically only supported for blank CD-R discs and rarely supported on DVD media. TAO mode is the preferred mode for creating DVD Video backups or data backups.

8. Drag and Drop – You can also drag and drop files, folders and directories directly from Windows Explorer into the Data Disc Burner.

Data Disc Burner Sample Procedure

Here is a fairly typical example of a procedure that you can use to burn a data disc:

1. Launch the Data Disc Burner.

2. Browse the “Disc Burners” feature to the Optical Drive of your choice.

3. Insert a media disc into that optical drive.

4. The top portion of the Data Disc Burner display should show the type of media inserted into the drive and also the number of Mbytes that are available for use (after a short time delay).

5. Click on the “Options” button and check Joliet File System under the File System Type and then click on “Track at Once” under the writing mode option.

6. Use the Add Files, Add Folder(s) and New Folder features in order to create your disc layout in the workspace area found below the controls section of the Data Disc Burner.

7. When you have completed your layout, click on the Burn Disc button in the top right had corner of the application.

8. A progress bar will show you how things are progressing as the disc burns.

9. At the end of the burn process, the hourglass will remain showing while the system closes the disc. The progress bar will show an increasing percentage as this process proceeds.

10. The disc closing process could take as long as 15 minutes, so please be patient.

11. When the project has completed, the system will show a message which reads "Disc has been burned successfully" and a "gong" will sound via your sound system. Please remove the disc from the drive."

12. Done.

†Note: Red Book Audio CDs can also be created with your Diamond Cut Software. This feature is accessed via the CD Button on the toolbar or via the CD menu item titled "Burn a CD".

DC Tune Library

An Alternative Way to Focus the View of Your Audio Restoration Work

You have seen that the most typical way to view your work is via the **Classic-Edit** Source and Destination Workspace(s) as well as the **Fast-Edit** time domain display. Alternatively, the DC Tune Library can be made the primary focus of your Diamond Cut system by ticking off the "DC Tune Library" feature in the View Menu. Then, the DC Tune Library provides you with an alternative way to view all of your audio restoration projects and the subsequent archive that you will be creating. The DC Tune Library includes a full-featured audio file archive within the context of your DCArt10 or your DC Forensics10 software. It supports files in the .wav, BWF (Broadcast Wave), .mp3, .wma, .ogg and .flac formats. It can search your hard drive and find all of your audio files and store their path, name, genre and other pertinent data within its file structure. After your DC Tune database has been constructed, it will provide you with lightning fast access to any of your audio files, provided that they are in any of the supported formats mentioned above. You can search your database, sort it by various parameters and create and recall playlists. The DC Tune Library completely replaces the old Playlist feature in DC6 and prior versions.

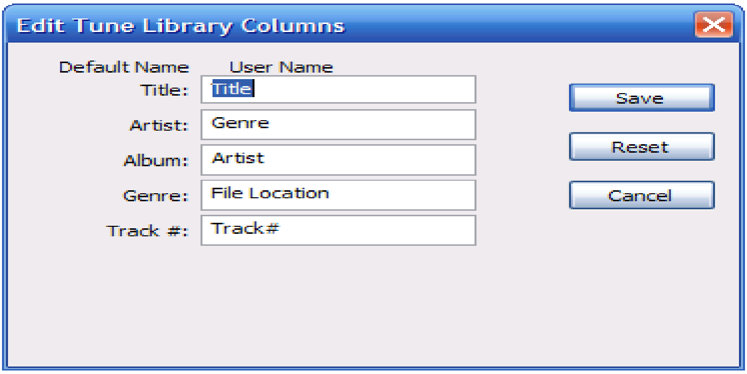
You can play any file directly through the normal play features of DCArt10 or DC Forensics10, or you can listen to a file or series of files by using the Preview feature found in the Diamond Cut filters and/or effects. You can even listen to the database by way of a complex series of filters and effects by previewing a file or playlist via the DCArt10 Multifilter. You can have the DC Tune Library start automatically by checking the "Start DC Tune Library" in the Preferences dialog (found under the Edit\Preferences Menu). You can also open this feature by going to the File menu, and then clicking on "Open DC Tune Library" and the spreadsheet view will appear. The system supports files having the .wav, broadcast wave, mp3, .ogg, .flac and .wma extensions. The system does not support files having the .wma format that incorporate Digital Rights Management (DRM) schemes.

To create a library from the files that exist on a particular hard drive or sub-directory, simply click on "File" and then on "Add Folder to Library". The preferences menu provides you with a number of options for obtaining information on a file, but .mp3 and .wma tag information will take precedence for those types of files. For files without tag information, such as Wave files, the Title, Artist, Album, and Genre information can be obtained from the directory structure of your hard drive. You can customize the way this information is used within the DC Tune Library's by using the preferences dialog. By default it looks something like:

<Genre>\<Artist>\<Album>\<Title>

Use this system in conjunction with the highlighting and the "Move Up" feature to align the file sorting to most closely resemble your database organization scheme. When adding a new folder containing multiple .wav files to the DC Tune Library, individual track numbers for files contained therein will be set to the order in which they were originally created based on their time-stamps. If individual audio files are added to the library (one at a time), they will appear in the DC Tune database in the order in which they were added.

You have the option to customize the names applied to the various DC Tune Library columns by using the "Customize Column Header Titles" feature found under the Tune Library Preferences menu (or by clicking on a column title with the right mouse button and scrolling to the bottom item). Just click on the Edit button in the preferences and the following dialog box will appear:



Customize DC Tune Library Columns Dialog Box

After you edit the User Names, just click on Save, and the columns will become customized. Please note that this column customization feature can also be accessed via the right mouse clicking on any of the column headers.

The system will take some time (anywhere from 30 seconds to 20 minutes, depending on your system and the number of files to be found) to build your library, but once it is constructed, the system will no longer need to search your hard drive each time you want to access a particular file. If you have files on multiple hard drives, you will need to perform the creation step independently on each drive. Ultimately, all of the file(s) information will be integrated into one DC Tune Library. Note that the DC Tune Library does not copy or change any of your original files; it simply saves their location and other information into the DC Tunes database.

This database is stored in the following path(s):

- - - \My Documents\My Music\DCArt10\DcTuneDb

And a backup copy is saved as:

- - - \My Documents\My Music\DCArt10\DcTuneDb.bak

The extension used for these files within the DC Tunes Library Database is .xml.

The first path is the primary database, while the second one (.bak) is its backup. A Backup copy of the database is created each time you exit the program. The paths to the various files are stored in the DcTuneDb directory, but not the files themselves, which are all left in their original directories. Editing a file (like removing it from the DC Tune Library) will not delete the actual sound file from your hard drive.

After a period of time, you may want to update the database by re-doing the search process. Files that already exist will not be duplicated (if the appropriate "Tune Library" preference is selected); only the new ones will be added to the library.

If you have deleted files on your hard drive but not in the DC Tune Library, you can update it by using the "Check Tracks" function under the "Tracks" menu. It will search for database entries that no longer have real audio files associated with them and highlight them for you.

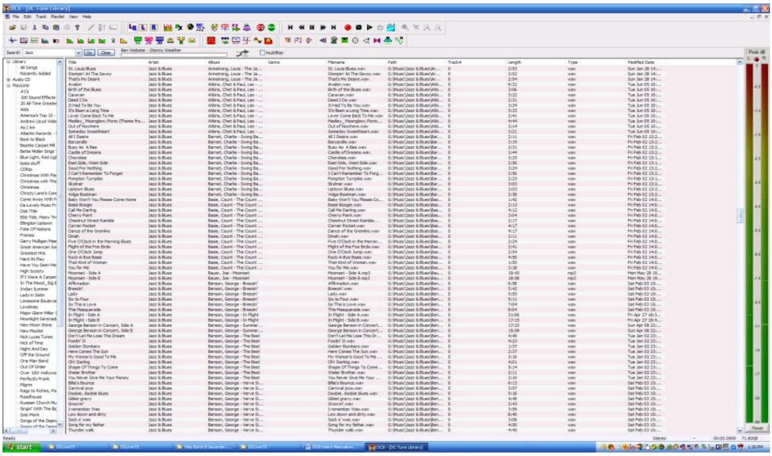
If you ever decide to re-create a database from scratch, you can delete the DcTuneDb and the DcTuneDb.bak folders and the system will re-create these when you attempt to rebuild your DC Tune Library. When you build a DC Tune Database, you have the option to use file tags to place items into certain categories based on file tag information. You can enable the tags feature in the DC Tune preference tab under the Edit Menu and are given several choices as follows:

- Use tags for track properties (MP3, WMA, flac, ogg
- Use BWF Info (Broadcast Wave File)

The desired tag extraction mode should be established before building your DC Tune database. If neither box is checked, no tag information will be used from any of your audio files.

The Library includes the following fields as viewed from left to right that can contain data concerning your audio files:

Song, Artist, Album, Genre, Filename, Path, Track #, Length, Type, Modification Date



The DC Tune Library Spreadsheet Display

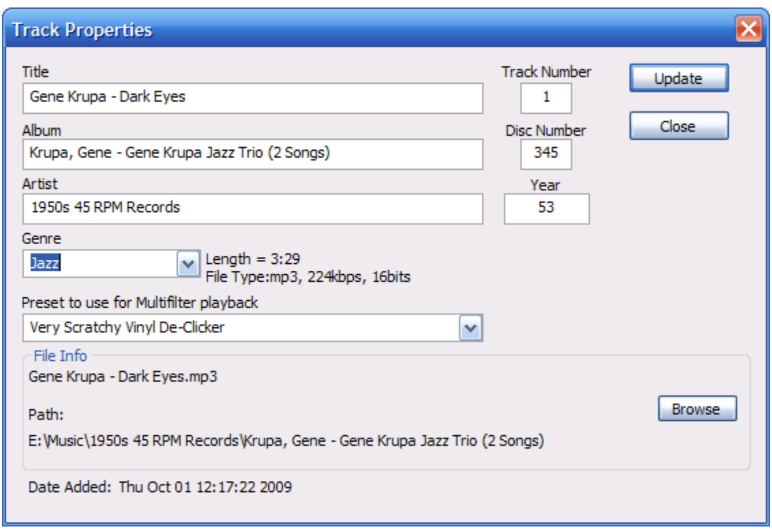
A number of actions can be performed after your DC Tune Library has been built, including the following that are all available on the menu that appears when you click your right mouse button on a file/track:

1. Display the Properties of a File
2. Play the Track(s)
3. Delete the Track(s)
4. Open the Folder for the track in question
5. Edit a Track using the Standard Diamond Cut features
6. Burn a CD from Selected Tracks
7. Create a Playlist from Selected Tracks

Additionally, you can search the database across all fields by using the "Search" field found in the upper left corner of the DC Tune Library. You can also perform a secondary sort by "Album" and then "Track" number when sorting on "Artist". If a track number is not found in the sort, the system will resort to the use of the file modification dates to establish the order of the items in the display.

After your DC Tune Library has been built, individual file access will become virtually instantaneous simply by double clicking on the desired item. The highlighted file will play and the system will continue to play the entire database thereafter until you click on the "Stop" button on the toolbar. The DC Tune Library also includes a search engine. It will find any string of characters that you type into the search box by looking at the entire directory pathway for all matches. If a certain column is hidden, items contained therein will not be used in your search. You can see the relative play location of a particular file via the play progress bar (the horizontal green display). You can advance or retard the location of play by pointing and clicking to a different location on the play progress bar and the system will jump and then play from the new file location. As play progresses through the library, the playing item will become highlighted.

If you only want to add a single file to your DC Tune Library, click on "Add File to Library" and browse to the desired item. Further functionality can be found by right clicking on a file and then on properties which brings up the following screen:



DC Tune Library Right Mouse Track Properties Dialog Box

Playlists and Tunes from the library can be played via the Diamond Cut "Play" button, or via any of the Diamond Cut Filters and/or Effects via the Multifilter "Preview" mode button. The Multifilter "Preview" function is an especially interesting way to listen to these files, since it allows you to string together a series of filters and effects that you can apply to the playback in real time. To commence a preview after bringing up a particular playlist, make sure that the first line in the playlist is highlighted and then click on the "Preview" button associated with the filter or effect of interest.

The standard Windows mouse selection commands can be used on the DC Tune Database. If you want to delete a particular item from the database, right click on it and then right mouse to "remove". If you want to remove a sequence of items, left click on the first item in the sequence, hold down the "Shift" key on your keyboard and then click on the last item in the sequence. Those items and all of the ones in-between will become highlighted. Then, right click on "remove" and they will be removed from your DC Tune Library. If you have a highlighted listing of files, and you want to un-highlight one item somewhere in the listing, depress the Ctrl key on the keyboard, and left click your mouse on the item of interest. To select all songs in the list, use the menu item, Edit/Select All.

Playlists are initially displayed in the order in which they are created. You can change the order by clicking on the desired column header of the DC Tune Library spreadsheet. Clicking on it once will put it in alphabetical order and clicking on it a second time will reverse that order, so you can have it ordered in either direction. Playlists can be re-ordered by dragging and dropping the songs from one position in the playlist to another. Note that only a Playlist can be re-ordered; if you have selected "All Songs" the order is determined by the sorted column.

Shuffle Play

Shuffle Play is a form of random access to the selections presented to you in the DC Tune Library spreadsheet display area. Instead of playing the files in order, the system randomly advances through the listing. After a file (tune) has been played, it will be omitted from the selection process for the next round of random selection. Thus, no tune will be played more than once during a "Shuffle Play" session. This feature is located just to the right of the play progress bar and its icon looks something like the letter "X" but with arrows pointing towards its right hand side. To activate this mode, click your mouse on the icon.

When it is in its active state, it will present itself in two shades of green. When it is inactive, it will present itself in two shades of grey. Each time this feature is enabled on a listing (even if the listing is the same as a previous listing) the files will be played in a different (pseudo) random sequence.

Playlists

Playlists are used to further organize your tune library. A Playlist is a subset of the tune library based on your criteria such as Album or Artist or just picked at random. Playlists are not only used to play tunes but also to Burn CDs and Create Batches for the Batch File Editor. It allows a sequence of files to be played or transferred to an audio medium without having to manually cue up each audio file in real time. This is also useful and particularly advantageous in a forensics legal situation wherein an expert may be required to play certain groups of files in real time as an element of their testimony.

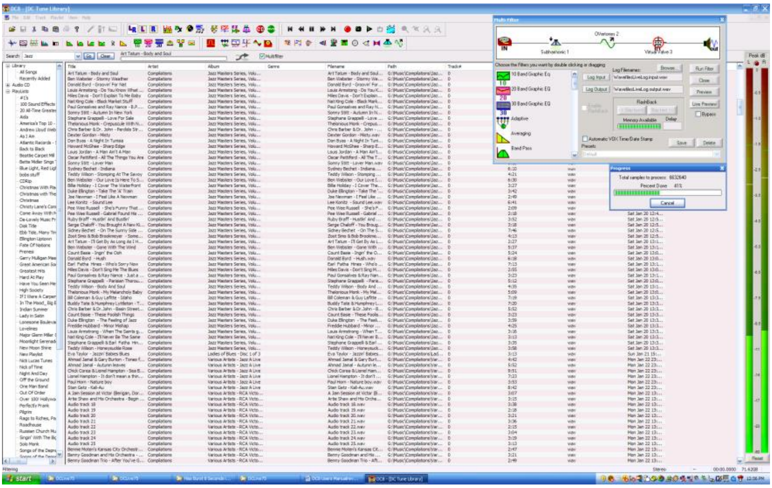
Multifilter Play Mode

Sometimes, it is desirable to apply filters, tone controls or effects to the list of files that you are playing through the DC Tune Library. The Multifilter checkbox in the DC Tune Library facilitates this functionality. Every filter and/or effect in the Multifilter becomes accessible via this feature and is applied to the file being played and does so in real time.

You can either use a selected preset, or use the currently selected preset when using the Multifilter for playback in the tune library. If you look at the track properties for any wave file, you can set the Multifilter preset that you want to use for that file. However, you also need to set the preference in the DC Tunes preferences page to “use the preset from the tune library”. The default mode is to <u>not</u> change the preset. Please note that placing too many filters into the Multifilter in conjunction with the DC Tune Library could lead to a situation in which the system “stutters” or “skips”. The reason for this is your CPU’s inability to keep up with the math required to process the data in real time. If this occurs, remove filters until the situation is resolved or use a more powerful computer. Please note that the DC Tune Library progress bar is not mouse - scrollable when using the Multifilter mode.

Continuous Playback Mode

A special mode will facilitate the repeated playing of a selected file or a list of files. The system can be placed into “Continuous Playback” mode by ticking off the checkbox having the same name found under the “Preferences Menu” under the “Tune Library” tab.



The DC Tune Library Operating in Multifilter Mode

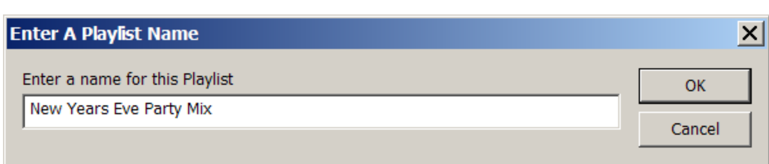
After checking the Multilter checkbox, the Multifilter will appear. When you click on the “Play” button, what you hear is what is being processed by the lineup of filters and/or effects set up in the Multifilter. A typical setup might include a 10 Band Graphic EQ, a Sub-harmonic Synthesizer and a Virtual Valve Amplifier. You can also use de-noising filters such as the EZ Impulse or EZ Clean filter. Actually, any of the filters or effects that are available in the Multifilter become available when using this DC Tune Library feature. Presets can be saved and affiliated with audio files. You have two preferences to choose from with respect to Presets for the DC Tune Library Multifilter function. They are found under Edit/Preferences/DCTune Library/Playback with the Multifilter. Select either “Use Preset from the Tune Library" or "Do Not Change the Preset". When in "Use Preset from the Tune Library" mode, the system affiliates a preset with an audio file via the right mouse button file properties dialog box. Go to the "Preset to Use for Multifilter Preset" listing and then scroll to the desired Multifilter preset (and settings) and it will be recalled in the future when that particular file is played. If a specific preset is not chosen, the DC Tune Library uses the "TuneLibDefault" preset. To de-activate the Multifilter feature, click on the "X" symbol in the top right corner of the Multifilter. For more information, please refer to the Multifilter section of this Users Guide.

Mono Mode

A Mono Mode feature is provided within DCTunes. This is most useful when a stereo amplifier is used to drive loudspeakers located in venues having different zones, but requiring proper stereo mixed playback of the tune(s). An example of this situation could be a restaurant setting having several dining areas or zones that need to be covered with background music.

Creating a Playlist from your DC Tune Database

Use the general techniques described earlier to select a group of tracks that you wish to constitute a specific playlist. After you have completed the selection process, and while still pointing your mouse at one of the selections, right click and select "Create Playlist from Selected Tracks". The system will choose the Album Name of the first item in the listing to use as the default the name for this playlist. If you do not want to use that name, you can simply edit it in the dialog box presented, as shown here:



The Playlist Title Box

The playlist will be stored in .xml format in the following directory:

- - - \My Documents\My Music\DCArt10\DCplaylists

The new playlist will now show up in the left hand margin of the DC Tune Database under “Playlists”. To play a desired playlist, click on the desired item, and then click on the “Play” button on the Diamond Cut toolbar. But, there are also a number of other options available to you on the menu which appears when you click the right mouse button on a particular playlist item:

1. Create New Playlist (using the existing one as the base)
2. Rename This Playlist
3. Delete This Playlist
4. Export This Playlist (dialog box provides export format alternatives)
5. Burn a CD from this Playlist
6. Run Batch Processing on this Playlist

You can also import pre-existing playlists having the following extensions:

.m3u,.wls (the older Diamond Cut Playlist Format), .cue and .pls. (These formats are sometimes referred to as M3U, WLS, CUE and PLS.)

To access this feature, go to the File Menu (of the spreadsheet display) and then go to “Import Playlist”. The system will convert the imported format into the required .xml format.

The system allows you to view and modify some of the properties of a particular file. To do so, right click on the file of interest in the DC Tune Library and then on “Properties”, and a dialog box will appear as shown below:

The DC Tune Library Track Properties Dialog Box

You can modify information directly in the dialog box. When you are satisfied with your changes, click on the "Update" button. Additionally, you can use the "Browse" button to find a file that has been moved or renamed.

You can use this feature to update the DC Tune database and the MP3 or WMA tag information for the selected files. If multiple files are selected, some fields are disabled such as the Title and Track Number. You can enter information for the group of files such as the Album Title and update all of the selected files at once.

<u>CD Creation / CD ROM Creation / CD ROM Burning / CD Burn</u>

In much the same way that you can construct a playlist, you can also set up a listing to create a Red Book CD. This feature is accessed via the right mouse button or the CD icon, and details can be found pertaining to its operation in the **CD Burner** section of this User's Manual found under the **CD Prep** section. It is worth noting that you can drag and drop items from the DC Tune Database directly into the CD Burner Dialog box.

Playlist Export Feature

Your Playlist can be exported into other programs that permit importing data such as disc label makers and CD-ROM printing programs. The feature will export the following parameters:

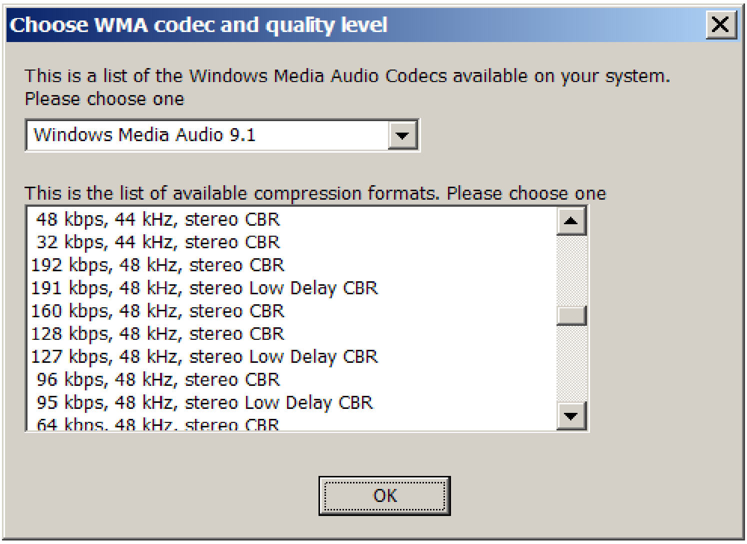
1. Title Number in sequence
2. Title of Selection

3. Length of Time of Selection
4. Total Length of Time of the Selections

Clicking on the Export button found on the right mouse button activates this feature which allows you to save it in a number of formats. When you click the Export List button, you can specify a file name. The default will be saved in the *.txt format which can be opened by many other programs. It can also be saved in your choice of other formats including .m3u, cue and .pls (sometimes referred to as M3U, CUE, and PLS).

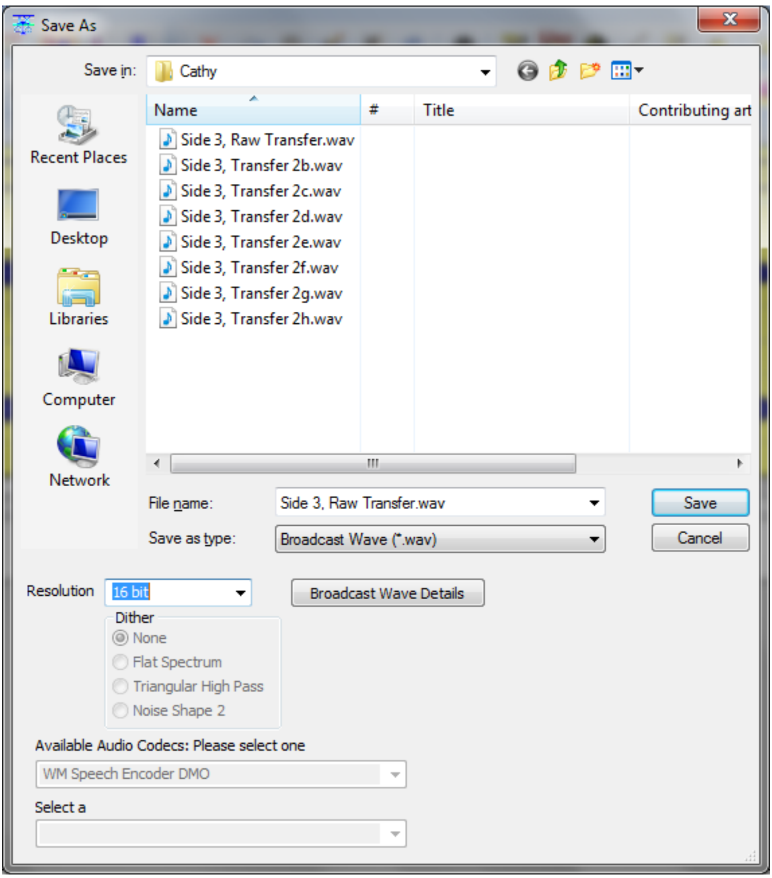
.wma File Encoding

You can encode a file in compressed .wma format if you so choose. To do so, click on "Save As" .wma and you will be presented with the following dialog box with a list of compression options from which to choose. In general, higher bit rates (kbps) produce better sounding files at the expense of greater memory space requirements. You decide which options best meet your needs.



The WMA Dialog Box with Compression Options

To view the other compressed formats that are available, click on Compressed Formats in the "Save as type" field and highlight "Compressed Formats". Then, click on the "Available Audio Codecs: Please Select One" field. A listing of available formats will be displayed. The contents of this listing will be dependent on your operating system and/or which Codecs you have installed.



Save As Dialog Box

DCArt10/DC Forensics10 has the ability to use an external Encoder to help you turn any .wav file into an MP3. This process simply allows you to download one of the popular encoders. To do so, go to your Edit/Preferences/Mp3 Encoder and define a path to that particular .EXE file. From that point on, if you choose to SAVE AS or Batch File Processing and want the end result to be MP3 files, DCArt10/DC Forensics10 will use that encoder to make the conversion. We recommend the LAME encoder as the most popular and highly regarded and it is the default encoder included with the software.

Playing CDs using the DC Tune Library

You can play CDs (Red Book Audio) using the DC Tune Library. Simply place your CD into your optical drive. After the drive "spins-up" (which will take a few seconds) double click on the words "Audio CD" which is in the "tree" listing within the left-hand column of the DC Tune Library display. You should then see the following appear as part of the tree:

"Audio CD: xx:yy"

where xx:yy is the length of the CD in minutes and seconds. Single clicking on "Audio CD: xx:yy" will then construct a listing of the tracks which will appear in the right-hand display of the DC Tune Library. To play these CD tracks, just click on the play button located on the tool bar. Most of the various controls previously described and associated with the DC Tune Library also apply to the CD Player (including Shuffle Play). However, the Multifilter does not work in conjunction with the CD player.

Import Playlists

This feature allows you to search your hard drive and import playlists into your DCArt10 software. It will search your hard drive for playlists in the .wls, .m3u and .cue formats and convert them into the .xml format for compatibility with the DCArt10 or the DC Forensics10 system.

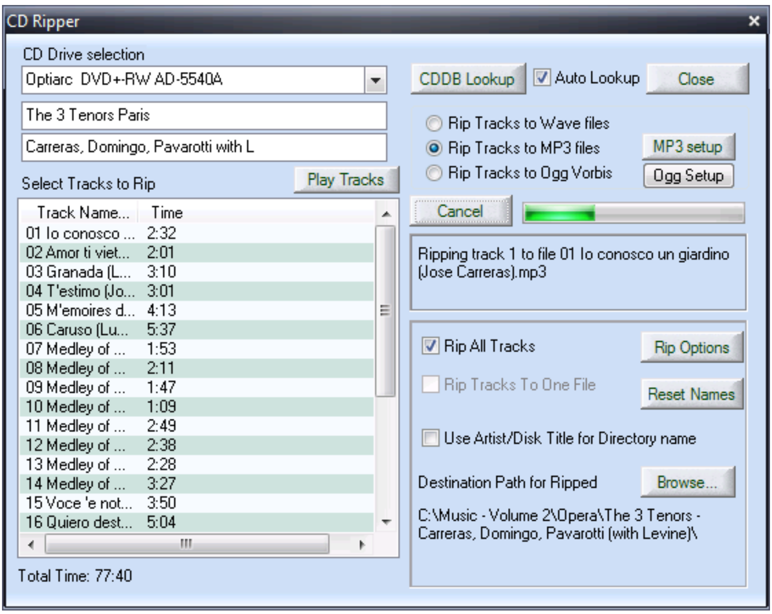
Close Source

This command closes a previously opened Source and/or Destination .wav file.

Rip CD Tracks

Rip CD Tracks to .wav, or .mp3 or .ogg

This feature allows you to "Rip" (convert) CD Red Book Audio CDs to Wave files (.wav) or MP3s (.mp3) or Ogg (.ogg). To operate this feature, first place the CD into the CD ROM drive in your computer and then launch the "Rip CD Tracks" feature found under the File Menu. A dialog box will appear with the following selections and features from which to choose:



The CD Ripper Dialog Box

- CD Drive Selection: Used to choose the CD ROM Drive that you want to use to rip your CD.
- Artist Field: Allows you to identify the Artist for the Ripped File
- Title Field: Allows you to identify a Title for your Ripped CD.

- “Rip All Tracks” precludes the necessity to highlight the desired tracks to be ripped from the CD.
- Select Tracks to Rip Box: This box shows all tracks which are available on the CD either as Track Numbers or Track Names depending on whether or not the CDDB had been utilized. Selecting Tracks to be ripped is accomplished as follows:
    - To select a single Track to be ripped, simply point your mouse towards the desired track, and use the left mouse button.
    - To select Tracks at random, point your mouse towards the track(s) of interest and use the Ctrl Key in conjunction with the left mouse button.
    - To select a range of Tracks to be ripped, first point your mouse at the bottom most track of interest in the list and use the Ctrl Left mouse button to highlight the same. Next, point the mouse to the top Track in the range of interest and click on Ctrl + the Left mouse button. The range of tracks between these two Tracks all will become highlighted.
    - To clear all selections, point the mouse to the right hand side of the Track selection box and left mouse click. All Tracks will become un-highlighted.

- A checkbox is provided for CD jitter correction. Potentially this can improve the accuracy of the transfer process, with the tradeoff of a slower ripping speed.
- CD Database automatically assigns track names to inserted CDs
- CDDB Lookup: CD Data Base Lookup correlates the tracks on your CD to Song Titles that can be found on the Internet. To use this function, first you must place the CD in the CD ROM Drive. Next, launch the “Rip CD Tracks” feature. After the track list is developed and you are connected to the Internet, clicking on the CDDB Lookup Button will find (if available) the CD title and song list and convert the tracks to song titles. This process will take place automatically by selecting the Auto Lookup checkbox, but will only work if you are connected to the internet at the time of the CD rip. If you want to edit/change the name of a particular track, use the Windows slow left-mouse double-click on the track of interest

(double click with about 1 second between clicks). It will then become highlighted, allowing you to change the track title. Setup for the CDDB system can be found in the Preferences section of the Edit menu under the “CDDB Setup” tab. For more details regarding CDDB setup, please refer to the preferences section of this manual.

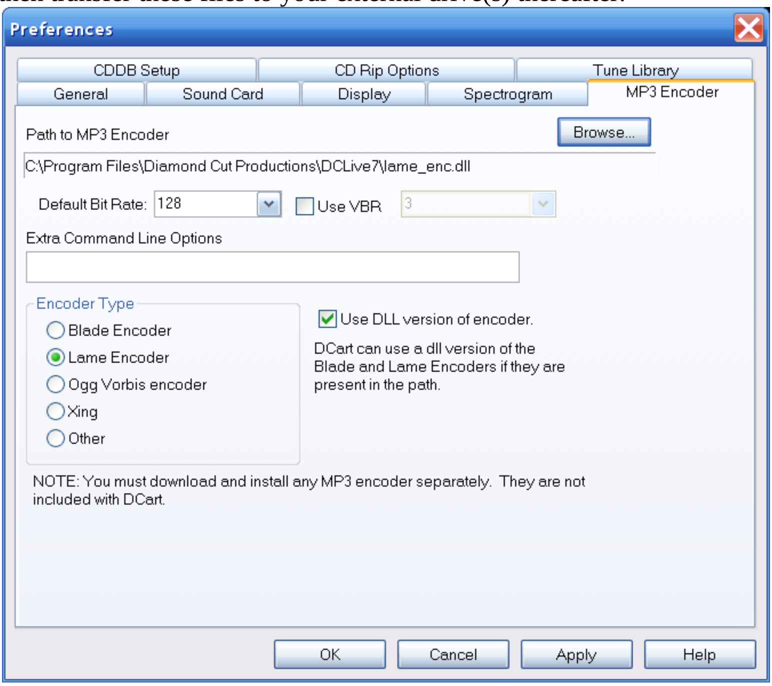
- Selector: Choose between the following:
    - Rip Tracks to .wav files
    - Rip Tracks to MP3 Files (You will need to install an external encoder for this feature to work. Please refer to the MP3 Preferences Setup section of this manual)
- Rip Tracks Button: Click on this button to start the “Ripping” process.
- Ripping Status Box: This is located directly below the “Rip Tracks” button and includes the following:
    - The system will indicate if the System is “Ready”
    - The system will indicate Error Messages
    - The system will indicate the progress of the ripping process via a “Progress Bar”
    - It will indicate when the System is “Done”
- Modes of Operation Checkboxes:
    - Use Artist/Disc Title for Directory Name
    - “Rip All Tracks to One File”: This can be useful in situations wherein no dead-time is desired between contiguous tracks or when you want to process all the files on a CD through a filter in one single operation.

Destination Path for Ripped Files: Browse to the desired Directory

An MP3 Setup button is provided to quickly bring up the MP3 encoder(s) preferences box should you desire to make some changes in its settings, precluding the necessity of having to go to the Edit/Preferences menu item.

Note 1: ID3V2 mp3 tags are automatically added by the ripper to MP3 files. These include the Title, Artist, Album, Track #, Maximum tracks and Genre.

Note 2: Ripping directly to an external (USB or Firewire) drive is not recommended. Timing errors in the data pathway can result in the creation of corrupted files. If you want to maintain your ripped files on an external drive, rip them to a directory on your "C" drive first and then transfer these files to your external drive(s) thereafter.



MP3Encoder Setup

CD Ripper Preferences

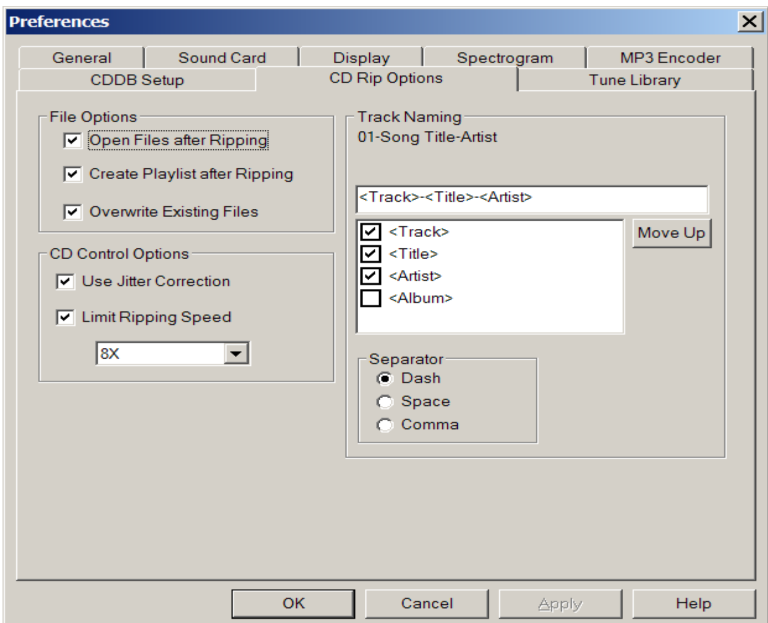
If you click on the "Rip Options" button, the CD Ripper Preferences Menu will appear.

File Options

This controls your file overwriting and file opening options.
The file format extension that is created by the "Create Playlist after Ripping" feature is .xml.

CD Control Options

One feature involves the limiting of the ripping speed of the system. This is useful sometimes when a CD ROM has become warped due to its paper label (or due to other variables). Disc “warpage” can create a situation in which the CD ROM will not rip, but slowing the disc down can sometimes overcome this difficulty. Also, it is generally good practice to leave the “Jitter Correction” on, but it substantially slows down the ripping process due to the extensive calculations required to perform this task. But sometimes better CD ROM ripping success may be had with this feature turned off when encountering extremely problematic discs. Experimentation is the only way to know for sure what will work when ripping warped (or out-of-balance) CDs.



The CD Ripper Preferences Menu

Track Naming
You control how the resulting ripped tracks are named via the Track Naming feature. Typically track names are simply the track # and song title, but many folks prefer to include more information in the title.

The system derives its information from the CDDB or via manual data entry in the Ripper and constructs the file name accordingly. Select the fields you want included in the name and arrange them by using the "Move Up" button. Four choices are provided along with 3 possible data separators including a Dash, Space or a Comma. To change the order, click on one of the attributes and then click on the "Move Up" button. That will cause that item to be moved one position to the left in the hierarchy. Repeat the process until you obtain your desired final result.

Open Destination

This command allows you to define the name and desired storage location of the processed version of the .wav file that you are about to create through the use of the various signal-processing tools of the DCArt10/DC Forensics10. The use of this command is optional since the software creates temporary files automatically.

Save Destination As

Since DCArt10/DC Forensics10 does not require the Destination workspace file name to be defined before your audio processing session, this command is used to define a Filename and directory location for your Destination file following the completion of an audio processing session, should you desire to save it. This function can also be activated by using the file folder icon in the File Tool bar of the workspace or by using the **Ctrl + S** hot key. Note: It is almost never necessary to manually save a file in the Classic Edit Mode. The program automatically does this for you.

Close Destination

This command allows you to close a file when operating in Classic Edit mode that has just been processed from the Source file. The working Destination file will have been stored on a temporary basis in a dctempxx.wav file. When you attempt to close the Destination file, you will be prompted to indicate whether or not you want to save it. If you do, then you will be prompted to define a path and a name for your processed file to be saved in.

Clone Source

Quickly copy a .wav file from the Source to the Destination Window

This feature takes whatever .wav file exists in the Source Window and replicates it in the Destination Window. It is only applicable in the Classic Edit mode and not the Fast Edit mode. It is particularly useful when operating the system in Selective Filtering Mode. Highlighted areas of the Source file that are selectively filtered will be modified in the corresponding time slot of the cloned file found in the Destination window.

Note: Sync files mode (found under the View Menu or via the Sync Files Icon) must be enabled for Selective Editing to work properly in Classic Edit mode.

Make Destination the Source

This command takes the file that has just been processed in Classic Edit mode, and makes it into the Source file in a new workspace window. This is a useful feature, since most sound jobs require several passes utilizing several different signal-processing techniques to affect a complete audio restoration. When using this command, the program will prompt you to name the file. It is recommended that you accept the suggested name. The original Source file may be deleted when making the Destination the Source by using the appropriate checkbox in the dialog box. *This command is grayed out until you actually have a Destination file.*

Delete Files

This feature allows DCArt10/DC Forensics10 to delete a file from a hard drive. Since .wav files tend to be large, this command will be used often. The software will prompt you to be sure that you want to delete the selected file before doing so. Remember that every minute of stereo audio sampled at 44.1 kHz consumes 10.584 Mbytes of disc space, which is useful to know when it comes time to clear up some disc space in order to get ready for your next sound restoration job.

Deleting a Wave file (Tutorial)

1. Warning! This operation cannot be undone!
2. Click on “File” and a pop down window will appear.
3. Click on “Delete File” and the Delete File Dialog Box will appear.
4. Choose the Drive and the Directory from which you desire to delete a .wav file.
5. Click on the Filename that you desire to delete in the Filename field.*
6. Another dialog box will appear, inquiring whether you are sure that you want to delete the chosen file.
7. If you click on "yes", the file will be deleted.
8. If you change your mind, and click on "no", the DCArt10/DC Forensics10 program will revert back to its initial window, and the file will not be deleted.

*__Note 1__: Multiple files that are sequential in the file listing can be deleted in one operation by clicking on the first item on the list, and then dragging the mouse pointer down to the last file you desire to delete. The files that are about to be deleted will be highlighted.

*__Note 2__: Multiple files that are not sequential in the file listing can also be deleted in one operation by holding down the CTRL key at the same time that you click on the appropriate item you wish to delete with the left mouse button. The files that are about to be deleted will be highlighted.

Print

This commands prints the screen as defined by your choices identified in “Print Setup.” It prints the present DCArt10/DC Forensics10 screen as “WYSIWYG” (**W**hat **Y**ou **S**ee **I**s **W**hat **Y**ou **G**et). Note that the screen is a very complex graphic with high-resolution images for Forensics applications. As a consequence, some laser printers will exhibit a problem using the Print command because of their rasterization demands – this just means your printer does not have enough internal memory and would be unable to print any graphic of this complexity. This is not generally the case with inkjet printers.

This function can also be activated by using the Printer icon in the File Tool bar of the workspace or by using the **Ctrl + P** hot key.

Printing Help-File Topics

Sometimes it is useful to be able to read Help file topics from paper rather than from your computer screen. This is accomplished in the following manner:

1. From the Help file, select the topic you are interested in printing.
2. Click on the "Print" icon with the left mouse button at the top of the Help File system.
3. Select the desired Printer and then click on the "Print" button (located near the top of the Help-file dialog box).

Printing a Screenshot (Tutorial)

Though DCArt10/DC Forensics10 includes its own Print function, you may still find the need to print a specific multi-layered screen.

1. When you've focused in on the screen sector that you want to print (i.e., a filter or dialog box), simply hit <u>Alt</u> and <u>Print Screen</u> simultaneously. This will copy the image onto your Windows clipboard.
2. Enter any paint-type program or even a word processor that accepts graphics and hit <u>Ctrl V</u> or use the Paste tool from the Editing menu.

<u>Note:</u> If you need to print the entire screen, just hit the "Print Screen" function on your keyboard. Do not use the "Alt" function.

<u>Print Preview</u>

This function simply shows you a snapshot preview of what you're about to print.

<u>Print Setup</u>

This command opens the Print Setup dialog box in which you can define the following parameters:

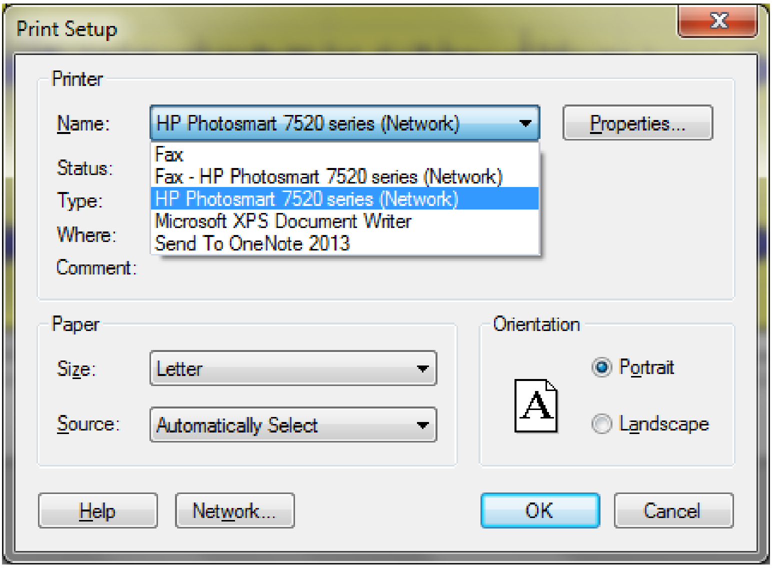
1. Choose the Default Printer or choose some other printer

2. Choose the orientation of your printout sheet:
    1. Portrait (This orients the paper vertically)
    2. Landscape (This orients the paper horizontally)
3. Choose the paper parameters that you desire:
    1. Size (Default value is 8 1/2 inches x 11 inches)
    2. Paper Source (Choose between the paper cassette or manual feed)
4. Properties: This allows you to select amongst the various printer dependent parameters provided by your specific device.

Page Setup

This feature allows you to select among the following printing choices:

1. Include File Information in printout (File name and Date of last modification)
2. Print the Source Waveform
3. Print the Destination Waveform or Spectrogram
4. Create a Page Title up to 80 characters long (the default title is "Diamond Cut Audio Restoration Tools"). To enter your title, just type over the default title. The maximum length that will appear on your printout will depend on the font that you choose, smaller fonts allowing longer character strings. To define the font size, go to the Preferences menu. Under Preferences, choose the "Display" tab. Under the "Display" tab, you will find a field in which you can enter an integer value for "Display Font Size".
5. Set Margins Spacing (a personal preference)



The Print Setup Dialog Box

Exit

Note: Exiting the program will also clean up all temp files in the Classic Edit Mode. It will ask you if you want to save any unsaved files before you exit.

The Edit Menu

Undo

This feature allows you to return to a previous version of a destructively edited file after using such features as Mute, Fade-In, Fade Out, Cut, and Copy / Paste / Insert. After an "Undo" is performed, it is removed from the "Undo" listing. You can also access the function using the **Ctrl + U** Hot Key.

All single file operations can be undone by using the Undo menu item. The default number of undo levels is 10. The number of undo levels is selectable in the Preferences dialog box (Edit/Preferences/General Tab). When you close DCArt10/DC Forensics10 (Exit the program) all undo information will be lost. Many Edit Menu functions will not appear in the list if no file has been opened.

Undo Procedure Using Classic editing mode (Tutorial)

1. Click on the "Edit Menu."
2. Click on "Undo."
3. Click on the menu level of “undo” that you desire to return to. The top "undo" in the stack represents the state of the .wav file prior to the latest operation which you performed, and the next down in the stack is the one before that, and so on. Variable levels of undo are provided, and are settable under the Preferences section of the "Edit Menu."

Undo Procedure Using Fast-Edit Mode (Tutorial)

Use the Fast-Edit history window to temporarily go back to the last step. This method does not permanently undo the last action. Use the right mouse button to delete the last operation. Double click or use the right mouse button menu in the Fast-Edit history window to delete the last action or series of actions. You will always be prompted by the computer to make sure this is what you want to do. This operation does not display the undo levels in the Edit/Undo Window as the Classic mode does.

Copy

This is the key with an icon consisting of two paper documents contained within its perimeter. It is used to Copy a highlighted selection of a .wav file from either the Source or Destination .wav file, and place it onto the program's clipboard. This function can also be accessed by using the Copy Icon in the upper left tool bar or by using the **Ctrl + C** Hotkey. You can also access Copy and several other functions by selecting an area and using the right mouse button menu.

Copy and Paste Procedure (Tutorial)

1. Determine the section of the .wav file that you wish to copy for the "copy and paste" operation. *You may use the Zoom-In feature if you would like to copy a very small portion of the Wave file.*
2. At the beginning of the section of the .wav file which you desire to copy, click down on the left mouse button and keep holding it down as you drag the timing bar (with the mouse) towards the right of the workspace.
3. Stop dragging the mouse at the end of the desired "copy" sector of the .wav file.
4. Release the left mouse button. You will notice that the sector between the two timing bars will remain highlighted in yellow. This will be the "copy" sector.
5. Click the right mouse button anywhere in the workspace area and a pop-up window will appear, providing you with a number choices.
6. Click the right mouse button on "copy" and the highlighted sector will be transferred to a temporary storage location on your hard drive (the “clipboard”).
7. After the transfer is complete, highlight the area in your .wav file where you desire to "paste over" the previously copied segment using the mouse drag procedure outlined in steps # 2 through 4.
8. Click the right mouse button again anywhere in the workspace area.
9. This time, click on "Paste Over" Or “Paste Insert”.

Important Note:

The timing rules used by "Copy and Paste Over/Insert" are as follows:

1. Copy and Paste Over operations always begin at the leftmost timing marker of the highlighted area of the workspace.
2. If the "Copy" sector is shorter than the "Paste Over" sector, the length of insertion is determined by the length of the "Copy" sector of the .wav file.
3. If the Copy sector is longer than the "Paste Over" sector, the length of insertion is determined by the length of the "Paste Over" sector of the .wav file.

Manual De-Click with "Copy" and "Paste Over" (Tutorial)

Note: This is one of many methods available to you in your Diamond Cut Software for manual interpolation of damaged sections of .wav files.

1. Listen to your .wav file and determine the location of the click, pop, or thud that you desire to eliminate.
2. Zoom-In on the section of the .wav file containing the click using the feature having the same name.
3. Continue Zooming-In alternately listening to the .wav file until you see the troublesome artifact in the DCArt10/DC Forensics10 workspace. It will take some training to be able to identify transients visually, so be patient during your learning curve.
4. Using the left mouse button, highlight a sector of the .wav file just prior or just after the transient event, being careful not to overlap the highlighted sector onto the actual transient. The highlighted sector must be at least as long (or longer) as the transient event.
5. Click on "Edit."
6. Click on "Copy."
7. Using the left mouse button, highlight the transient event itself.
8. Click on "Edit."
9. Click on "Paste Over."
10. Zoom back out and listen to the .wav file.

Important Note:
The replacement algorithm used in the Impulse Noise Filter is much more sophisticated compared to one used in this manual de-clicking procedure. Whenever possible, you should use the Impulse Noise Filter to de-click a record or the interpolator "I" key or the "O" key on your keyboard for manual interpolation of a highlighted area. Manual interpolation can easily be accomplished on tablet computers too by highlighting the event needing interpolation and then touching the manual interpolation icon. Also, worth consideration, is the use of the Pencil Tool. Only in the unusual or extreme case wherein these other tools have been unable to remove a particular artifact, should you use this "Paste Over" process.

Cut

Just as the title implies… Cut allows you to highlight an area of your file and then remove it from that file. For example, this feature is useful when it is necessary to reduce the musical portion of a segment for a competitive event in which the total length of the musical program is governed, and you do not want to eliminate either the beginning or the end of the song to achieve that end. Additionally, it is useful for eliminating long sectors of silence from Forensics files. You can also access this function by using the **Ctrl + X** hotkey.

Splicing out a portion of a Wave file (Tutorial)

1. Highlight and play the portion of the .wav file you believe that you would like to splice (cut) out.
2. Play the sector (using the Play button on the toolbar) to make sure that you have identified the correct timing for the segment you wish to remove. Re-highlight the correct area if necessary. The use of Markers (Markers Menu) and the Time Display (View Menu) can provide you with a high degree of timing precision prior to "cutting" a sector from the file.
4. Click on the Edit Menu.
5. Click on "Cut".

Important Note:
Fast Edit Mode can perform multiple cuts on very long files almost instantly. Classic Edit mode takes longer to achieve a "Cut" operation.

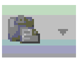
Manual De-Clicking with the Cut feature (Tutorial)

1. Zoom-In on the area of interest in the .wav file.
2. Highlight the click or pop impulse using the mouse. This should be a very small segment of the file.
3. Click on the Edit Menu.
4. Click on "Cut".

Important Note:
Removing clicks by cutting, although very easy to use, is not recommended because it actually shortens the total program length

from the original. Instead, consider using the manual interpolator "I" key (or the pencil tool) which preserves the file length and can produce very good file sector substitutions.

Paste



As the name implies and similar to every word processor you've ever used, Paste allows you to take whatever element you've cut or copied onto your clipboard and put it back into your current file in one form or another. We've created many different ways to paste your material (10 alternatives are provided in the Paste menu structure). When navigating this feature with its icon, click on the down-arrow to expand out the menu of choices. It will only come to life if there is something already on the copy clipboard to be pasted in some manner. We've thought of everything but "Paste Eat", which we've reserved for the Elementary School edition of the product.

Append to End

Paste Clipboard Contents Directly To End of a File

The "Append to End" feature takes whatever file or portion thereof you have copied onto the clipboard and attaches it to the end of the displayed .wav file. The resulting file will become larger so you may have to zoom-out to see it in its entirety.

Insert at Start

Paste Clipboard Contents Directly to the Beginning of a File

The "Insert at Start" feature is the complimentary function to the "Append to End" feature. Simply stated, it takes whichever file has been copied to your clipboard and attaches itself to the beginning of the displayed .wav file. As a result, the total newly formed file length will become larger so you will have to zoom-out to see it in its entirety.*

Time Domain Manual Interpolation Technique)

The Diamond Cut Time Domain Interpolator allows you to manually correct a recording impulse noise defect such as a tick, pop, click or thud. Some folks fondly refer to this technique as the Diamond Cut "Oterpolator" named after the "O" hot-key that is used to activate it. This system employs time domain interpolation techniques to calculate an accurate replacement signal and is primarily optimized for short time interval situations.
To use the time domain paste "oterpolator", simply highlight the area in the Source file in which you are observing a noise event. Next, strike the "O" Hotkey on your keyboard. The noise event will be replaced with a new waveform. The new waveform is calculated by a high-order modeling algorithm utilizing up to a maximum of around 2,200 samples of data per channel (up to approximately 0.05 seconds on a 44.1 kHz sampled file {50 mSec}). If you try to exceed the algorithm's limit, you will be prompted as such. If that happens, consider using the Diamond Cut Bi-Modal Interpolator (the "I" key) which is capable of longer interpolation time intervals.

Manual De-Clicking with the Time Domain Interpolator (Tutorial)

1. From the Source Workspace, listen to your .wav file using the Play feature, and determine the approximate location of the click, pop, or thud that you desire to eliminate.
2. Zoom-In on the section of the .wav file containing the click using the feature having the same name.
3. Continue Zooming-In, alternately listening to the .wav file until you see the troublesome artifact in the DCArt10/DC Forensics10 workspace. It will take some training to be able to identify transients visually, so be patient during your learning curve.
4. Highlight the transient event with the mouse drag procedure.
5. Depress the "O" key on your keyboard. The transient will be replaced with new signal approximating the audio waveform that should have been there.

Note 1: If you are interested in interpolating only the Left or Right channel with the Time Domain "Oterpolator" routine (rather than both channels together), just use the L or R channel selector buttons on the

toolbar first. You must be using WDM drivers, however, for this method to work; it will not work with MME drivers.
Note 2: Other methods of manual interpolation include Paste Interpolate (Bi-Modal Technique) and Pencil Editing. Please refer to those sections of this users guide for details.

Paste Interpolate (Bi-Modal Technique)

DCArt10/DC Forensics10 provides you with an automatic switchover method for inserting manual interpolations onto a waveform called "Paste Interpolate" via the "I" key. It is used in the same manner as the standard Time Domain Paste Manual Interpolator. This bi-modal interpolation technique uses a combination of time and frequency domain algorithms to determine the best replacement signal and is better optimized for longer time interval interpolations of up to around 19,800 samples (up to 0.45 seconds on a stereo 44.1 kHz sampled file {450 mSec}). It automatically converts to a frequency domain system when the selected area of the waveform is greater than a few milliseconds in length. Along with the capability of fixing very large impulse noise events (events which are very long and tall) with improved accuracy, it can also be used to remove other recording anomalies such as coughing, chair movement or short duration PA system feedback bursts from a recorded concert performance. Additionally, it can be used to correct things like "fret" noise from a studio recording of a guitar and other similar studio recording related problems.

Its interpolation capability has been lengthened and its accuracy improved on long events compared to the original Diamond Cut time domain only counterpart. Previous versions of the Paste Interpolate function utilized only time domain curve fitting techniques. This version now uses time domain as well as frequency domain interpolation techniques. This combination of two techniques results in more accurate results, especially on longer time interval events. From a user's perspective, it works in the same manner as the time domain Paste Interpolator, but uses the original Diamond Cut **"I"** Hotkey. If you want to interpolate the Left Channel only, use either the **"J"** or the **"SHIFT + I"** Hotkeys. Conversely, if you want to interpolate the Right Channel only, then use either the **“K”** or the **“CONTROL + I”** Hotkeys. It automatically switches from a time domain interpolator to a frequency domain interpolator on highlighted signals which exceed several milliseconds.

Note 1: Generally speaking, the best interpolations for short impulsive events (250 samples and below) are achieved via the Time Domain Interpolator routine (the “O” key
Note 2: Generally speaking, the best interpolations for long impulsive events (250 samples and above) are achieved via the Bi-Modal Interpolator (the “I” key).
Note 3: Notes 1 and 2 are not hard and fast rules and the best interpolation results are due to a combination of the method and the nature of the audio material that you are working with. Experimentation is the best way to make the determination. When all else fails to provide a satisfactory result, try the “Direct Spectral Editor” (found under the Edit Menu) which uses only frequency domain techniques. Compare its result with the result produced by the **“I”** key and its affiliates.
Note 4: Other methods of manual interpolation include Paste “Oterpolate: (Time Domain Technique – the “O” Hotkey), Pencil Editing and Direct Spectral Editing. Please refer to those sections of this users guide for details on their operation.

Note 5:  When using a tablet, you may find it more convenient to touch the manual interpolation icon button after highlighting the area in need of repair.

Paste Over

Paste Over allows you to insert the portion of the .wav file located on the clipboard over the top of a different location in your .wav file or to other .wav files. (This operation will delete the portion of the .wav file that previously had been in the particular location, installing the temporary file in that position instead.) The “Copy” and “Paste Over” feature in DCArt10/DC Forensics10 can also be used to manually “de-click” or “de-pop” a sound source (see tutorials under “Copy”). This function can also be accessed using the **Ctrl + V** Hotkey or by right clicking the mouse after you’ve copied or cut a portion of the .wav file.

For various tutorials of Copy and Paste Over, you can refer to the Copy section of this manual.

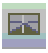
Paste Insert

Unlike Paste Over, Paste Insert does not wipe out the sector of the .wav file where you desire to place the contents of the "Copy" temporary file. Instead, "Paste Insert" adds the contents (from your clipboard) to the area of interest. In other words, you are pasting into the targeted area and the Diamond Cut system just shifts the existing material over in time. This process lengthens the time duration of a .wav file.

Paste Mix

Paste Mix allows you to add or "mix" one file (or a portion thereof) to a second file. This feature is useful for creating "voice-overs," or inserting special effects on top of a previously created sound track. This feature works in conjunction with the Copy function. In many cases it will require that two files be opened, one in the Source Workspace, and a second in the Destination Workspace. But this is not mandatory in that you can "paste mix" a portion of a file back onto itself if desired. The file that you open in the Source Workspace can be the file onto which you will be mixing. The File which you will be establishing as the "voice over" or special-effect, might be the one opened in your Destination Workspace. In other words, you can mix the Destination file into the Source File, in this example. The process can also be performed in reverse, wherein you can mix a portion of the Source file into the Destination File. These processes are undo-able, so that you can experiment until you are satisfied with the result. To use this feature, you will be highlighting the portion of the Destination File that you want to mix into the Source file. You will then use the Copy command to place it on a clipboard. Then you will highlight the Source file location in which you want the voice-over mixed in. When you run Paste Mix, you will be able to adjust the Source and Destination gain settings over a range of from +12 dB to – 100 dB.

Paste Crossfade



Paste Crossfade is the cousin of the Paste Mix feature. It operates in a similar manner, with the difference that there is a time varying function applied to the gain settings, so that a “cross-fade” effect can be produced. This feature is useful when you want to fade one song (or file) into another, with no “dead-air” in between. When you run Paste Crossfade, you will be able to adjust the File-1 (clipboard) Start and Stop Gain settings as well as those for the target file (File-2). You have available four gain controls in total. You will also be able to control the dynamics of the cross-fade by selecting Linear In, Log In, or Log Out. This feature is undo-able when executed under the Edit Menu. The “Crossfade” feature is also available under the Filter Menu.

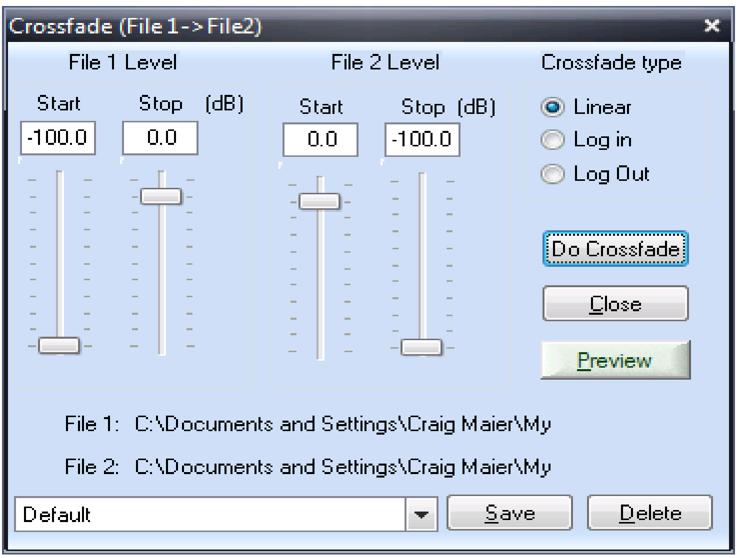
Using the Paste Cross-Fader Tutorial (Classic Edit Mode Only)

This method of cross fading is undo-able.

1. Open the File (song) into the Source Workspace that you desire to be the first in your cross-fade timing sequence.
2. Open the File (song) into the Destination Workspace that will be the second song in your cross-fade timing sequence.
3. Highlight, using the mouse drag procedure, the file to which you will be cross-fading in the Destination Workspace. You must highlight the file from the point of segue all the way to the end of the song, assuming that you desire to maintain the entire song in the sequence.
4. Click on "Copy" under the Edit Menu. This procedure may take a bit of time as this relatively large file is copied onto the clipboard.
5. Next, highlight the end of the file located in the Source Workspace. The area that you highlight will determine the cross-fade timing interval. The longer you make this interval, the slower will be the cross-fade sequence.
6. Click on "Paste" under the Edit Menu.
7. Next, click on "Paste Cross-fade" under the paste menu.
8. Set the gain controls in the dialog box as follows: (default values will work)
    - File 1 Levels:

- Start = -100 dB
- Stop = 0.00 dB
- File 2 Levels:
- Start = 0.00 dB
- Stop = -100 dB

9. Choose the cross-fade timing that you desire. Linear usually produces a pleasing effect.
10. Click on "OK"
11. After the processing has been completed, you may click on the play button on the Toolbar to hear your results.



Paste Crossfade Dialog Box

Paste As A New File

This feature provides a convenient means to chop a large file into smaller pieces, and assign new .wav file names to these subset files. It is useful for creating a number of .wav files that could be listed and quantized for CD-R indexing from a single large file such as that which you might have from transferring a Vinyl LP, or a concert tape recording to DCArt10/DC Forensics10. In Forensics applications, this feature is useful to isolate shorter conversations from a large file and then to create a file listing of those conversational snippets. A popup window will appear in which you can redefine a file name for each “chopped” file subset when using this command.

Paste Silence

This command simply inserts a predetermined amount of silence somewhere (your choice) in your .wav file. You can select from the Beginning of the file, End of the File or the Beginning of a selected area. This command will increase the total length of your .wav file.

Paste Bleep (tone)

Insert A Tone To Cover Objectionable Material

The Paste bleep tone feature interjects a 440 Hz Sine wave (A above middle C) at a –10 dB level over the highlighted area of a .wav file. It is intended to be used to “bleep out” unwanted verbiage or other sounds. It is commonly used in the Forensics audio field to redact portions of an audio file. To use this feature, merely highlight the area of the .wav file in the Source Window that you want to “bleep out.” Then, navigate to the Edit/Paste/Bleep (tone) feature. The highlighted area of the .wav file will be replaced with the bleep tone. This is an undoable function (it can be reversed).

Select All

This command selects the entire .wav file. It can also be activated by using the **Ctrl + A** Hotkey or a double left click within the file area. If you have markers present in the file, double clicking will highlight the area between the two closest markers.

Pencil Editing

This device is provided for sample-to-sample level editing of your waveform. Though not as useful for general waveform drawing, it can be very useful for eliminating tiny clicks and pops that will show up if you are zoomed in closely on the waveform. It will not become active until you've zoomed into this level. You can access this feature by using the Pencil icon located in the toolbar or using the **Ctrl + E** Hotkey.

Remember, you must be zoomed in before this feature this will activate. Watch your pencil icon; when you've reached the correct zoom in resolution, it changes from grayed-out to active.

Manually De-clicking With the Pencil Icon (Tutorial)

1. Zoom in on your problem area using the Zoom icon (a click will look like a sharp mountain peak).
2. Single click on the pencil icon with the left mouse button. The button will remain down and the pencil will be movable via your mouse.
3. Move the pencil along the waveform area and depress the left mouse button to make the pencil write a new signal.
4. Sometimes, just drawing a flat line in place of an impulsive "click" will remove the click and leave no discernable audio residue, especially if the event only lasts a few samples in length.

Mute

This feature uses direct hard disk editing to allow you to mute a selected portion of your .wav file. With Mute, the audio file sound level is reduced to nothing in the selected area, but the area remains as part of your file. The Mute feature is also useful for getting rid of noise at the beginning or the end of a recording. You can access this feature via the **Ctrl + M** Hotkey or by depressing the right mouse button and using that menu. The Mute function does not alter the length of your file.

Important Note:
Do not mute the beginning or the ending of a .wav file before operating any of the Impulse Noise filter(s). Doing so will cause them to function at an extremely slow rate of speed during the muted section, because it will have a very difficult time calculating a signal to noise

ratio on a signal containing all zero’s. Perform the .wav file muting function after all other filter operations have been completed.

Muting Procedure (Tutorial)

1. Determine the position in the Source or Destination workspace in which you desire to apply the Mute Function and highlight that area.
2. If you change your mind regarding the section that you desire to mute, merely double click the left mouse button anywhere in the workspace area, and the entire file will again become highlighted in yellow. Then, repeat step 1.
3. Click on "Mute" (Edit Menu), and a dialog box will appear which says "Mute will set the selected section of the file to silence. Do you want to continue?" With the left mouse button, click on either "Yes" or "No."

Manual De-Clicking with Mute or Interpolate (Tutorial)

1. Listen to your .wav file using the Play feature, and determine the location of the click, pop, or thud that you desire to eliminate.
2. Zoom-In on the section of the .wav file containing the click using the Zoom In feature.
3. Continue Zooming-In alternately listening to the .wav file until you see the troublesome artifact in the DCArt10/DC Forensics10 workspace. It will take some training to be able to identify transients visually, so be patient during your learning curve.
4. Using the left mouse button, highlight the transient event, being careful to highlight the complete event, and not just a portion of it.
5. Click on "Edit."
6. Click on "Mute" or use the “Paste/Interpolate” function or depress the “I” (Interpolate) hotkey
7. Zoom back out and listen to the .wav file.

Though you are sometimes actually muting a segment of the audio file, if you use this method carefully, you may not hear it in the final playback if the event is short-lived.   The Interpolate method (“I” or

“O” hotkeys) will generally yield a more transparent result compared to “Mute”.

Fade-In

Fade-In does what the name implies when applied to the beginning of a .wav file. You can choose between linear or logarithmic envelopes, and you can also choose the time period for the Fade-In by selecting the portion of the .wav file over which you desire the Fade-In to occur. Lastly, you can choose the "start level" for Fade-In as well as the "stop level." ("Level" is the start and stop loudness for the Fade-In.) Fade-In operates on the selected file (which can be the Source or Destination file).

Fade-In Procedure (Tutorial)

1. Listen to the beginning portion of your .wav file and determine the position near the beginning of your .wav file during which you desire to produce a "Fade-In" effect.
2. At the beginning of the sector of the .wav file that you desire to apply the "Fade-In" effect, click down on the left mouse button and keep holding it down as you drag the timing bar (using the mouse) towards the right of the workspace.
3. Stop dragging the mouse when you arrive at a location just prior to the actual beginning of the signal portion of the .wav file.
4. Release the left mouse button. You will notice that the section between the two timing bars will remain highlighted in yellow. This is the sector during which you have chosen to apply a "Fade-In" effect.
5. You can click the right mouse button to hear if you have chosen the correct portion of the .wav file to apply the "Fade-In."
6. Click on the Edit Menu function.
7. Under the Edit menu, Click on “Fade-In”.
8. Choose the type of Fade that you prefer; either Linear or Logarithmic.
9. Set the "Start Level" slider to zero gain (all the way down). The default setting for this control is zero gain.

10. Set the "Stop Level" slider to 0 dB (unity gain). The default setting for this control is unity gain.
11. Click on OK. The "Fade-In" function will be performed on the chosen portion of the .wav file.

Note: After a Fade-In has been performed; there may be a sector of your .wav file containing some noise at the very beginning just prior to the start of the Fade-In. This can be eliminated with the Mute function.

Fade-Out

Fade-out also does what the name implies. It contains all of the features outlined in the "Fade-In" description except that it normally works near the end of the file. Fade-Out runs under the Edit menu, and unlike the various Filter functions, operates directly on the selected file (which can be the Source or Destination file).

Fade-Out Procedure (Tutorial)

1. Determine the position near the end of your .wav file where you desire to apply the "Fade-Out" effect.
2. At the beginning of the sector of the .wav file where you desire to apply the "Fade-Out" effect, click down on the left mouse button and keep holding it down as you drag the timing bar (with the mouse) towards the right of the workspace.
3. Stop dragging the mouse when you arrive at a location in the file where you want total silence to occur.
4. Release the left mouse button. You will notice that the sector between the two timing bars will remain highlighted in yellow. This is the sector during which you have chosen to apply the "Fade-Out" effect.
5. You can click the right mouse button to hear if you have chosen the correct portion of the .wav file to apply the "Fade-Out."
6. Click on the Edit Menu function.
7. Under the Edit menu, Click on Fade-Out - - -
8. Choose the type of "Fade-Out" which you prefer, either Linear or Logarithmic.
9. Set the "Start Level" slider to 0 dB (unity gain). Unity gain is the default setting for this control.

10. Set the "Stop Level" slider to the zero gain position (all the way down). Zero gain is the default setting for this control.
11. Click on OK. The "Fade-Out" function will be performed on the chosen portion of the .wav file.

Note: After a "Fade-Out" has been performed, there may be a sector of noise after the "fade-out" and the end of your .wav file. This can be eliminated with the Mute function.

Single file Operations

Because of the nature of several Diamond Cut operations, the Cut, Copy, Paste, and Fade menu items operate on the file that is currently selected. This means that a Cut will delete a section of the Source or Destination file if it is the currently selected file in the workspace. Likewise, a Fade operation will modify the highlighted section of the selected file (Source or Destination).

Snap Selection to Zero Crossing

Prevent Editing Glitches with Snap To Zero Crossing

The Snap Selection to Zero Crossing editing feature takes the beginning and ending of a highlighted section of a .wav file and moves both of them to the closest zero crossing point. This is used to minimize the introduction of transients, which could be produced at editing points. This feature is only completely effective on monophonic files since stereo files rarely share zero crossing points on both channels on a highlighted section of a file. Thus, on stereo .wav files, the “Snap” feature moves the highlighted area to the closest average zero crossing value between the two channels. The “Snap Selection to Zero Crossing” feature can be invoked from the Edit menu, via the right mouse button or via the “Q” hotkey on your keyboard. To operate this feature, simply zoom in and highlight the desired section of the file and invoke one of the three options just mentioned above.

Delete All Temp Files

Delete All Temp and Fast Edit Files With One Click Of The Mouse

This command does exactly what the name implies to the temp files in your Temp file (dctmp) directory. Think twice before invoking this command to assure yourself that you are not deleting something important! It does not delete all .pkf (peak) files and .ses (Fast Edit History) files.

Gain Change

DCArt10/DC Forensics10 provides a Gain Change feature that is useful for correcting loudness deficiencies on recordings, or to provide the additional headroom required before running the graphic equalizer filter (or any other filter or effect that increases signal loudness). Gain Change can be very creatively applied using the contour graphical interface (gain in dB vs. time in percentage of the file length). It can also be utilized globally on a file to amplify or attenuate a signal, or selectively to bring out a weak vocal, etc.

The following is a summary of the control parameters and the range of adjustment provided for the Gain Change algorithm:

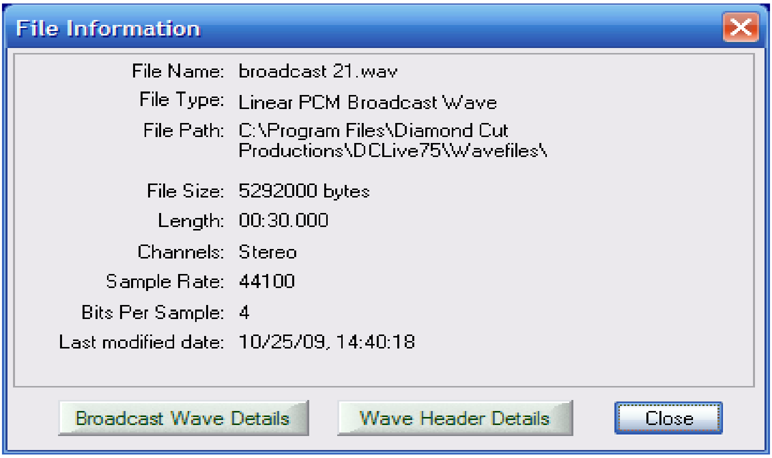
A. Type (Fade In / Fade Out / Gain Change)
B. Slope (Linear / Logarithmic / Curve)
C. Gain Ranges:
    1. +20/-100 dB
    2. +/-20 dB
    3. +/-10 dB
    4. +/-3 dB
D. Start Level (dB)
E. End Level (dB)
F. Shape (Gain vs. Time):
    1. Straight Line: (2 Green Cursors) (start and end gain values)
    2. Curved Line: (4 Green Cursors) (curvilinear inflection point controls)

The Gain Change graph shows you how you have programmed the gain to change as a function of the selected .wav file time axis. You can use the mouse to drag the two (or four) green control points to establish the time relationship that you desire. Often, a flat line is appropriate; however, sometimes the loudness of old 78s decreased toward the end of the recording by a few dB. This can be corrected by a gain correction starting at 0 dB and ending with perhaps 3 dB (depending on the severity of the problem). The reason this occurred is that the early cutting lathes did not provide automatic gain (or frequency response) compensation controls. When the curve shape is selected, two additional green cursors appear. The two additional green cursors can be moved both vertically and horizontally allowing you to create numerous curvilinear gain vs. time relationships.

Important Warning:
Digital systems, like analog systems, can be overdriven to the point of "clipping" the signal. This will produce non-desirous distortion (except on rock n' roll). Before applying a gain increase to a .wav file, study the amplitude of the signal and be sure that you are not adding so much gain as to exceed the dynamic range of the system which is most often set to 2^16 LSB's (or whatever the resolution of the recording that you have digitized). If you do, signals will appear to flatten out horizontally at their peaks on the Source or Destination Workplace displays. If you do indeed notice clipping after a gain change, you can "Undo" this function.

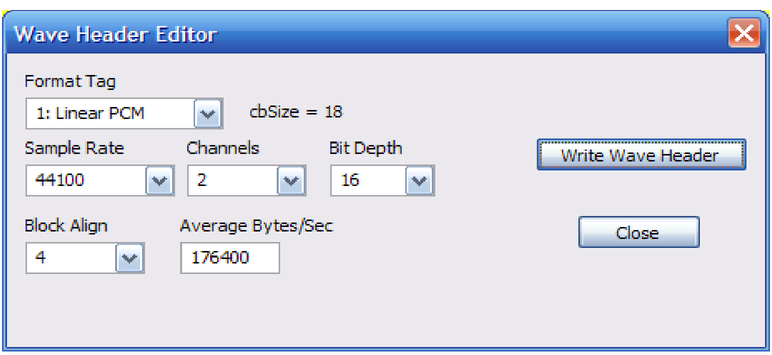
File Properties

The File Properties feature shows you information concerning the file that is opened in your Diamond Cut software. The Figure below shows an example of the sort of file header data presented and/or editable.



The File Properties / File Information Dialog Box

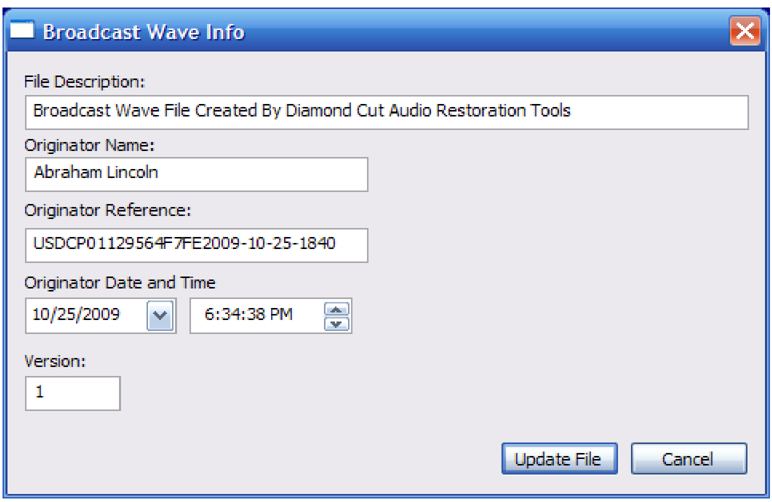
Noteworthy is the fact that this information is also available under the View Menu. However, it can only be edited here via File Properties as found under the Edit Menu. If you want to Edit the .wav header, click on the "Wave Header Details" button and a screen similar to this one will appear:



The Wave Header Editor Dialog Box

From this dialog box, you can edit/change some of the parameters associated with the opened .wav file. In general, it is not desirable to do so unless you are thoroughly familiar with .wav file structures and desire to accomplish some very specific task that could not otherwise be accomplished. After you have changed the desired parameters, click on “Write Wave Header” and the changes will take effect. You can use this feature to repair corrupted .wav file headers.

Your Diamond Cut Software also supports the Broadcast Wave (BWF) .wav file format. BWF facilitates metafiles in conjunction with .wav files. More information on BWF can be found in the Glossary of Terms section of this users guide. You can use the “Save As” command found in the File Menu to create this sort of a file. If you want to View or Edit a BWF, click on the “Broadcast Wave Details” button and the following Dialog box will appear:



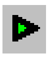
The Wave Header Editor Dialog Box

As you can see, there are a number of data that you can add to the .wav file header not associated with normal .wav files. After you edit your BWF, click on “Update File” and the file header will be modified.

Playback Controls

If you are new to audio processing on a PC, fear not. The playback controls will be a few familiar friends that you’ll recognize instantly on your tool bar. These tools closely match their “real world” counterparts on every cassette or tape deck that you’ve ever seen. They simply help you quickly and easily navigate through your .wav file in any direction and speed you choose. Because the laws of hardware do not bind us, we’ve been able to improve these controls and make them more accurate and useful than their hardware brethren.

Play

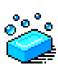


Click here to play your file. If the display is enabled, the highlighted portion shall be played. If the entire file is highlighted, the entire file will play. A shortcut alternative to access this command is via the spacebar. There is also a **Play+n** key (located to the right of the loop-play button) which will play the highlighted portion of your file including approximately 1 second on each side of the highlighted sector. So, play will commence 1 second before the highlighted sector and end one second after the highlighted sector.

Loop Play

The Loop Play key is located just to the right of the Play button. Clicking on it will cause the system enter into an infinite loop-play of the highlighted area of the file. Use the stop key to terminate this function or click on the loop play key again to force it to stop. The spacebar can also be used to stop a loop play operation.

Scrub Audio



The “Scrub Audio” (scrubber) feature (when enabled) allows you to use your mouse to play a portion of a .wav file with its playback speed being proportional to the left or right position of your mouse. It provides you with a method to find a cue point more easily than via the standard “Play” and “Pause” controls. It is also very useful in file transcription work and also can be used to correct the speed of a file having a variable speed error “on the fly”. To use this feature, just point the mouse where you want to commence play and then click down on the left mouse button. Move the mouse left or right to change the speed of playback. Moving your mouse towards the right advances playback in the forward direction while moving it to the left reverses the direction of playback. Please note that this feature only works with WDM soundcard drivers. It will not work with MME drivers.

Rewind

You can click here to move the cursor backwards to the beginning of the highlighted area of a file. This function does not operate when previewing a filter or effect. It works in conjunction with the play and pause buttons. To use it, first pause play with the pause button on a highlighted file portion. Then, click on the rewind button. Next, when you click on the play button, play shall commence at the beginning of the highlighted area of your file.

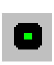
Pause

This is the key with two vertical lines contained within its perimeter. When activated, the playback of the .wav file will pause at that location. Play can be resumed by activating the play button, depressing the spacebar on your keyboard or by simply hitting the Pause button again. Essentially, the spacebar "toggles" your system between "Play" and "Pause".

Fast Forward

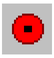
You can click here to move the cursor forward within the highlighted area of a file. This function does not operate when previewing a filter or effect.

Stop

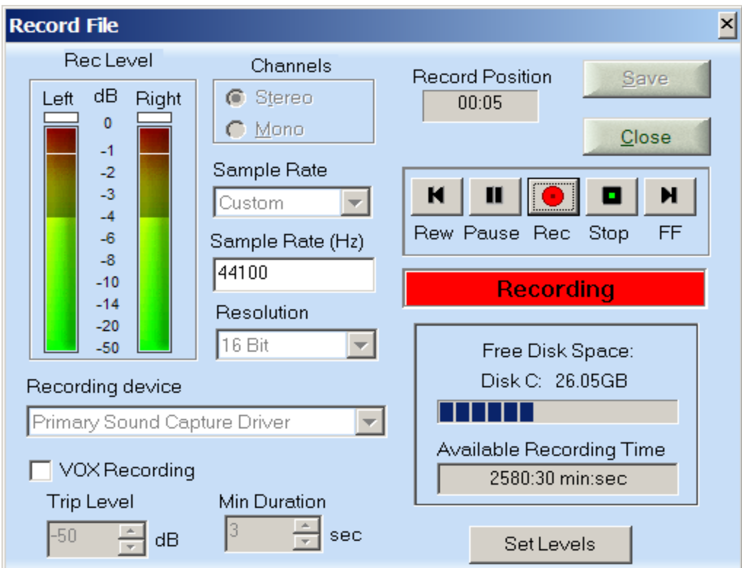


This is the key with the black square with a green dot contained within its perimeter. It is used to stop either a record or a playback session. Unlike the Pause feature, hitting the play button will commence play from the beginning of the highlighted area of the file thereafter.

Record



This is the button with the red square. It is used to place the DCArt10/DC Forensics10 program into Record mode. You can also access this function by hitting the **Ctrl + R** Hotkey. Pressing either the hot keys, menu item or clicking on the Record button will activate the Recording Window. From there, you see the following dialog box:



The Recording Window

- **Recording Level Meters**- Just like on a standard audio or video recorder, you press the Pause button on the Record window, start your source audio playing, and these meters will indicate the amount of signal flowing into your sound card. No level adjustment can be made here. Simply go to your sound card's mixer to increase or decrease the signal.
- **Recording Device**- If you've set up your sound card(s) correctly, this list will display them and allow you to choose your desired recording device. If nothing is displayed here, it's time to revisit both Windows and your sound card

installation instructions. The recording device setup is also available under the Edit/Preferences/Soundcard pathway.

- **VOX Recording**- This check box activates VOX recording. If you want to trigger your recorder automatically on some sort of input signal, this is where it happens. You simply set a Trip Level that is measured in dB. Record will remain paused until a signal above the level you've indicated is received on input. Record will stop when the system senses no input at that level for a period determined by the "Minimum Duration" settings. For more info, see below.

  Trip Level - Threshold Range is from +1 dB to –100 dB. This "trips" the system into recording when the desired level is reached.

  Minimum Duration - Time Range is from 0 to 360 seconds. After the system begins to record, it will stop automatically when the desired signal stops occurring for the amount of time you've dictated in this field.
- **Channels** - Allows you to select Mono (1 channel) or Stereo (2 channel) recording
- **Sample Rate** - Allows you to choose your desired sampling rate. Here are the choices:
  a. 192.00 or 96.00 kHz- Normally associated with DVD recording and playback, these sampling rates provide bandwidth well beyond the audio spectrum and consumes disc space at very high rates.
  b. 48.00 kHz- This rate will produce slightly higher values of bandwidth than 44.1 kHz, but is primarily used for pro-audio applications. It's effective top-end bandwidth is just shy of 24 kHz.
  c. 44.10 kHz- This rate will produce a full 20 kHz recording bandwidth, and it consumes disk space at a moderate rate - - - 5.29 Mbytes per minute per input channel. This is the same sampling rate used on commercial Compact Discs.
  d. 22.05 kHz- This rate will produce a recording bandwidth of around 10 KHz, which is adequate for the restoration of old acoustical recordings. This setting consumes disk space at the rate of 2.645 Mbytes per minute per input channel.

e. 11.25 kHz- (This rate will produce a recording bandwidth of only 5 kHz, which is useful for the restoration of old spoken word recordings, telephone conversations, and first generation movie soundtracks.) It consumes disk space at the lowest rate of only 1.3225 Mbytes per minute per input channel.
f. Custom- Using this setting, you can fill in any sampling rate you choose under the Custom "Sample Rate (Hz)" field via your keyboard. This could be useful in some situations, provided your sound card can support anything but the standard sample rates. The range of sample rates which can be entered as a Custom value is any integer extending from 100 Hz up to 210,000 Hz (210 kHz). Your sound card/sound card driver may limit this range of capability.

- **Sample Rate (Hz) -** If you choose "Custom" sampling rate, this is the field where you can manually type in the sampling rate you want. (Your sound card must support this rate, however in order for it to work properly.)
- **Resolution -** This is where you'll choose the bit width of your recording. You can choose from the following:
  a. 8 Bit (used mostly for Forensics & Games)
  b. 16 Bit (most common - CD Quality PCM Audio)
  c. 20 Bit (un-common professional grade audio)
  d. 24 Bit (common professional grade audio)
  e. 32 Bit Float (Floating Point for special applications)
  f. 32 Bit Int (Integer for special applications)
- **Record Position -** This window, like the tape counter on your tape deck, simply tells you where you are in your recording.
- **Transport Controls -** Just like a common tape recorder, you can Rewind, Pause, Record, Stop, and Fast Forward in your recording.

- **Current System Status -** This gray box normally defaults to the "Stopped" position, but also turns Red while recording and Yellow on "VOX Waiting" mode. It is shown in Green when in the "Paused" mode.
- **Free Disk Space -** Tells you how much space is available for recording on your hard disk.
- **Available Recording Time -** Does a quick calculation on the fly of your hard disk space, sampling rate, bit width and gives you an estimate of how much recording time remains.
- **Spacebar Control -** Normally, the spacebar controls audio playback. However, when the record dialog box is showing, the spacebar reverts to a toggle function alternating between record and record-pause mode of operation.

VOX Recording

As indicated above in the Record window, **VOX** (**V**oice **O**perated **X**mit {transmit} – an old military communications term), or signal activated recording, is possible with DCArt10/DC Forensics10. This feature allows the Diamond Cut recorder to start and stop itself based on the presence or lack of presence on an audio input signal. The recording activation process is triggered by signal level sensing as determined by your setting of the Trip Level control. The record de-activation process is triggered by a combination of the signal level sensing system in conjunction with the Minimum Duration control setting. For example, you will want to set the Minimum Duration to a value long enough so that the recorder does not stop between songs on an LP. In that example, you would probably want to set the control for about 10 seconds. To use the VOX feature, bring up the Recording dialog box and check **VOX** recording and set the Trip Level and Minimum Duration appropriately for your specific application.

In the DC Forensics10 Version, you can also perform automatic time stamping of **VOX** events when you have the Log to Disk function selected (see the Live Mode section for more information). This function adds a time stamp for each recording event and is accurate to less than 1 second. It is very useful for Forensics surveillance recording situations.

Extended Recording

Record Material of any Length- No More 2 Gbyte Wave file Limit

Wave files are limited in length to 2 Gbytes. In some situations, it is desirable to record for a period of that exceeds this 2 Gbyte limit. The Extended Recording Feature accommodates this situation. It works by automatically opening a new file when a particular file approaches the 2 Gbyte limit. When you save the extended file(s), you will see that the file(s) will be named in the following format in your directory of choice:

xxx_part1.wav
xxx_part2.wav
xxx_part3.wav
xxx_partn.wav

Each file will have to be played or processed independently thereafter.

Recording Signals onto your Hard Drive (Tutorial)

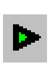
1. Before making your first recording, please quickly review the tutorials for “Getting Started”.
2. This process needs only to be performed once, if you only have one sound card. Your chosen setting will be saved. However, if you have multiple sound cards, this process will have to be repeated anytime you desire to change the input or output configuration to DCArt10/DC Forensics10.
3. Now, click on the Red Record button found on the toolbar. You will see the Record dialog box appear on your screen. Determine your settings and select the desired recording device from the drop down window.
4. Click on the Pause (Record Pause) button in the recording window. If your source is analog, adjust the output level of the sound source or the input sensitivity of your sound card until the green VU meter bar graphs are modulating vertically to the maximum degree possible short of activating the red overload indicators. Please note that this adjustment may consist of a hardware control (gain or volume) of the output signal feeding into your sound card OR you may use the input mixer of your sound card to perform this function. Sample the

loudest portion of the sound source to assure that no overloading will occur when the transfer is made to your hard drive. Please note that digital sources (like SPDIF inputs) are not gain adjustable. Whatever the gain settings that were used initially on this type of source, translates to the gain that you will get when transferring to the hard drive via DCArt10/DC Forensics10. If the original analog to digital transfer was either overloaded or under recorded, it will remain that way during the transfer to your hard drive. Gain corrections (or de-clipping) can be performed at a later time.

5. To commence recording, simply click on the Record button.
6. The "Record Position" is analogous to the tape counter of a conventional tape recorder. It uses real-time measurement. (Minutes: Seconds)
7. To pause the recording, click on the Pause button on the toolbar or in the Record dialog box. You may continue to record from the pause position by repeating the method outlined in step #5.
8. Recording can be terminated by clicking on the Stop button. (Stop is the square button containing a smaller black square within its perimeter).
9. To save your recording, click on the Save button in the "Record File" dialog box. You will notice that a name has already been assigned to your file by the software program. You can either keep the assigned name for your file, or rename it at this time.

Important Note:
DCArt10/DC Forensics10 is compatible with .wav files that were originally created by other programs. It is not necessary to record your .wav files using the Diamond Cut recording routine in order to use the programs processing features.

Play



This is the key on your toolbar with a black and green arrow, which is pointing towards the right. This key is used to play your file. The playback will begin at the leftmost portion of the yellow highlighted area of your .wav file. You can also access this function by pressing your keyboard spacebar.

Playing a Wave file (Tutorial)

1. Launch DCArt10/DC Forensics10.
2. If you haven't already done so, you must define your Output Device.
   A. To do so, click on “Edit /Preferences/Soundcard and then “Device I/O Selection”.
   B. Choose the output device that you desire, and then click on OK.
3. Next, click on “File”, and then on “Open” and browse to the file that you wish to play.
4. You will notice that the waveform will appear in black on a yellow background. This area of the window is the Source workspace. If your .wav file is monophonic, you will see just one waveform displayed in the workspace. If your .wav file is stereophonic, then you will see two waveforms displayed, with the top waveform representing the left channel and the bottom waveform representing the right channel. The length of your .wav file is displayed in Minutes: Seconds: Milliseconds in the timeline at the bottom of the Source Workspace. It’s a good idea to also employ the Time Display window by clicking on View and then checking Time Display. This feature makes time measurements within your .wav file quite easy to do and with a very high degree of precision.
5. To play the file, click on the Play button on the toolbar or depress your keyboard spacebar. You will notice the playback cursor begin to "march" across the workspace as the system plays the file.
6. To terminate playback, click on the stop button on the toolbar or hit the space bar again.
7. If only a portion of the .wav file had been highlighted for playback, and now you desire to playback the entire file, double click on the workspace background, and the entire file will become highlighted, and ready for playback.

Pausing and Resuming Playback

1. To pause playback, click on the Pause button on the toolbar.
2. To resume playback, you can either click on the Play button, Pause button again or depress the keyboard spacebar. If the play button preference (in Preferences) is set to “Play selected area”, the playback will resume from the beginning of the selected area if the

space bar is pressed; the spacebar performs a stop and then a rewind.

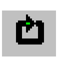
Note 1: Toggling with the “Loop Play” button behaves like toggling with the spacebar including the rewind to the beginning of the selection operation.

Note 2: If “Play from Cursor” is selected (in Preferences), playback will resume from the stopped position.

Playing and Pausing using the Right Mouse Button

1. To start the playback of your .wav file at a specific location, use your mouse to move the mouse arrow pointer to the desired location in the Source or Destination workspace.
2. Single click the right mouse button and you will notice a "Play from here" option in the menu. Click it and “Play” shall commence from that point forward.
3. The file will continue playing until its end unless you click the right-hand mouse button again anywhere in the Source or Destination workspace. Playback will terminate the moment the right-hand mouse button is clicked.

Play Looped



This button allows you to play either the entire file or a highlighted area over and over again, back to back. Playback will continue until the button is depressed again or the space bar is pressed. You can also access this feature by using the **“L” Hotkey**.

Note: This feature will not work if the “Play from cursor” method of playback is selected in the Preferences menu.

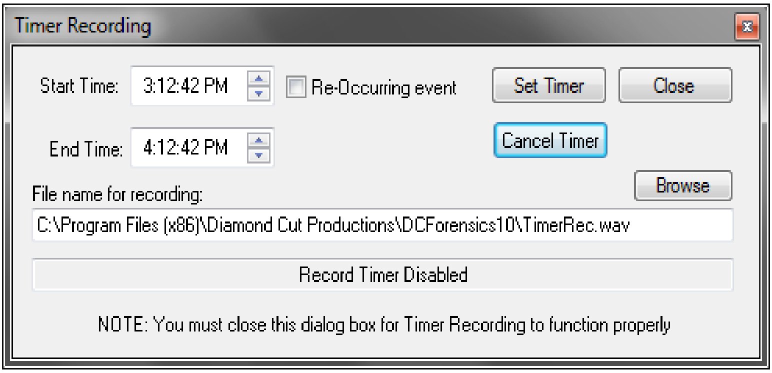
Time Bracketed Play Range

This feature allows you to play a range of time around a selected area. It is useful when you need to listen for small anomalies on a sound recording in order to find their exact location(s). You can choose between the following, which can be activated from the keyboard using the following keystrokes (shown in brackets): To use this function, select an area by clicking and dragging. You can play this selected area by using the spacebar or the Play button. You can also play the selected area PLUS a specific amount of time in front and behind the selected area by simply hitting the 1 thru 4 keys on your keyboard. For example, if you select an area 10 seconds in length, hitting the 2 key will play starting at a point 2 seconds before the selected area and continue for an area representing 2 seconds after the selected area.

Play Range +/- 0.5 seconds: (Keyboard Key # 1)
Play Range +/- 1 second: (Keyboard Key # 2)
Play Range +/- 2 seconds: (Keyboard Key # 3)
Play Range +/- 4 seconds: (Keyboard Key # 4)

Timer Recording

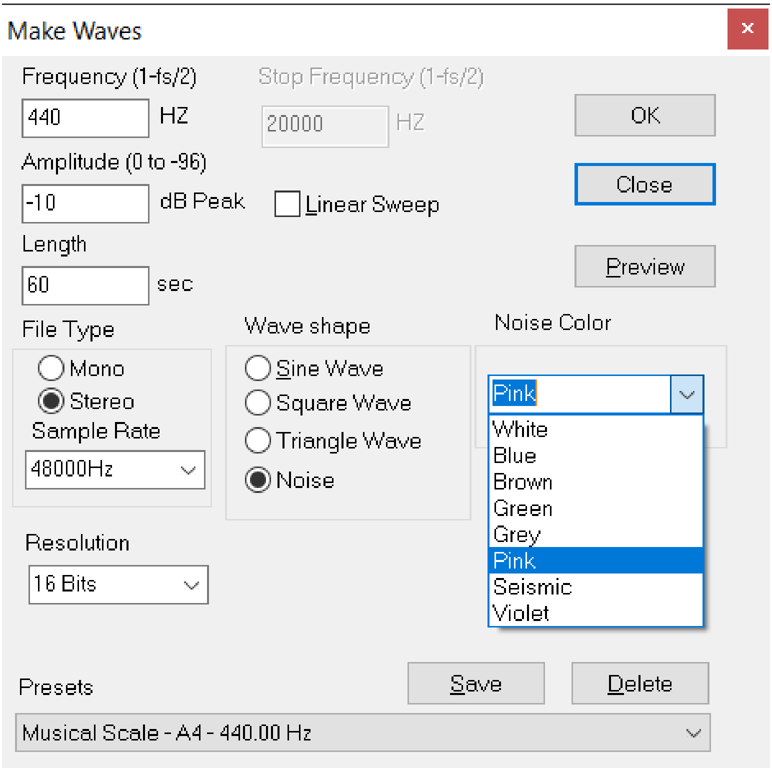
The Timer Recorder feature found under the Edit Menu is useful for such applications as recording your favorite radio broadcast (sometimes referred to as “time-shifting”) at a defined time, or to commence a surveillance recording based on a particular time of day. It is a single event 24-hour timer, which clears itself after the event has passed, or can repeat its action the following day by checking the “Re-Occurring Event” box. To set a “Start Time”, click on the number to be changed in the “Start Time” field, and use the up and down arrows until the readout agrees with the desired start time. The same process is used for setting up the timer’s “End Time.” The Re-Occurring Event checkbox will cause the recorder to record the same time span every day. Use the “File Name for Recording” in conjunction with the “Browse” feature to define the file location and name into which the recording shall be placed. ***You must leave the software running for the Timer Recorder feature to work and you must close the Timer Recording window***. You can minimize it, but do not exit the program. If you want to disable the Timer, click on the “Cancel Timer” checkbox.



The Timer Recording Window

Click on the red record button found on the tool bar in order to set your recording parameters such as sample rate, resolution (bit depth), channels (stereo or mono), and sound card location. The record dialog box will appear on your screen. Then, select the parameters that you desire and close the recorder. Those values will be saved and used by the Timer Recording feature.

Make Waves Signal Generator



The Make Waves Signal Generator

The "Make Waves" signal generator function provides you with comprehensive capabilities associated with a fully programmable audio signal generator (sometimes referred to as a function generator by engineers). It is also a "Make Noise" generator as well. It can produce Sine Waves, Square Waves, and Triangle Waves of adjustable frequency and amplitude. The frequency of those periodic waveforms can also be swept from one value of to another. The Make Waves generator is also capable of producing pseudo random noise signals having various spectral distributions as defined by their "color". The basic random noise signal that the Make Waves generator produces is

White noise, but you can also synthesize Blue, Brown, Green, Grey, Pink, Violet or Seismic noise by choosing the appropriate noise checkbox. These noise distributions can be produced directly in one step. (Various Noise distributions can also be produced from a White noise signal source in conjunction with the appropriate Multifilter preset if desired.) When the Make Waves generator is in stereo mode, the random noise signals will be incoherent with respect to one another. (The noise signals are created by two independent noise sources and not just one, thus yielding random phase relationships between the two channels). Make Waves can produce a file containing the signal or noise of interest, or it can produce the chosen signal in real time by way of the Preview button (to stop a real time Preview, click on the Preview button again) or use the Cancel Button found on the progress bar dialog box. The Preview Button is essentially a toggle switch (on/off). The Make Waves generator is useful for calibrating and verifying the performance of the audio equipment used in your sound restoration laboratory, especially when used in conjunction with the onboard Spectrum analyzer, Spectrogram, Histogram, VU meter or the X-Y plotter. These tools will also be useful to help you better understand the functionality of some of the filters provided in the Diamond Cut software program. The Sweep and Random generator are especially useful for characterizing the frequency response (and/or transfer function) of electrical and acoustical systems when used in conjunction with Diamond Cut measurement tools like the VU Meter, X-Y plotter, Spectrum Analyzer or Spectrogram. The Triangle Wave generator is useful for characterizing the time-domain linearity of an audio system or component. Included with the Make Waves Generator presets are all of the musical notes running from C0 to D9# (referenced to a 440 A4 of the equal-tempered scale) as well as a number of useful test signals from which to choose.

Expanded functionality can be obtained when the Make Waves Generator is used in conjunction with the Edit/Paste/Mix function found under the Edit menu. You can mix an unlimited number of waveforms together using the Paste Mix function to create complex waveforms of any shape or other signals containing noise plus specific waveforms. You can further enhance that capability of this system using the Pencil editor, but you will first have to zoom-in on the section of the file you want to manually modify. Burst signals can be created by using the Edit / Mute function on large parts of the file and variable amplitude signals vs time can also be created with the Edit / Gain Change or the Edit / Fade In or Edit / Fade Out functions. Interestingly, you can create a combination of swept amplitude and swept frequency (AM – FM) signals together, just as an example of the overall capability of the system.

The following controls with their adjustment range are provided within the Make Waves Generator:

1. Start Frequency: 0.01 Hz to 100,000 Hz. (100 kHz)*
2. Stop Frequency: 0.01 Hz to 100,000 Hz. (100 kHz)*
3. Length: (Duration of the tone burst) 10 Millisecond to 600 seconds. (Data entry is in seconds.)
4. Amplitude: 0 dB to -145 dB
5. Linear Sweep check box (on or off)
6. Sine, Square, Triangle Wave or Random (white noise)* selector
7. File Type (Stereo or Mono)
8. Sampling Rate {Factory Pre-Programmed} (8, 11.025, 22.05, 44.1, 48, 88.2, 96, 176.4 and 192 kHz)
9. Sampling Rate {User Programmable} (100 Hz to 210 kHz)**
10. Resolution (8, 16, 20, and 24 bits)
11. Presets: Includes common signals as well as the Musical Scale ranging from C_0 (16.35 Hz) to D_9# (9,956.06 Hz)
12. Noise Color: Chose between White, Blue, Brown, Green, Grey, Pink, Seismic or Violet.
13. Use the OK button to create a file having your selected signal or noise.
14. Use the Preview to hear the selected signal or noise in real time and use the Cancel button to stop it or click again on the Preview button (it will toggle back and forth).

Random Noise Characteristics and Usages

White Noise: White noise is often used to obtain the impulse response of an electrical circuit, in particular that of amplifiers and other audio equipment. It can also be used to evaluate audio computer DSP algorithms and their transfer characteristics. It is used extensively in audio synthesis, typically to recreate percussive instruments such as cymbals or snare drums which have higher noise content in the upper end of their frequency spectrum. It is also used as a sleep aid or as a masking element to help soundproof a room.

Blue Noise: Blue Noise is also known as azure noise and gets it name from the optical field since the color blue is on the higher end of the frequency spectrum of visible light. In audio applications, Blue Noise is used for dithering, a process where noise is added back into an audio signal in order to smooth out the sound and reduce the audibility of various distortion products. Some A-D converters use this type of noise to 'dither' the LSB of the digital signal produced.

Brown Noise: This comes from the physics principle of Brownian motion and diffusion. Its energy falls off with increasing frequency with a 1/f^2 manner. Brown noise can sound like a very uneventful ocean surf since it has more bass content than white noise, making it more pleasant to listen to. It makes for a good sleep aid.

Green Noise: Green Noise is White Noise with emphasis in the mid-band region around 500 Hz. It is good for setting up PA system in which the spoken word need the most energy to cover a large audience. Use this noise in conjunction with a 1/3 octave equalizer to achieve good PA performance for speech oriented and optimized sound systems.

Grey Noise: This distribution of random noise varies with sound level based on how humans perceive "loudness". The Fletcher-Munson contour curves of equal loudness are based on Grey Noise signals.

Pink Noise: Pink Noise contains equal energy per unit octave and is used extensively in acoustical measurements and Public Address system setup and calibration in conjunction with a 30 band graphic equalizer.

Seismic Noise: This is a random noise signal containing primarily sub-sonic frequencies. It is useful in sonar physics experiments as well as a special effect in very large theater sound installations having very large sub-woofers. It is not very audible, but can be physical felt under some circumstances. You need to be careful with using this noise signal since most sound system will not reproduce it, but the amplifier and speaker drivers may still be trying to respond potentially causing damage to the system.

Violet Noise: Violet Noise has a rising power density with increased frequency. The human ear has limited sensitivity to very high-frequency noise (or hiss) and the ease with which white noise can be electronically differentiated (high-pass filtered at 6 +dB/Oct) allowed some early realizations of a dithering process as applied to digital audio using violet noise.

Noise Traits by Color

Color (or type)	**Characteristics**	**Slope**	**Density**
White	Equal Energy per unit Hertz	Flat	Uniform
Pink	Equal Energy per unit Octave	-3 dB/Octave	1/F
Brown	Analogous to Brownian Motion	-6 dB/Octave	1/F^2
Red	Another name for Brown Noise	-6 dB/Octave	1/F^2
Blue	Azure Noise (proportional to F)	+3 dB/Octave	F
Green	Pink Noise with Emphasis at 500 Hz	Non-Linear	Subjective
Violet	Differentiated White Noise	+6 dB/Octave	F^2
Grey	Psychoacoustic equal loudness Curves	Proportional to SPL	Varies with loudness
Seismic	Random Noise below 100 Hz	-3 dB/Octave	Semi-audible
Black	Analogous to black body radiation	Silence	No Signal

***Note 1:** The frequency programming precision of the Make Waves Generator in the DC Forensics10 version is greater than the standard DCArt10 product. Keyboard entry of the desired frequency must be used to attain this higher level of resolution. The actual frequency resolution in the DC Forensics10 Make Waves Generator is 0.01 Hz (10 mHz or milli-Hertz). Thus, values like 59.99 Hz or 60.01 Hz are possible to create with this system.

****Note 2:** To create a Make Waves file having a custom sample rate value, simply highlight the sample rate field and use your keyboard to enter the desired sample rate value. Note that <u>only</u> very high-end soundcards are capable of handling sample rates above 192 kHz (such as 210 kHz). Less capable soundcards may hang your system when presented with sample rate signals greater than their maximum design capability.

The following chart summarizes the Harmonic Distortion components produced by the various waveforms provided by the “Make Waves” generator at a 44.1 kHz sample rate and 16 bit resolution:

Wave-form	**THD %**	**3rd Harm.**	**5th Harm.**	**7Th Harm.**	**9th Harm.**
Sine	**0**	**0**	**0**	**0**	**0**
Square	**44**	**33.3**	**20**	**13.8**	**10.8**
Triangle	**12**	**11**	**4.1**	**2.0**	**1.3**

***Note:**

Using the Making Waves Signal Generator (Tutorial)

1. Click on the "Edit" Menu with the left mouse button.
2. Click on "Make Waves" with the left mouse button.
3. Choose between Sine Waves, Square Waves or Random by clicking on the appropriate box with the left mouse button.
4. Set the desired "Frequency" by clicking on the number(s) that you desire to change with the left mouse button. Use the keyboard to enter the new values with the "Start frequency" slider control. The allowable range is from 0.01 Hz to 100,000 Hz. (100 kHz)
5. Set the "Length" of the file, which you desire to create using the mouse, and direct keyboard entry. Data entry must be in terms of seconds. If you desire 2 minutes, enter 120. If you desire 10 Milliseconds, enter 0.01.
6. Set the "Amplitude" which you desire, anywhere from 0 dB to -145 dB. 0 dB is the largest signal value which you can produce, with the peak value of the waveform being + / - 32,767 LSB's referenced to a 16 bit file.
7. Click on “OK” and a Source file will be created for you containing the signal that you have just defined.

Important Note 1:
If the sweep generator function is desired, click on "linear sweep" and then adjust the "stop frequency" control to the desired value. The generator will then produce a linear sweep of frequencies ranging from the start frequency value to the stop frequency value over the interval of time defined by the "length" control.

Important Note 2: The Sweep and Random Generator is useful for characterizing the frequency response or Transfer Function of electrical and acoustical systems when used with other Diamond Cut Measurement tools (VU Meter, Spectrum Analyzer, Spectrogram, and/or the X-Y Plotter).

Important Note 3:
Pink noise can be created from the random white noise generator by applying the “White to Pink Noise Converter, 20 kHz” to the white noise file or simply by choosing Pink directly as your Noise Color. Alternatively, a white to pink noise converter is also found in the preset list under the Multifilter. First, create a sample of random (white) noise using the Make Waves Generator. Use a sampling rate of 48 kHz for this procedure. Next process the resulting file through the indicated Multifilter preset and the resulting file will be Pink Noise. White Noise can also be converted to various Noise colors by using the appropriate noise Converter in the Multifilter.

Important Note 4:
Brown (Red Noise or Brownian Noise), Violet Noise and/or Seismic Noise can be created in a similar manner as described in Important Note 2. However, the converters are found as presets under the appropriate names in the Multifilter.

Important Note 5:
When planning to convert White Noise to Pink, Violet, Red (Brownian) or Grey Noise, it is very desirable to start with a 48 kHz file in order to obtain the maximum accuracy following the conversion process.

Important Note 6:
A number of factory presets are provided for the Make Waves Generator consisting of commonly used audio signals. You can add to that list as desired.

Change Sample Rate / Resolution

The Change Sample Rate feature allows you to convert a .wav file from any common sample rate to any other common sample rate. It also allows you to convert from any common resolution (or file depth) to any other common value. These features are provided with a user selectable interpolation quality since improved interpolation takes more CPU horsepower and therefore more time to make the conversions. This way, the user can make this tradeoff. The following controls are provided with the associated ranges of the Change Sample Rate / Change Resolution Feature:

New Sample Rate: 11,025, 22,050, 32,100, 44,100, 48,000, 88,200, 96,000, 192,000 Samples/sec.

Resolution: 8 Bits, 16 Bits, 20 Bits, 24 Bits, 32 Bits FP

Conversion Quality: Choose between –

- CD Quality (16 bit interpolation)
- Pro Quality (24 bit interpolation)
- Master Quality (32 bit interpolation)

Note 1: The tradeoff between these conversion quality choices is the time that it takes to make the conversion; higher quality levels of interpolation requires longer processing time.

Note 2: The Conversion Quality used by the Batch Processor is non-selectable and is set to “Master Quality”.

Dither: Choose between –

- None
- Flat Spectrum: White noise is used as the dither. This is usually not the best choice unless you are doing analytical work.
- Triangular High Pass: A shaped noise spectrum that does a good job of reducing distortion. The name comes from the type of digital filter used. The noise is biased to the high frequency end of the audio spectrum and is less audible than the flat spectrum dither.

- Noise Shape 2: Our own proprietary noise shaped spectrum. This distribution has less mid-band noise than the triangular spectrum, but more at the high end of the audio band.

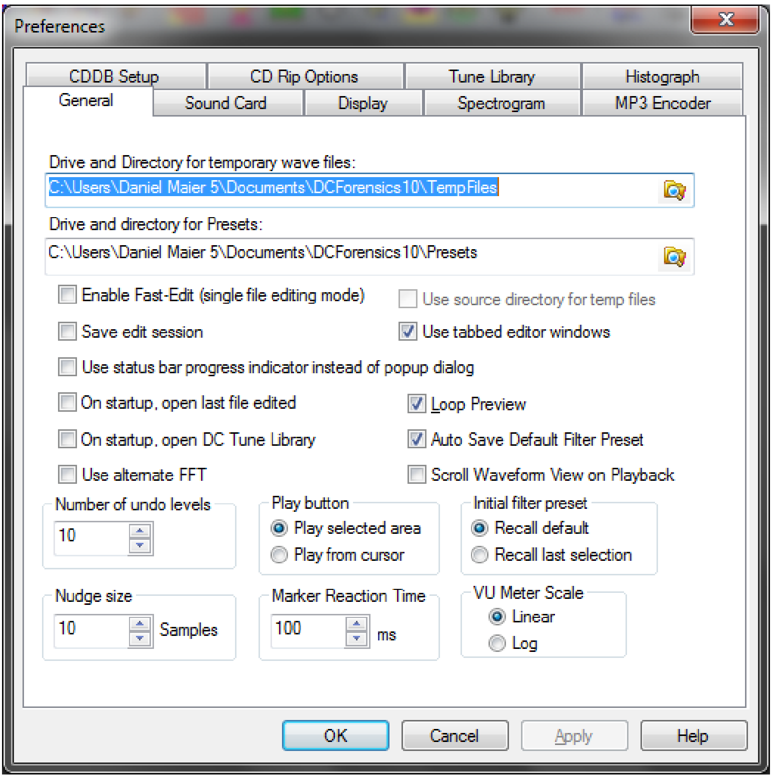
Note 1: In general, choose Triangular or Noise Shape 2 with your ears helping you to make the decision as to what is best.

Note 2: Usually dithering is used when lowering the bit depth…like going from 24 to 16 bits.

To perform a conversion, bring up the file to be converted in the Source window. Select the desired conversion parameters from the outlined choices. Run the filter and the results will be displayed in the Source window after a period of processing time. The file will automatically be re-named by incrementing its name by a value of one, so long as there are no other files with that same incremented name in existence in the chosen file directory.

Preferences

The Diamond Cut "Preferences" menu allows you to customize the system to your own liking.



Configure the System Your Way!

Preferences are broken down into eight sub-categories that can be accessed by clicking on the desired Tab at the top of the Preferences window.

General:

1. Drive & Directory for Temporary .wav files (999 maximum)

2. Drive and Directory for Presets
3. Enable Fast Edit Check box: This turns Fast Edit Mode on or off. If a file is open and you want to change editing modes, your file must be closed before or after switching modes or you must close down the program and restart it in order for this change to take effect. When the Fast Edit mode is turned off, you will be operating in the Classic Edit (Source & Destination) mode.
4. Save Edit Session: When this is checked, you will be queried each time you exit the program as to whether or not you want to save the File Edit History file. If this feature is not checked, the Edit File History file will be automatically deleted when you exit the program. By saving the edit history, you maintain the ability to go back to any previous editing state without destroying the original file. Keep in mind that this action will use more disk space since all of the intermediate edits are stored in temporary files on your hard drive.

Note: *The Edit History (or Session File) is stored in the same directory as the .wav file. It contains the file extension .ses and should not be deleted manually because it will leave the associated temporary files on your hard drive.*

5. Use Status Bar Progress: This is an indicator bar rather than a popup dialog box. When it is checked, the progress of a particular filter will be indicated at the bottom of the screen.
6. Auto Save Default Filter Presets: This feature allows you to choose between the action of returning the filter setting called Default to factory values or to your last personal setting after closing a filter.
7. Number of Undo Levels Saved: This parameter allows you to choose the number of undo levels of destructive editing which DCArt10/DC Forensics10 will maintain as stored files on your hard drive. The default value for this parameter is 10.
8. Nudge Size: This parameter defines the resolution of the left and right arrow keys on your keyboard as they apply to the .wav file-highlighting feature of DCArt10/DC Forensics10. This parameter is defined in terms of samples. After highlighting a portion of a .wav file, you can fine tune or "nudge" the highlighted area using the left and right arrow

keys, and the Shift key. The resolution of each click on an arrow key is defined by the value of “nudge size.”

9. Play button (options): Allows you to choose “Play Selected Area”, which defaults playback to the beginning of a selected area of the beginning of the file each time you press “Play”, or “Play from Cursor”, which, like a tape player, starts from wherever you last played the file.
10. Marker Reaction Time: This parameter allows you to compensate for you and your computer system’s lagging reaction time when dropping markers on the fly using the “M” keyboard accelerator. Its units are calibrated in milliseconds.
11. Loop Preview on or off: This feature, when checked, will cause a “previewed” section of a .wav file to repeat itself endlessly until the filter or effect is canceled or the preview button is clicked on a second time.
12. VU Meter Scale: Choose between Linear or Logarithmic scales. “Linear” produces a scale running from -60 dB to 0 dB and “Log” produces a scale running from -100 dB to 0 dB. (The linear choice produces a displacement function which is analogous to the old electro-mechanical D'Arsonval type VU meters of the past, except that it is in bar graph format rather than arc format.)
13. “On Startup, Open Last File Edited” is an alternative start-up mode that you can choose by checking this box.
14. Scroll Waveform View On Playback: Normally, the play cursor moves across the time domain display as a .wav file is being played and stops at the end of the highlighted area. “Scroll Waveform View On Playback” is an option wherein the display will keep pace with the playback position. As the cursor reaches the end of the display area, the display will page one frame to the right and the cursor will start again at the left hand side. The Zoom level is maintained and can be changed with the Zoom X2 feature while playing. You will need to use the “Play from Here” function found on your right mouse button to use this feature or use the Zoom X2 functions to display an area that is different than the highlighted area.
15. On Startup, Open DC Tune Library: When checked, and the software is launched, the DC Tune Library will automatically be launched too.
16. Use Source Directory for Temp Files: This mode is

particularly useful to Forensics users. It allows all of your work to be put into a single directory for easy archiving. When this mode is not used, it may be difficult to identify all of the files used in a fast edit session because they will be in the temp directory. This is only available in the Forensics version of the software.

17. Use Alternate FFT: This feature is a diagnostic tool used by the Diamond Cut Company for troubleshooting purposes.

Sound Card:

1. Driver Type Check Box: Select between MME and WDM, depending on which driver best works with your particular sound card. If your sound card is older, chances are that the MME selection will work better. If you have a newer sound card, then WDM may be your choice. We recommend always trying WDM drivers first.
2. Preview Buffers: (2 to 50) (Raise this value if you encounter stuttering during preview). This parameter applies to preview mode only. It allows you to choose the size of the buffer space that is used by the preview feature. You can select between 2 to 50 buffers. 1 buffer = 4096 samples. The larger the buffer which you choose, the longer the sample which you will hear before the system repeats itself (stutters) if your system is not fast enough to run a particular algorithm in real time. However, the larger the buffer, the longer will be the delay time (latency) before you hear the results of a preview session. As a general rule, low sampling rates coupled with low bit depths (low resolution) require smaller buffer sizes compared to deep bit depths (high resolution) and high sampling rates. Also, .wav files that are stereophonic require more buffers than monophonic recordings.
3. Input Device Selector: (Browse to your desired sound card Input)
4. Output Device Selector: (Browse to your desired sound card Output)

5. "Re-Initialize on Play" sometimes fixes problems with WDM soundcard drivers, but is not applicable to MME's. If you are observing irrational behavior with your WDM drivers, try checking off this option as it may clear up your soundcard problem.
6. Play Chimes after Long Operation Completes Checkbox: Provides you with a distinctive audible annunciator ('gong' sound) after the completion of long operations such as the Batch File Editor, CD Ripper, CD Burner, and the Data Disk Burner.

Important Note: Most tech support calls that describe "no playback" or "no record" symptoms relate to sound card issues. DCArt10/DC Forensics10 uses all standard Windows calls to playback and record audio. The Diamond Cut software does not communicate with your specific sound card directly. Therefore, any sound card that works under Windows will automatically work with DCArt10/DC Forensics10. If you are having problem playing or recording audio, make sure your sound card is working with other applications and make sure you have the correct inputs and outputs selected under Edit/Preference/Soundcard.

Display:

1. Check "Don't use peak files (.pkf) for display" (no time display shown when checked). *This does not apply in Fast Edit Mode.*
2. Check "Sync Mode Scroll Tracking" (synchronizes Source and Destination displays)
3. Check "Clean Display" which removes .wav file path name from the display
4. Check "Show Splash Screen on Startup" (this feature saves time following software launch when turned-off)
5. Display Colors: This set of parameters allows you to choose the Source and Destination Workspace background, highlighted, and waveform colors. When you click on either parameter, a color palate will appear, and you can click on the color combinations that suit your vision (or personal taste) the best.
6. Display Time Format (X-Axis)*:
    a. Minutes: Seconds

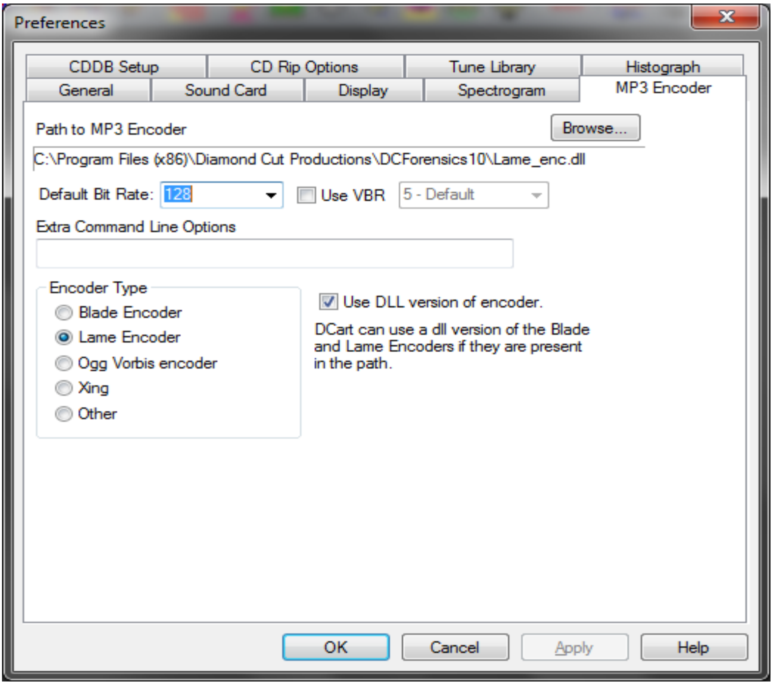
    b. Samples
7. Display X-Axis Time Scale (checkbox)
8. Display Y-Axis Amplitude Scale:*
    a. Samples
    b. dB
    c. Percentage
9. Display Font Size (Enter desired value in terms of "points". The range is from 2 to 16.)
10. Sync Mode Scroll Tracking (synchronizes the Source and Destination mode scrolling in Classic Edit mode).
11. Don't use Peak Files for Display checkbox – turns off the time domain display when checked.
12. Lock Markers – Prevents your Markers from being accidently moved when this box is checked.

*Note: These user selectable parameters are reflected both in terms of the displays but also exportable values where applicable.

Spectrogram:

Note 1: Please refer to the Spectrogram section of this manual for details.
Note 2: Left mouse double clicking in the Spectrogram window can also access the Spectrogram preferences.

mp3 Encoder:

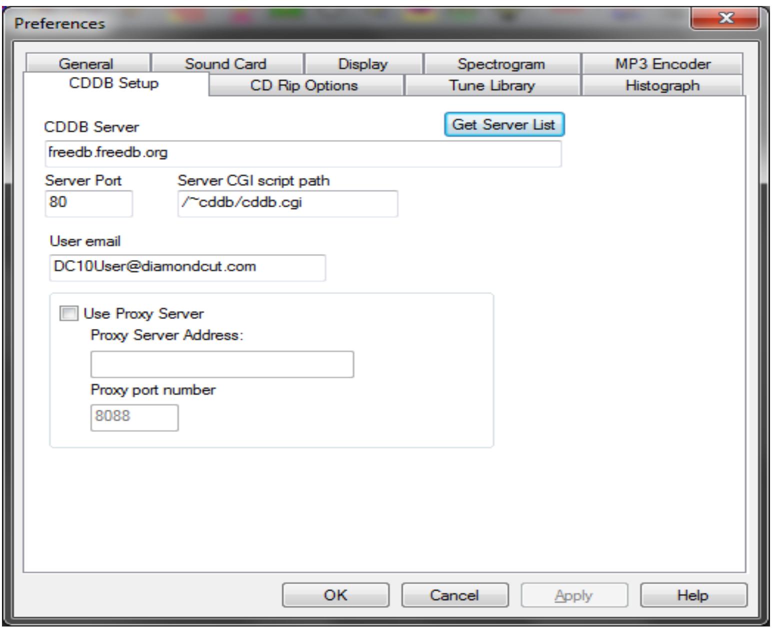


mp3 Encoder Preferences Setup

Setup is provided for an externally provided MP3 Encoder. At this time, we accept popular 3rd party encoders that are available from the Internet. To allow the program to Save or Batch Process files into MP3 format, you must download an encoder and choose it in this screen. The Browse button allows you to show DCArt10/DC Forensics10 where the encoder is located on your hard drive. In general, higher bit rates produce better sonic results at the expense of greater memory consumption. Note that the system supports both CBR (Constant Bit Rate) and VBR (Variable Bit Rate) encoding techniques.

Tune Library:
Please refer to the DC Tune Library section of this documentation for details on these preferences.

CDDB Setup:
Your software has the capability to "Rip" Red Book CD Audio to .wav files. For details regarding the operation of this feature, please refer to the section of the manual called "Rip CD Tracks" which is found under the Filter Menu of the program. The CDDB Setup allows you to establish the parameters so that your system can fetch Track Titles off of the Internet and use them as Track Names rather than the Track Numbers assigned by your Diamond Cut software. Here is a screen shot of the CDDB Setup Tab:

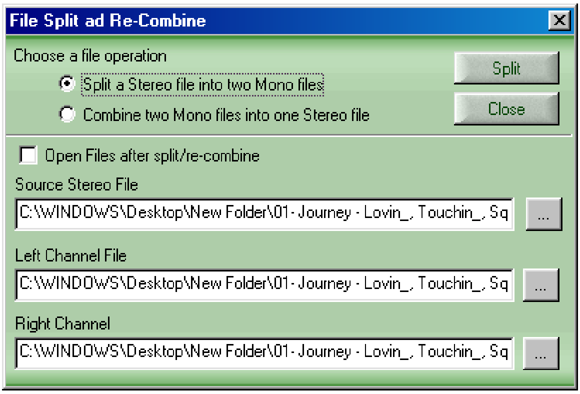


CD Database Setup

You will need to be on the Internet for the CDDB feature to function. Generally, the default settings provided will work for you. However, it is a good idea to click on the "Get Server List" and select the server closest to your locale. If you are running a firewall on your system, you will have to check the "Use Proxy Server" box and enter the proxy server address and proxy port number.

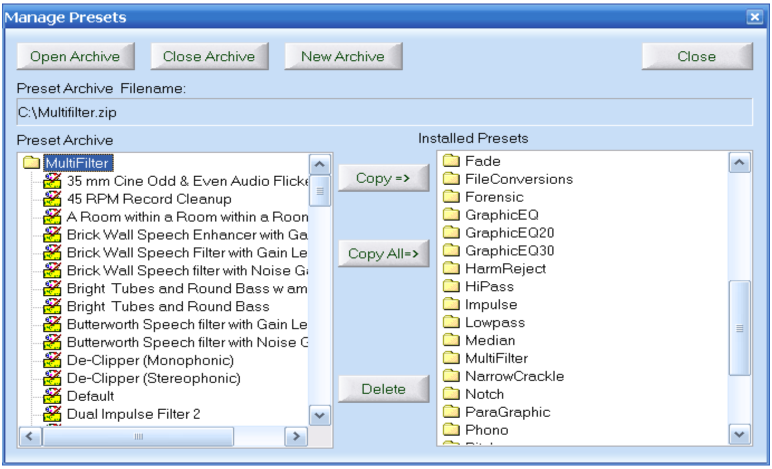
File Split and Recombine

Sometimes it may be desirable to split a stereo .wav file into its left and right components, establishing individual .wav files for separate channel processing. For example, if one of the stereo channels needs some equalization but not the other, using this splitting and re-combining tool can facilitate that function. This easy to use tool allows you to either Split a Stereo file into two Mono files or combine two Mono files into one Stereo file. You can select whether or not to open these files after the process is complete. Note that the L/R buttons on the toolbar allow instant selection of either or both channels for playing and processing.



Easily Split and Recombine files with this feature

Manage Presets



The Preset Manager

The Presets Archive stores all presets, either factory produced or user produced in folders according to their filter (or effect or other) and have the file extension of .pst. A nice feature of DCArt10/DC Forensics10 is the ability to share presets you have created with other users via email, or web pages like the Diamond Cut Users Forum. You can export any or all of your presets to share them with other users by sending them via a single (.zip) file to the Presets sharing portion of the Diamond Cut Users Forum. Go to www.diamondcut.com and then click on the Users Forum link. Once there, you will see the preset exchange area. This exchange provides both uploading and downloading capability with an area for you to describe the preset(s) that you are posting. You will find approximately 200 presets that you can download from that site that have been created and uploaded by Diamond Cut Users over the years.

Your current presets are stored in a Presets folder in the Diamond Cut directory on your hard disk as a single small file for each preset. Since DCArt10/DC Forensics10 comes with over 1000 presets, it would be cumbersome to find and share them by sending or receiving one file for each preset. The Manage Presets window makes this easy. You simply create an Archive by clicking on the button and giving the new archive a name. Now you can copy any individual or group of presets into this archive. Double clicking a preset folder will open it and reveal its individual presets. This archive file will exist as a single filename .zip file on your hard disk and will contain all the presets you copied into it. Copying presets does not delete them from your own preset list. This single .zip file can then be sent to another user and opened using the Open Archive function. The user can then place any or all of the received presets into any of his or her personal preset folders.

Saving a Preset (Tutorial)

1. Establish the desired filter settings and/or states for a particular filter or effect application.
2. Click on the "Save" button.
3. Using your mouse, place the cursor at the beginning of the data entry field, and double click the left mouse button.
4. Delete any characters in the data field with the "delete" key on your keyboard.
5. Type in a descriptive name for your setting (up to 32 characters in length).
6. Click on "OK". Your preset setting will then have been saved.

Recalling a Preset (Tutorial)

1. With the left mouse button, click on the down arrow located on the right hand side of the setting list (located at the bottom of the Filter Dialog Box).
2. With the left mouse button, single click on the filter or effect preset setting description you desire from the list.

Deleting a Preset (Tutorial)

1. With the left mouse button, click on the down arrow located on the right hand side of the setting list (located at the bottom of the Filter Dialog Box).
2. With the left mouse button, single click on the filter or effect setting preset that you desire to delete.
3. With the left mouse button, click on the "Delete" button. A question box will appear. If you still desire to delete the

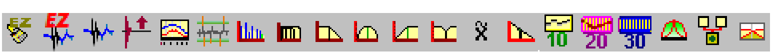
particular filter preset, click on "Yes" and it will be deleted. If you do not, click on "No" and the preset will be retained.

Importing a preset (or a set of presets)

Download the presets of interest (perhaps onto your desktop for convenience). Presets Archives are simply zip files containing one or more presets. The presets themselves have the extension of .pst, but they are placed in a .zip wrapper.
Once the preset archive is open, on the left side of the dialog box you will see the presets contained in the archive, select the presets you want to import and hit copy or copy all if you want all of them. Copy them to the appropriate filter or effect. Basically, the importing of a preset is just the opposite process as exporting a preset or a set of presets.

Exporting a preset (or a set of presets)

Exporting a preset of a set of them just uses the inverse process of importing them. Set up a directory in which to place a file folder to hold them someplace (often on the desktop for convenience).
The presets themselves have the extension of .pst, but they are placed in a .zip wrapper by the software (manage presets feature found under the edit menu).



The Filter Toolbar

Batch File Editor

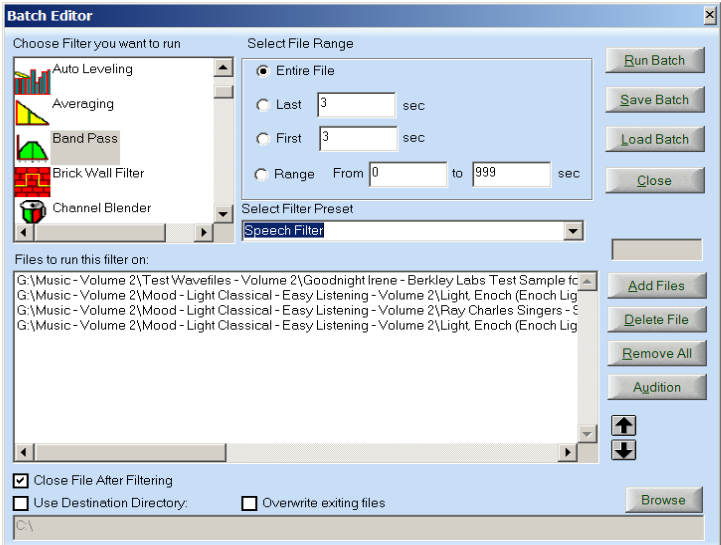
A Batch File editor is provided, so that you can assemble a group of audio files with similar problems and apply filters and their associated parameters to this entire list of files. This will allow you to create your setups and then go out and mow the lawn or perform other chores while your computer and DCArt10/DC Forensics10 are performing their chores unattended by you. The Batch File Editor functionality can be greatly enhanced via the use of the Multifilter which is one of its elements. That way, multiple filters and/or effects can be applied to a group of files in one pass. The number of files that the batch processor is capable of handling results from a total path length of 128K characters as the sum total for all file paths that you want to batch process. The maximum number of characters allowed per file name and path is 2048. The system will remember the last path that you used.

The following items are supported by the Batch File Editor (depending on the version of the software that you own):

- 10 Band Graphic EQ
- 20 Band Graphic EQ
- 30 Band Graphic EQ
- Adaptive Filter
- Auto Leveling System
- Auto Voice Filter
- Averaging Filter
- Band Pass Filter
- Big Click Filter
- Brick Wall Filter(s)
- Cell Phone Noise Filter
- Change Sample Rate or Resolution
- Change Speed
- Channel Blender
- Concatenate Files
- Continuous Noise Filter
- Convert AIFF Files to Wave Files
- Convert mp3 Files to Wave Files
- Convert Wave Files to AIFF Files
- Convert Wave Files to mp3 Files
- De-Clipper
- Big Click Filter
- Dynamic Spectrum Subtraction Filter (DSS)
- Dyna - Bass Processor (Dynamic Bass Processor)
- Dynamic Noise Filter
- Echo Effect
- Expert Impulse Noise Filter
- EZ Clean Filter

- EZ Enhancer
- EZ Impulse Filter
- EZ Forensics Filter
- File Conversion Feature
- Filter Sweeper Filter
- Harmonic Reject Filter
- High Pass Filter
- Low Pass Filter
- Median Filter
- Multi-Filter
- Narrow Crackle Filter
- Normalize
- Notch Filter
- Overtone Synthesizer
- Paragraphic EQ
- Phono Preamp (VPP)
- Polynomial Filter
- Punch and Crunch Effect
- Reverb Effect
- Spectral Filter
- Stretch and Squish Effect
- Sub-harmonic Synthesizer
- Virtual Valve Amplifier (VVA)
- Voice Garbler
- Wind Noise Filter

Note: The DC Forensics10 version supports all of the listed items.



Process Multiple Files Simultaneously!

The Batch File editor includes the following controls:

1. A "Filter Menu" which contains all of the DCArt10/DC Forensics10 filters.
2. A "File Time Range Selector" which allows you to select the time interval of the .wav files on which you desire to apply the filter / filters or effects. You can select between:
   A. Entire File
   B. Last XX Seconds
   C. First YY Seconds
   D. Range from XX to YY Seconds
3. The .wav file listing which you will create
4. A "Select Filter Preset" box
5. "Run Batch" button
6. "Save Batch" button (for saving your batch settings)
7. "Load Batch" button (for recalling a batch)
8. "Close" button
9. "Add Files" button to bring up the file manager
10. "Delete Files" button

Auto Leveling

Match Volumes of Numerous Wave Files with Auto Leveling

The Auto Leveling routine is contained only in the Batch Editor which is found under the Filter Menu. It allows you to make .wav files sound very similar in loudness across a number of files. It performs this function by calculating a combination of the RMS and Peak values of each of the files and normalizing them while accounting for both parameters and assuring that none of the .wav files overload. This functionality is quite different compared to the Gain Normalize feature in which a single file is normalized in gain with respect to its own peak value. The Auto Leveling routine has no specific controls unto itself. It uses the generic controls associated with the Batch Editor. To operate the Auto Leveling feature, highlight the Auto Leveling Icon. Then merely use the “Add Files” feature in the Batch Editor to list the files of interest by placing them in the “Files to Run this Filter On” listing box. Lastly, click on the “Run Batch” button. The resultant processed .wav files will be found to be very similar in overall loudness from one file to the next.

Concatenate Files

Concatenate Files Feature Available in the Batch Processor

Another feature, which is only available in the Batch File Editor, is the Concatenate File feature. It takes multiple files and connects them together (the tail of the first file to the head of the next, etc) and defines the resultant file with a unique name. This feature uses the standard controls associated with the Batch File Editor having no ancillary controls of its own.

Batch File Editor (Tutorial)

1. Under the "Filter" menu, click on the "Batch File Editor".
2. Choose a filter that you wish to run a batch of .wav files on by using the left mouse button. *
3. Select the desired filter preset for your batch processing. If the desired settings have not already been saved under a preset

name, you can double click on the filter icon and then set the parameters you desire for the batch processing run.

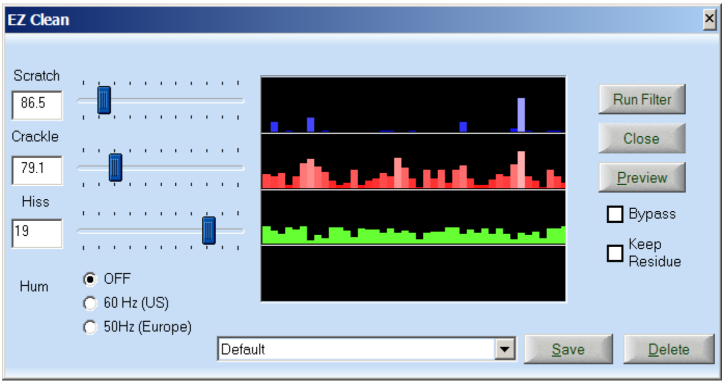
4. Click on "Add Files".
5. Select the desired .wav files from the file manager. The file listing will appear in the .wav file-listing box.
6. To delete a particular .wav file from your batch-mode listing, click on it, which will highlight it, and then click on "Delete Files".
7. When your list is complete, click on the "Run Batch" button.
8. The Batch listing can be saved by clicking on the "Save Batch" button.
9. To recall a particular batch, click on the "Load Batch" button and select the desired batch using the file manager.
10. If you have a lot of files to process and you are using an old operating system (like Windows 98 SE or Windows ME), you may want to check the box labeled “Close Files After Filtering”. By default, the files are left open in the editor window after the batch file is run. This can cause Windows to run out of resources when many files (>20) are processed. Checking this box will close each file after it has been processed which eliminates that resource limit.

Note: If multiple filters need to be run, you can use the Multi-Filter to create your desired filter sequence for Batch Processing files with more than one filter at a time.

EZ Clean Filter

EZ Clean One Step Scratch, Crackle, Hiss and Hum Removal

Sometimes it is very desirable to be able to clean up a piece of audio quickly without any hassles. The EZ Clean Filter® is designed to do just that by using adaptive filtering techniques. It removes scratches, clicks, crackle, static, hiss and hum with one very simple user interface. It includes four time domain visual Noisegraphs® to help tune it up, coupled with only three slider controls and a few checkboxes making it extremely easy to use. After initial setup, they system will adapt to varying noise environments. The tradeoff associated with the EZ Clean Filter® is that the clean-up process may not be as finely optimized as can be accomplished using the stand alone filters, each having a full array of controls over numerous variables. But, for a lot of routine audio clean-up work, you will find that the EZ Clean Filter® to be very easy to use and adequately effective.



The EZ Clean Operating Table

Scratch Control

The top control on the EZ Clean Filter® is used for attenuating large clicks, pops, and scratches in the audio material. The normalized range for this control is 0 to 100. A good starting point for this control is 50. To "tweak" (or fine tune) this control (or any of the EZ Clean Filter® controls), use the left and/or right arrow keys of your keyboard. The top graph on the EZ Clean Filter® Noisegraph® indicates the number of events being repaired per unit time vs. time. Taller bar graphs indicate that more aggressive filtering is taking place.

*Because the EZ Clean Filter uses adaptive elements within its structure, it will take a short amount of time to initialize itself when previewing or running.

Crackle Control

This control sets the threshold for the detection and interpolation of small clicks, crackle, static and impulses. Like the Scratch control, low numbers represent its least aggressive behavior, while large numbers provide a more aggressive response. The normalized range for this control is 0 to 100. A good starting point for this control is 50. The

Noisegraph® graph for this control is immediately adjacent to its control. Its Noisegraph® also indicated the number of events being repaired per unit time vs. time. Taller bar graphs on the Noisegraph® indicate more aggressive filtering.

Hiss Control

The hiss control attenuates exactly what the name implies as well some continuous noise content over the entire audio spectrum. This system is relatively automatic in that it determines its noise fingerprint “on the fly” and is dynamic. The Hiss control setting will be very much dependent on the audio presented to it, so there is no universal position from which to start. Very noisy material may require relatively low value settings to be effective while relatively quiet material may need a much more aggressive setting. Its optimum setting will also depend on the bandwidth of the source material that it’s presented with. Like the other EZ Clean Filter® controls, its range runs from 0 to 100. Its Noisegraph® graph is adjacent to the Hiss control indicating the aggressiveness of the filter on the material vs. time. If the Hiss Noisegraph® starts to “clip” (hitting full scale), this is an indication that the filter is set too aggressively and will produce “artifacts” in the resultant sound. There will be some time delay (latency) after the Hiss control is changed before the system properly re-initializes to a new starting point noise fingerprint.

Hum Selector

The bottom portion of the Noisegraph® indicates the presence of Hum. This Noisegraph® is only active when the filter has been activated. Red indicates the presence of Hum in the 50 Hz ranges +/- 2 Hz and Orange indicates 60 Hz +/- 2 Hz depending on which mode is selected. Taller bar graphs indicate louder Hum levels, which are plotted vs. time on the Noisegraph®. You must determine the appropriate checkbox to attenuate the Hum, selecting between either 50 or 60 Hz depending on the indication on the Noisegraph®. (In general, recordings made in Europe will contain 50 Hz Hum while recordings made in North America will contain 60 Hz Hum.) If clicking on one of the two frequencies causes the Noisegraph® to produce a steady stream of Noisegraph® signals, then there is probably Hum present at that frequency. The Hum Filter will remove the fundamental component of the selected value and two harmonics thereof. So if 50 Hz is selected, 50, 100 and 150 Hz components will be attenuated. Similarly if 60 Hz is selected, the system will attenuate the 60, 120 and 180 Hz Hum components.

EZ Clean Ancillary Controls

The EZ Clean Filter® also includes the following ancillary controls:

- Run Filter: Clicking on this button will run the filter on the selected file.
- Close: Clicking on this will shut down the EZ Clean Filter®.
- Preview: This allows you to hear the effect that the EZ Clean Filter® settings has on the sound as applied to your wavefile. You can adjust the controls while you are previewing in order to achieve the optimum results later when you decide to "Run" the filter.
- Bypass: Clicking on this bypasses the EZ Clean Filter®, allowing you to hear the file both before and after (A to B audio comparison) filtering.
- Keep Residue: This allows you to hear the noise components being removed by the filter.
- Presets: A number of descriptive factory presets are provided which you can use as a getting started aid when using the EZ Clean Filter®. You can also create and save your own presets.

***Note:** Since the EZ Clean filter is adaptive in nature and can also be used in real- time, it will take a few seconds for it to measure the signal to noise ratio at the beginning portion of a file. This can produce a slight latency of filter action depending on the source material. If this is an issue, consider copying a few initial seconds of the file onto the clipboard and then pasting that to the beginning of the file to be processed. This will give the filter something to measure for use as its initialization process. That extra beginning of the file can be removed later with the edit/cut command.

Multi-Filter

DCArt10/DC Forensics10 provides the capability of cascading up to 20 filters or effects together and running them as if they were a single filter. It also provides for a “Live” (real time feed-through) mode of operation. The elements of the string of cascaded filters can all be unique, or repetitions of the same filter/effect or a combination thereof. The multi-filter also allows you can even mix Diamond Cut filter and/or effects with VST plugins. This allows you the flexibility to construct your own favorite sequence of filters, along with their presets, and be able to save the cascade along with all of the filter parameters under one user defined single preset name.

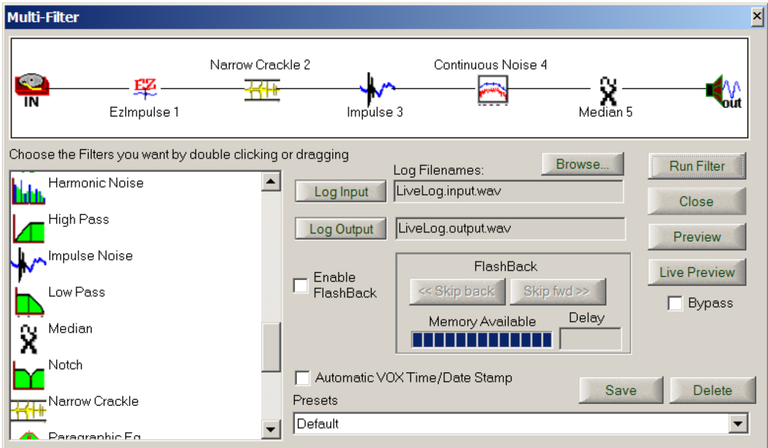
The filter/effect combinations thereby created can by previewed or run like any other single filter. In Live Preview mode (real time feed-through) you can also run all of the filters in the chain through a full duplex sound card, using the computer as a digital signal processor. Live Preview mode is particularly useful in surveillance and live broadcast work. For details on this mode of operation, please refer to the section on “Live Preview” mode below.

The best way to envision the Multi-Filter is to merely view it as another DCArt10/DC Forensics10 filter that can be customized and sequenced. To create a customized sequence (cascade) of filters merely click on the Multi-Filter icon or select Filter/Multi-Filter, and the Multi-Filter dialog box will pop up. On the left, you will see the source input (In) to the system. On the right of the screen, you will see the output (Out). The input is that .wav file which is present in the source window, (or in the case of Live mode, it can be the sound card input signal) as is the case with any other filter. The output of the filter can be previewed, bypassed, or run just like any other filter. It can also be run in “Live Preview” mode as well (see below).

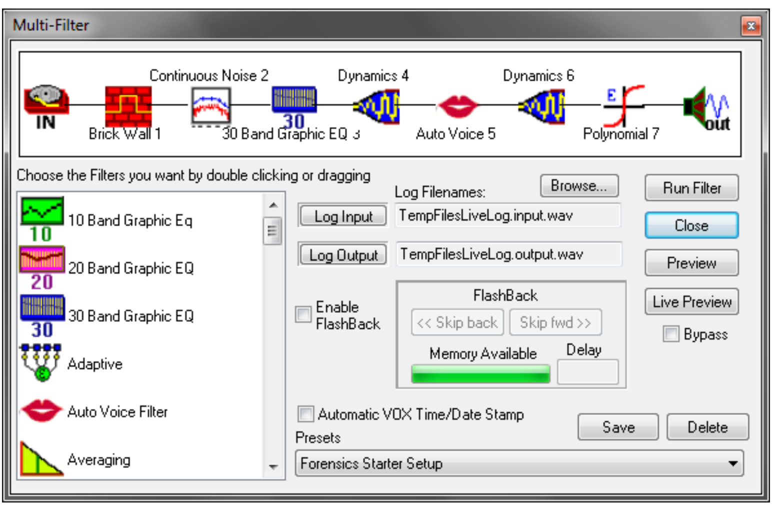
To create your filter sequence, merely drag and drop the desired filters from the filter selection grouping into the signal pathway in your prescribed order. If you want to change the order of the filters in the cascaded chain, merely left click on the filter in the pathway, and drag it to a new location along the signal path. To get rid of a filter, simply drag it away from the signal pathway with the left mouse button, releasing the button when the filter is away from the signal stream. To view the parameters of any given filter or effect in the Multi-Filter, double click on the filter icon in the cascaded chain and the particular filter dialog box will pop-up. Adjust the parameters to the values that you desire, and they shall be saved as part of the sequence of filters that you have just constructed. As with all of the filters, the parameters may be previewed while adjusting the parameters. Be aware that when long chains of filters have been created, there will be a delay in time before the controls react to your changes. If you desire to delete a filter, drag the filter out of the chain and release the left mouse button.

After you have developed a Multi-Filter chain that you find to be particularly effective for a certain type of task, you can save its settings under a descriptive name just like any other DCArt10/DC Forensics10 filter for ease of recall later.

It is important to note that improved audio performance will be achieved using the Multi-Filter as opposed to a sequence of .wav file processing, performed one step at a time. The reason is that there will not be the quantization error buildup since the Multi-Filter will maintain the highest possible resolution for the signal throughout the process from the first filter to the last one in the sequence. Sequential one step at a time filtering must always convert back to a .wav file as the intermediate step between filter operations, producing a buildup of quantization errors. This error build-up is avoided with the Multi-Filter.



Run Multiple Filters Simultaneously!



Forensics Starter Setup preset in the Multifilter

The "Forensics Starter Setup" preset found in the Multifilter is a very good place to start a Forensics enhancement project. It includes a plethora of useful filters all having good starting point settings. From those starting point values, you can adjust each filter element for optimal results for a given enhancement project.

Live Preview

Live (real time feed-through) mode allows you to bypass the hard disk recording process, making your computer into a real-time feed-through digital signal processor. Signals are fed into your sound card, processed through the filter or filters chosen, and then fed back out of the output of the sound card for use "live." Live mode is useful for professional applications such as broadcasting, or surveillance work where a signal needs to be processed in real-time without the intervening process of hard drive recording. It is also very handy for use in a Ham Radio, Short Wave listening (DXing) or even as a permanent part of a home audio system as a real time signal processor. This feature is particularly useful for real time audio restoration as may be required by radio stations desiring to play old vinyl or 78s directly on the air. It is also useful for cleaning up live news audio feeds or noisy telephone audio on talk radio shows.

This feature requires a full-duplex sound card and a fast computer. We recommend the fastest computer that you can afford, since the faster that your computer can perform the mathematical functions, the more filters that you can cascade in "Live" mode. System "stuttering" is an indication that you have exceeded your computer's ability to keep up with the data processing in real-time. Also, latency (the time delay for processing) is reduced as the speed of your computer is increased. For professional applications, we recommend a Pentium 4 or higher for "Live Preview" mode when used in conjunction with the Multi-Filter feature. However, slower computers will run with increased latency and a reduced maximum number of allowable cascaded filters.

Live mode is activated by clicking the “Live Preview” button in the Multi-Filter window. To adjust the input parameters for your sound card, double click on the input icon (the turntable) and a dialog box will pop-up. The following input parameters are adjustable:

- Mono or Stereo
- Sampling Rate (up to 192 kHz)
- Resolution (8 bit to 24 bit)

- Input level indicators are also provided

Adjust the parameters appropriately. Keep in mind that stereo signals will consume about twice the computer resources compared to monophonic signals. So, if you do not need stereo, do not operate the Live mode in stereo, else you will be adding extra latency to the signal and reduce the number of filters that can be run without “stuttering”.

Drag and drop the filters of interest into the signal path. Adjust their parameters by double clicking on each filter, producing its dialog box.

To run the "Live" mode, merely click on the "Live Preview" button, and the signal will be processed through the filters/effects in the signal path. To adjust the output level, double click on the output device (the loudspeaker icon) and a dialog box will appear to facilitate this function. You can also create .wav files in real time while using the Live mode. This feature is also contained in the output device dialog box.

To delete a filter from the line-up, simply drag it out of the pathway or use the right-mouse function called “Delete Filter”. To clear the entire signal pathway, use the right mouse function called “Delete All Filters”.

Note: To minimize Live Preview Mode latency, use the smallest number of preview buffers and the highest possible sampling rate that your system will support. Minimum latency will generally occur at 96 kHz or 192 kHz sampling rates. (Conceptually, this is probably counter-intuitive.)

Important Note: When using "Live" mode, it is not advisable to connect the soundcard I/O in an effects loop on a mixing console. Doing so can produce echo effects due to the latency of the computational process. Instead, connect the soundcard directly in the signal pathway which you desire to process.

Live Log to Disk Mode (DC Forensics Versions Only)

You can now "Log to Disk" (to your hard drive) both the input and/or output content of the signal path being used in Live mode via the Multi-

Filter. These logging features coupled with Live Preview can also be used as recording monitors when the speaker icon is clicked and used. You are able to listen and/or log the signal as you record. These logging features can be useful when it is necessary to create an archival recording of a surveillance or of a broadcast session. It is possible to log both the unprocessed and processed data simultaneously to your hard drive for later retrieval. To perform the Log to Disc functions, merely click on the appropriate Log button(s) that you want to set up. To log the raw source data, click on the “Log Input” button and it will become highlighted. To log the processed data, click on the “Log Output” button and it will become highlighted. You can log the Input and Output signals simultaneously if you desire by clicking and therefore activating both logging functions. Whenever you click on the "Live" button, whatever signals are being presented to the Multi-Filter will be logged to the disk under the Filename(s) indicated next to the “Log Input” and “Log Output” toggle buttons.
Both input and output logging can be stopped and started during a live session, and neither has to be enabled when live mode is first started. Log to disk data will be appended to the files that are shown in the log filename boxes. If the Multi-Filter is closed and re-opened, a new temp filename will be selected in the log filename boxes. The Automatic VOX and Time/Date Stamp feature applies to both the input and output logging function when checked.

Live Mode VOX Time/Date Stamp (Forensics Versions Only)

By checking the box labeled Automatic VOX Time/date Stamp, DC Forensics10 will add a marker each time a recording starts in VOX mode. This is useful for remote location or surveillance recording. By enabling this function, the system will record only when audio is actually present to be recorded and each recording event will be marked with the exact time and date of the individual recording. This function is useful only in Live mode.

Flashback (DC Forensics Versions Only)

Flashback lets you quickly Review and Clean an audio stream Live without interrupting your recording

Flashback is primarily designed to be used in real time intelligence gathering and Forensics surveillance situations. It is a feature that allows you to go back and listen to something that you may have heard during real time operation of the software that may be of immediate interest and needs rapid clarification, especially in covert military situations. Essentially, it is analogous to instant playback, instant review or instant replay. This feature is available when you are operating the Multi-Filter in Live mode and is only available in the Forensics version of the software. It uses RAM to store up to 300 seconds (5 minutes) of audio data in stereophonic 16-bit, 44.1 kHz format on a FIFO basis. Therefore, it will require about 53 Mbytes of available RAM in order to properly function. It can be operated with or without the log to disc function(s) activated. Faster sampling rates used in the Multi-Filter recorder will de-rate the available Flashback time availability. Flashback always operates on the output signal of the Multi-Filter. Therefore, signal processed data as set up in the Multi-Filter is fed into the Flashback buffer memory.

The controls for the Flashback feature are as follows:

- Enable Flashback: Check this box to Enable the Flashback Function
- Skip Backward: Moves the play pointer backwards in 5-second increments per click of the mouse.
- Skip Forward: Moves the play pointer forwards in 5-second increments per click of the mouse. (Of course, this function only is operational after you have Skipped Backward first, else you would have a time machine on your hands)
- Delay: (Seconds): This displays the amount of time shift backwards from real time that you are monitoring via the Flashback memory.
- Space Available: This indicates the relative amount of RAM which is available for the Flashback function which depends on the amount of "Skip Backwards" delay you have set.

Available Flashback Time as a Function of Sampling Rate				
Sampling Rate	44.1 kHz	48 kHz	96 kHz	192 kHz
Available Flashback Time	5 Minutes	4.5 Minutes	2.3 Minutes	1.2 Minutes

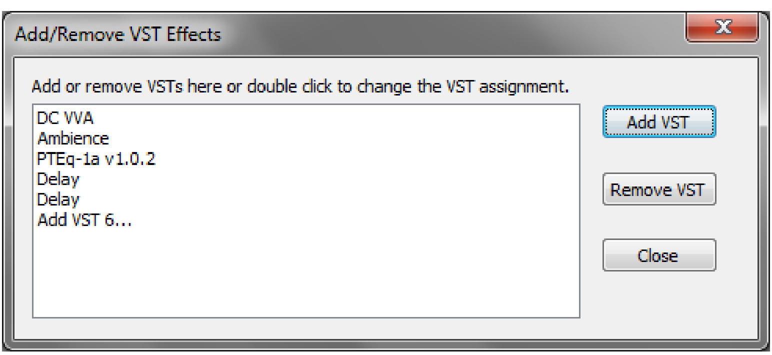
<u>Note: You can delete filters (or all of the filters) from the signal path via commands found on the right mouse button.</u>

<u>VST Plug-in Filter Support</u>

DCArt10/DC Forensics10 includes the ability to accept (or host) up to six standard VST plug-ins. There are thousands of compatible plug-ins available which offer many types of audio processing routines. Plug-ins range in price from free to super expensive and are available over the Internet. (DirectX plugins are not supported.)

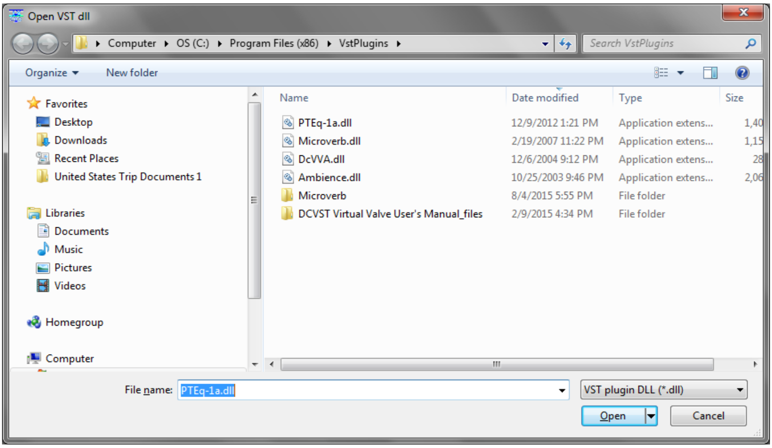
The VST Toolbar

Before you attempt to apply a VST Plug-in tool, you must first install it onto your computer and then add it to the list of available VST .dll routines inside the Diamond Cut Software Program.
To add an installed VST into your Diamond Cut Software, click on the add/remove function next to the VST slots under the Effects menu. A dialog box will appear as shown here:



Add VST Plugin Dialog Box

Next, click on one of the “Add VST x” slots and then click on “Add VST” and steer to the location of your installed VST plugin .dll using the dialog box, as shown here:

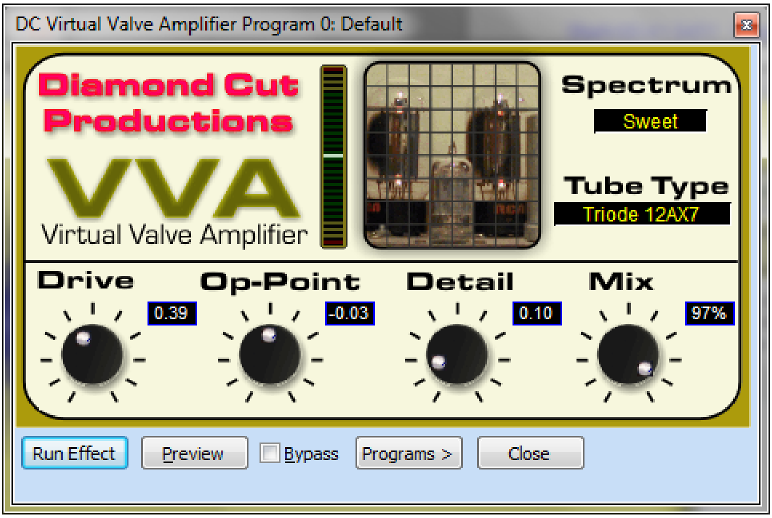


VST .dll location dialog box

That operation will assign that particular VST plugin to the selected slot. You can do this for up to a total of six VST plugins. The installed VST .dll plugins are generally found in one of the program folders under the “C” drive.

You can use the Delete button to remove VST Plugin tools from the “Availability” list.

To use a VST plugin, just go to the Effects menu and then click on one of the populated VST Plug-ins slots. A screen similar to the one shown here will then appear. Like all of the Diamond Cut Software tools, you have the standard “Run Effect”, “Preview”, and “Close” buttons available.



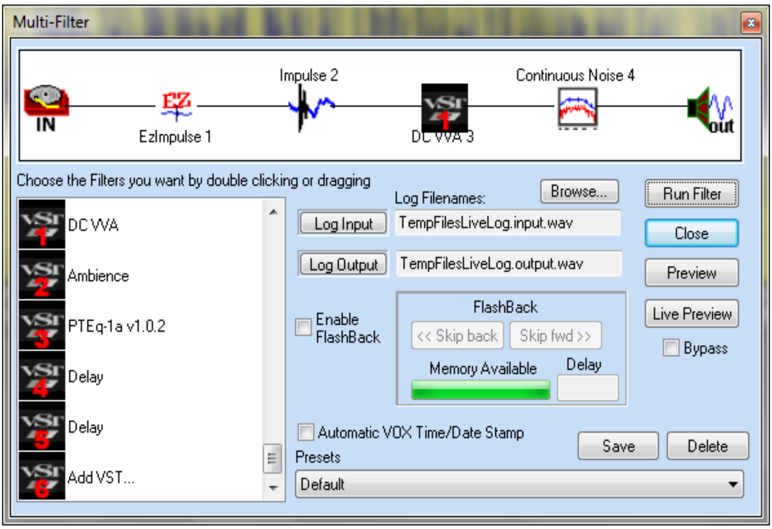
A VST Plugin launched in Diamond Cut Forensics10

A VST Effect in Context

Once you have added tools you can enable them for actual use by double clicking on the desired plug-ins. Clicking on a tool will bring up that specific plug-in's graphic controls. It can then be dragged and placed anywhere on your workspace. Chaining of VST Plug-ins is possible using the Diamond Cut Multifilter. Since VST Plug-in tools are written by programmers other than the authors of DC Forensics10, we can't predict what you will ultimately see on the multifilter screen.

Usage from this point forward is like any other built in tool with DC Forensics10; just preview and adjust or run the VST filter or effect to suit your needs.

Using VST Plug-ins in the Multi-Filter



VST Plug-ins Integrate into the Multi-Filter

As can be seen above, the new VST icon(s) are available in the Multi-Filter so that these types of tools can be freely used with all the native DC Forensics10 tools. Please see the section in the users' manual describing the Multi-Filter topic for specifics on the use of these tools when cascaded with others.

Caution:
Being able to use audio tools designed by dozens of different companies is a powerful feature in your Diamond Cut software. However, it must be remembered that these tools are all independently designed and created by their respective manufacturers and are not created or supported by Diamond Cut Productions, Inc. Correctly written VST Plug-ins should work fine in the program, but it is possible to download a poorly written or buggy plug-in. If you find that a specific plug-in is behaving strangely, we recommend contacting the VST plugin manufacturer or simply deleting or uninstalling it from your hard drive.

Note: The system supports only 32 bit VST plugin modules

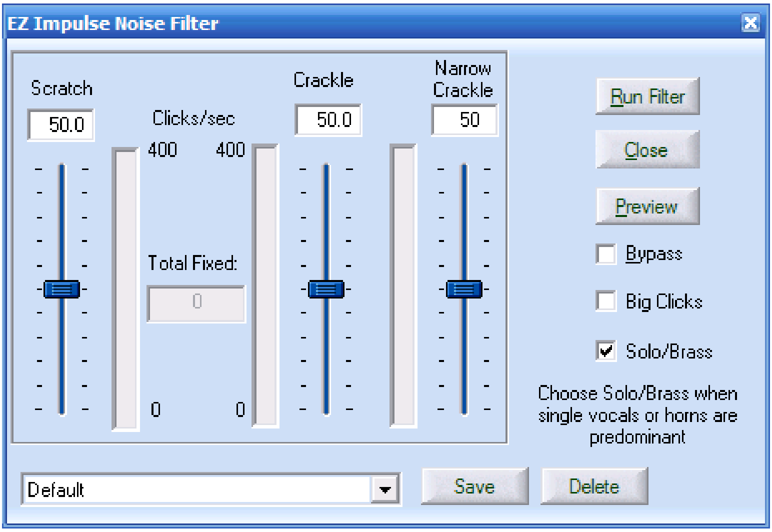
Impulse Noise Filters (Attenuates Clicks, Ticks, Crackle, Snaps, Pops, Thuds, Static & Buzz)

Four Impulse Noise filters are provided in the DCArt10/DC Forensics10 suite. They are all found under the Filter sub-group called “Impulse”. The EZ Impulse Noise Filter provides you with a high degree of automation (adaptive techniques) to simplify the impulse noise reduction process, with the tradeoff of reduced user controllability. This is the “goto” filter of choice for most impulse noise reduction work. The Expert Impulse Noise filter provides a very high degree of adjustability for those who want to customize the system to their own personal preferences. Some people like create a custom impulse filter via multiple Expert Impulse Noise Filters in the multifilter with each one set to different modes and parameters. However, this impulse filter is difficult to use, especially for the casual user. The Narrow Crackle filter specializes in handling impulse noise having narrow pulse width. The Big Click Filter specializes in very large clicks and thuds only. All of these filters are non-linear algorithms used to eliminate pops, ticks, clicks, and crackles from audio recordings. It is also useful for the elimination of "static" interference from AM, FM, or Short Wave radio broadcasts. It can also be quite effective for attenuating high frequency line related “buzz” from certain recordings. All of these types of noise signals generally look like impulses (although sometimes referred to as "spikes"), and therefore the name “Impulse Noise Filter”. The algorithm essentially monitors for fast events, and when their value exceeds a threshold value, the algorithm blanks out the portion of the file wherein the fast event occurred, and re-inserts a waveform that is an approximation of the signal that likely would have occurred during the event. The phase of the inserted signal is aligned to match the point in time in the file where it is inserted so that there is no phase discontinuity, and therefore almost no artifact will be injected into the .wav file.

EZ Impulse Noise Filter

The EZ Impulse Noise Filter™ combines the noise rejection characteristics of the more user controllable performance of the “Expert-Impulse” Filter with some automation and the resultant ease of use. The purpose of this filter is the attenuation of record clicks, ticks, crackle, and pops. It can also be used to reduce radio static interference and can be used to remove some high frequency “buzz” related to the power line frequency. It has only three controls that need to be set by the user. One control determines the aggressiveness of the EZ-Impulse Filter’s response to large scratches or impulses. A second control affects its response to “crackle” which is another term for smaller and densely populated impulses. The “Narrow Crackle” control affects its response to smaller impulsive events. Three bar graphs provide the user with visual feedback as to the effect that each of the controls has on the signal. A tally of “Total Clicks Fixed” is also provided for use as a relative reference and an operational aid. Before using the EZ Impulse Filter, the file should be Gain Normalized to -3 dB first for optimum results. Use the Normalized Gain Scaling feature found under the CD Prep menu to accomplish this task. Set it for -3 dB.

The following is a listing of the EZ-Impulse Filters controls:



EZ Impulse- Powerful and Easy To Use!

- **Scratch**

This control sets the threshold for the detection and interpolation of large clicks, pops and scratches. Low numbers are less aggressive, while large numbers are more aggressive. The normalized range for this control is 0 to 100.00

- **Crackle**

This control sets the threshold for the detection and interpolation of small clicks, crackle, impulses and high frequency “buzz”. Like the Scratch control, low numbers are least aggressive, while large numbers provide a more aggressive response. The normalized range for this control is also 0 to 100.00

- **Narrow Crackle**

This control sets the sensitivity of the system to narrow pulse width crackle. Like the other controls, increased values represent increased sensitivity to this type of noise. The normalized range for this control is 0 to 100.00 with a good place to start being around 50.

- **Big Clicks Checkbox**

This feature enables a “Big Click Filter” having a fixed ratio setting of 1.4. For details pertaining to the functionality of a “Big Click Filter”, please refer to the section of this User’s Manual having that name.

- **Bar Graphs**

All three filter controls have their own “Clicks per Second” bar graph having a range of display of 0 up to 400. These three graphs are useful for setting their respective controls. Indications that are too low may mean that the system is not sensitive enough to pick up the impulses (or there are none or not many), while indications that are too high may indicate that the Wave file is being damaged by overreaction of the filter(s). Using these graphs, in conjunction with the “Preview” mode makes the setting of the EZ Impulse Filter very simple and effective. The slider control green zones are good places to start with. Most of the time, you will find the best result to be within that zone; rarely, the control(s) will have to be set higher or lower than those zones.

- **Total Fixed**

The Total Fixed accumulator counts the total number of impulsive events repaired by the system. It responds to the Scratches and Crackles, but not to the Narrow Crackle function. It is interesting to compare these data from one recording to another. It is somewhat of a litmus test for the grade of the recording that you are working with.

- **Bypass**

This control allows you to hear the unprocessed signal for comparison purposes. It can be used “on-the-fly” during Preview and slider adjustments. Do not forget to uncheck “bypass” before running the filter after you have satisfied yourself with the slider control adjustments.

- **Solo/Brass**

Choose this setting when brass instrument solos or up-front vocal solos are causing distortion when using the EZ Impulse filter. This feature will suppress distortion produced by those instruments or strong lead vocals. It should not be used on material that does not contain up-front brass or up front vocals as the filters performance will be compromised under those circumstances.

EZ Impulse Noise Procedure (Tutorial)

After your file has been gain normalized to -3 dB (found under the CD Prep Menu), the EZ Impulse Filter operating procedure is as simple as one, two, three.

1. Place both the "Scratch" and "Crackle" and "Narrow Crackle" controls to their lowest positions.
2. Using "Preview" mode, adjust the "Scratch" Control upwards for the best balance between effective scratch removal and minimized distortion. Generally, that will occur at a value around 55.
3. Next, advance the "Crackle" control upwards for the best balance between effective crackle removal and minimized distortion. Generally, that will occur at a value around 50.
4. Lastly, if there are any residual tiny crackles left behind, raise the Narrow Crackle control until they are attenuated by the system. Generally, that will occur at some value between 25 and 85.

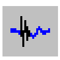
Also, an array of presets is provided with the EZ Impulse Filter that may produce reasonable results without the need for adjusting the filters controls. However, the optimum results will usually be obtained by following the above-mentioned four-step procedure since impulse noise removal is source material dependent.

Note 1: Use the Impulse Noise filter(s) first in your Audio Restoration Sequence after file gain normalization to -3 dB. Never pre-filter the signal in any other way prior to this restoration step. The filter needs the maximum bandwidth signal to help it to perform its function.

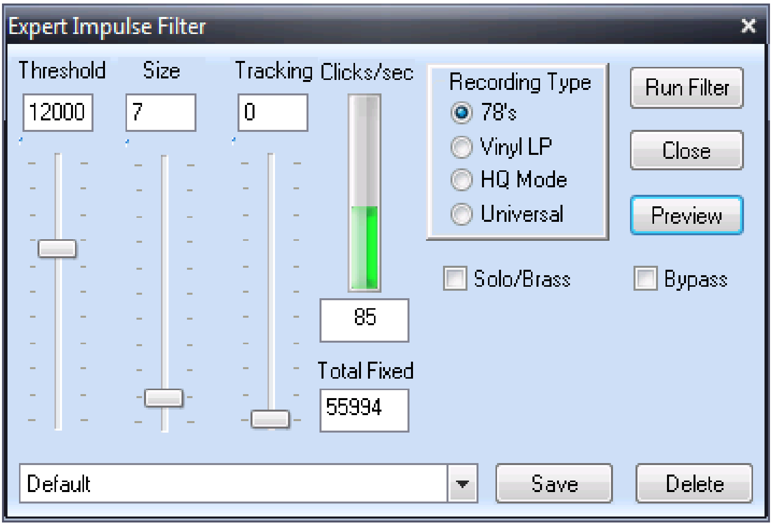
Note2: Improved performance of this filter can be achieved by recording at a 48 kHz or 96 kHz sampling rate. The filter takes advantage of the ultrasonic content of the impulse signal provided by a higher sample rate in order to improve detection and discrimination of impulsive noise. You can down-convert to 44.1 kHz immediately after the de-clicking process has been completed if you so choose, since the remaining filters will not specifically benefit from a higher sampling rate.

Note 3: Up-Sampling a file to 96 kHz that was originally sampled at 44.1 KHz or lower will not provide the same benefits outlined in Note 2. The EZ Impulse filter primarily benefits from a 48 kHz or a 96 kHz sampled file that was originally digitized (transferred) at one of these higher rates from its analog source during the Analog to Digital conversion process.

Expert Impulse Noise



As your expertise with the product grows, so too will your demands from the various filters. The Expert Impulse Noise Filter approach allows you to control all of the parameters associated with a basic impulse filter and therefore obtain the most flexible results. Since it contains no adaptive elements, it does require a larger degree of experience to use compared to the EZ Impulse Filter. Multiple Expert Impulse Noise filters placed in the multifilter will often provide the best results, with each instance set up with different modes and/or parameters. Generally speaking, the Expert Impulse Noise Filter is designed to remove impulsive noises from a signal such as clicks, ticks, scratches, pops and static. It can also be used to attenuate harmonically rich “buzz” from a recording. The Expert Impulse Noise filter is primarily intended to be used in extremely unusual phonographic situations or (primarily) for Forensics applications in order to filter out buzz or static from a communications link. You will find that the EZ Impulse filter solves almost all impulsive noise problems that you will encounter, but for the rare few situations that occasionally pop up. So, try the EZ Impulse filter first before experimenting around with the Expert Impulse Filter. As with all of the Diamond Cut Impulse Filters, the minimum sample rate used should be 44.1 kHz. Lower sample rates will produce inferior results. The following is a summary of the Expert Impulse Noise Filter control parameter functionality and their adjustment ranges:



Expert Impulse Noise Filter

- **Threshold**

This is the Voltage derivative signal level above which the program decides that an impulse noise event has occurred. It has a range of adjustment from 1 to 18,000 (in DAC counts normalized to a 16 bit system). A good starting point for the threshold value is 1/3rd of the full-scale value of the envelope of your .wav file. For example, if the .wav file is almost full scale in amplitude (+/- 32,000 counts) set the threshold at around 10,000.

For 78 RPM records, (with the Tracking control set to 0), start with a Threshold value of 1000, and adjust up or down depending on the results obtained. Lower values of Threshold will produce a higher degree of de-clicking. If it is set too low, however, you will produce distortion on your recording. If you do not want to start at 1000, use the 1/3rd of full-scale rule for your initial setting.

Note: Threshold should be set to its lowest value (slider down) for Vinyl LP/45 RPM applications. Adjustments should be made using the Tracking adjustment for these applications.

- **Size**

This is the number of samples during which the "click" or "pop" event must remain to be defined as an impulse noise event. It has a range of adjustment from 2 to 60 samples. Short "clicks" require a smaller setting compared to longer "pops." A good range of values to start with for Vinyl LP applications is in the 10 to 15 area. A good range of values to use for non-Vinyl (like 78-RPM records) is between 3 to 7 samples.

- **Tracking**

This is the value of rectified output Voltage from a High-pass filter that is used to modulate the threshold Voltage of the filter. When there is a lot of high frequency information present on the recording, like the crashing of cymbals, it is desirable to move the threshold higher in value so that the transients contained in such sounds are not mistaken to be impulse noise events. Tracking has a range of adjustment from 1 to 100 (in relative units). Tracking is most useful on "high fidelity" recordings that contain a lot of "real" high frequency information such as loud dynamic cymbal crashes, or exaggerated sibilant sounds, which may be interpreted by the Impulse Filter as impulse transients.
Most 78s de-click best with the tracking turned all the way down (to a setting of 1). For LPs, start with a setting of around 25 to 30, and adjust the value upwards if distortion is heard on high frequency passages or sibilant sounds, until the distortion disappears. However, if the tracking is set too high, adequate de-clicking may not be obtained.

Note: Tracking should be set to its lowest value (slider down) for non-Vinyl LP applications. Adjustments should be made using the Threshold adjustment in these applications. If the threshold control is set too high, the Impulse filter will not completely de-click your recording. If it is set too low, the filter will create distortion on your recording when high frequency signals are present, especially on the higher registers of the audio scale.

- **Preview Mode**

When enabled, this allows you to hear quickly the results of your chosen settings in real time. If your computer is a slower model, the system may "stutter" when this is enabled, however, the feature can still be quite useful for finding the best settings, since the "stutter" will not appear on the final product when the filter is “Run”. The slider controls can be adjusted "live" when the preview mode is enabled. Preview mode is invoked merely by clicking on the "Preview" button on the filter dialog box.

- **Vinyl LP Mode**

When this is enabled, a different type of detector algorithm is utilized which is more optimized for the wider bandwidth and narrower clicks and pops encountered with Vinyl LPs. Vinyl mode works most effectively on .wav files that are sampled at 44.1 kHz up to 96 kHz. When this mode is turned off, the detector is more optimized for slower and wider impulse noise. Unlike most of the other controls, you cannot switch Vinyl LP mode on or off "live" when preview mode is invoked. Use the following table for determining the correct mode for this selector based on the type of material you are working with:

Sound Source	**Vinyl LP Mode**
Vinyl LP (Stereo or Mono)	"On"
45 RPM	"On"
FM Impulse Noise	"On"
FM Stereo Impulse Noise	"On"
Acoustical 78s	"Off"
Electrical 78s	"Off"
Cylinders	"Off"
Hill and Dales	"Off"
Movie Soundtrack "Pops"	"Off"
AM or Short-Wave Static	"Off"
Forensic Audio	"Off"

Note 1: When you run the Expert Impulse Filter, a dialog box will appear which indicates the Clicks/Second and the Total Clicks Processed. The Clicks/Second statistic is relative to the timing of your .wav file, and not the time frame in which your computer is processing the data. This feature is provided to help you determine if you are "trashing" (creating distortion) your .wav file due to Thresholds or Tracking values that are set too low. When the algorithm's parameters are set too aggressively, the Clicks / Second number will become extremely high, which could be an indication of impending distortion of your .wav file (although it depends largely on the condition of your Source material).

Note 2: Multiple runs of the impulse filter(s) set un-aggressively will produce better results than running an impulse filter aggressively. Consider setting up multiple Impulse Filters strung together in the Diamond Cut Multi-Filter to facilitate this approach easily.

- **Hind Quaternion Mode (HQ Mode)**

DCArt10 and DC Forensics10 have included an enhancement to the Impulse filter called Hind Quaternion mode (a name related to the method used in the click detector portion of the Impulse filter). This mode can be used with 78s, vinyl, or any source containing impulsive type noise or clipping distortion. Its operation couples several related variables in this new detector algorithm together in a logical ratio-metric relationship allowing you to vary them with one control. The Size slider controls these new variables. The other controls on the Impulse filter still perform their previously defined functions. In HQ mode, you will find that you have a much larger degree of control of the detector algorithm, especially through the use of the Size control. Small fast rise time clicks will be detected optimally with small values of Size while larger and slower events will be best detected with larger settings. The advantage of this mode is an improvement in the ability of the detector to reject musical transients while still maintaining good click sensitivity.

- **Universal Mode**

Universal Mode provides an adaptive element to the detector portion of the Expert Impulse Filter algorithm. It can be used with any type of impulsive noise, whether the source is from records (78s, 45s, 33.3s), radio static, or tape head static discharge. Universal Mode is of particular advantage in some wireless surveillance applications where occasional static needs to be mitigated. The advantage of this mode is its ability to adapt to changing signal environments. The Tracking control is its primary means of adjustment. The Tracking control affects the systems overall centered value of sensitivity while the Threshold control sets the minimum value that the system can apply to the detector. The size control does the same thing as with the all of the other modes associated with the Expert Impulse Filter. The disadvantage of the Universal Mode detector is its diminished transient response. This will cause the system to have some difficulty in dealing with the leading edges of rapidly changing audio material, since it will take some time to re-adjust itself. This could cause some transient leading edge distortion and/or leading edge missed impulse detection.

Important Note: The various Impulse Noise filter modes (LP, 78, HQ, Universal) all take a somewhat different approach toward identifying clicks and pops. Since these types of impulses can come in an almost infinite number of sizes and types, a user may select the mode that provides good results over a very wide range of impulsive noise environments.

- **Solo/Brass Mode**

DCArt10/DC Forensics10 provides you with a discriminator routine which identifies brass instruments (and certain types of vocal solos) and excludes them as candidates for interpolation. Since brass musical instruments (like solo trumpets) produce waveforms that are quite similar in form to impulsive noise events, this option can be quite useful in preventing distortion from being interjected upon those types of instruments. This mode of operation is also useful on some "close-miked" vocals where the background instrumentation is low in sound level comparatively speaking. The tradeoff with using Solo/Brass mode of operation is reduced de-clicking capability during the actual presence of the brass instrument or vocal in question. This mode should not be used on material that does not contain strong solo brass instruments or up-front lead vocals.

Note 1: Do not mute the beginning or the ending of a .wav file before operating the impulse noise filter. Mute the extraneous noises from the beginning and the ending of your .wav file at the end of all of your audio restoration processes.

Note 2: There will be occasions wherein 78-RPM recordings will benefit from the de-clicking action of the vinyl mode impulse filter in conjunction with the tracking control. If the clicks are small and short, it is worth giving it a try.

Note 3: Multiple passes through the impulse filter, especially in vinyl mode while using the tracking control, can produce ever-improving results, especially when Solo/Brass mode is enabled.

Expert-Impulse Noise Operating Procedure (Tutorial)

1. Highlight the portion of your .wav file on which you desire to apply the Impulse Noise Filter. (You may choose to highlight the entire file or any portion thereof.) Sometimes, when confronted with extremely stubborn clicks or pops, or radio "static" it may be useful to use the Zoom-In feature first before running the Impulse Noise Filter on a "grouping".
2. Click on the "Filter Menu" with the left mouse button.
3. Click on "Expert Impulse Noise".
4. Start with the "Threshold" control at a setting of approximately 1000 for 78s.
5. Start with a "Size" setting of between 3 to 7 samples for non-Vinyl applications, and use a setting somewhere in the 10 to 15 sample range for Vinyl LP and 45 RPM record applications.
6. If you are de-clicking a Vinyl LP record, click "Vinyl LP" on with the left mouse button.

<u>**Note:**</u> This feature is also utilized for 45-RPM records. If you are de-clicking a 78-RPM record or something similar, make sure "Vinyl LP" is turned-off. (It is important to re-emphasize that Vinyl LP mode works best on .wav files that have been sampled at 44.1 kHz or higher.)

7. If you are de-clicking a Vinyl LP record, set the threshold control to its lowest value, and perform all of your adjustments with the tracking control, starting with a setting of 25 to 30. If you are de-clicking a 78-RPM record or something similar set the tracking control to its lowest value and perform all of your adjustments with the threshold control.
8. Click on "Preview”.
9. Listen to the "Previewed" version of the processing parameters that you have just set. If your computer is too slow, it will "hick-up" or "stutter." (Do not be concerned that your final sound restoration will sound like this, since it will not!) Try to listen "through" the stutter to judge what the Filter is doing. If the "stutter" is too annoying to make a judgment of the performance of the filter settings use Run filter mode on a selected portion of the .wav file directly into the "Destination" workspace. Alternately, run the filter, and then listen to the

Destination Workspace in order to make judgments regarding your settings. Iterate until you are satisfied with the results.

Note: Setting the filter too aggressively may cause excessive stuttering during Preview. Even a fast computer will stutter when this filter is finding hundreds of clicks per second. If this is happening, readjust the filter to be less aggressive.

10. When the filter is running, you will see a display of "Clicks / Second" and "Total Clicks Processed." Generally speaking, when the threshold is set too low, the program will begin to react to sound transients rather than just noise transients. If the "Clicks / Second is greater than 30, there is a good chance you are catching sound transients, and creating distortion on the output of the filter. Most records will show less than 10 clicks per second when the settings are correct (except in extreme circumstances). Keep adjusting the threshold setting until the clicks are being removed and distortion is not being produced on the filter output. (The distortion that can occur will be most prevalent on the sibilant sounds.) Keep in mind that lower value settings of the threshold control will cause the algorithm to be more sensitive to removing clicks and pops. However, if it is set too low, distortion will also be produced on the sibilant sounds.
11. If the algorithm is capturing the larger impulses but not the smaller ones, try decreasing the "Size" adjustment, and re-evaluate the results. (You may also have to decrease the threshold control.)
12. When you determine the best setting of the controls for your particular .wav file, click “Run Filter”. When the filter has completed its operation, the results will appear in the "Destination" workspace.

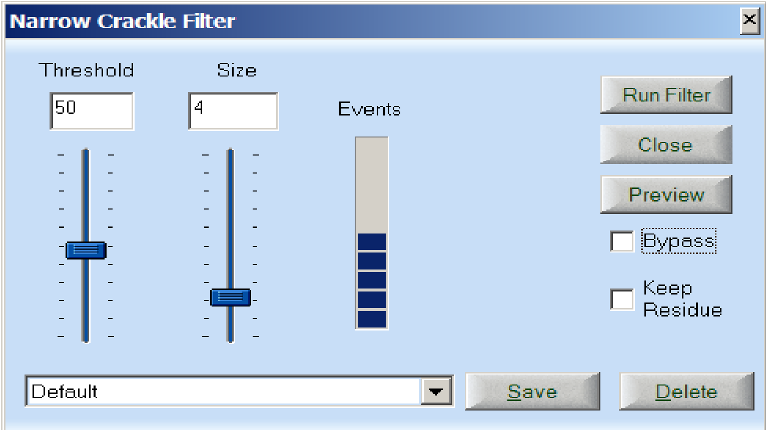
Note: Notes 1, 2 & 3 found at the end of the EZ Impulse Filter Tutorial also apply to the Expert Impulse Noise Filter.

Narrow Crackle Filter

The Diamond Cut Narrow Crackle Filter (NCF) is a special impulse filter designed to identify and interpolate narrow pulse-width crackle type noise. It is especially effective on mint condition LPs having only very small impulsive noise due to vinyl defects or static electricity discharge effects. This filter is also useful on other recording types provided that larger impulsive noises have been removed by one of the other Diamond Cut Impulse Noise filters first. Additionally, it can be used to remove some power-line frequency related high frequency "buzz" (which is high frequency & harmonically rich "hum"). The advantage of the Narrow Crackle filter is that it is the least invasive of all of the impulse noise reduction filters in the Diamond Cut suite. We have found it to be effective in reducing the residual crackle sounds on not only Vinyl LPs, but also transcription acetate recordings and 78s after all of the other impulsive noises have been removed first. It is not effective for removing impulses due to poor record handling; use the EZ Impulse filter for those situations. The Narrow Crackle Impulse Noise Filter is very easy to operate, having only two controls which are as follows:

- Threshold: This control sets the relative amplitude at which this filter detects narrow crackle impulsive noise having a range of 5 to 100. Adjust this control until downwards until the impulsive noise is reduced but not so low as to reduce the bandwidth or produce distortion on the target signal. A good place to start with on this control is a setting around 30. Settings below 10 can produce distortion and also a Low Pass Filter effect on the signal.

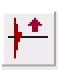
- Size: This parameter sets the pulse width at which the system is sensitive, having a range running from 1 to 15. Larger values of size represent longer pulse width values. Use the smallest value of size to do the job at hand. A good value to start with is around 3 to 5.



The Narrow Crackle Impulse Noise Filter

The Narrow Crackle Filter's "Events" bar graph provides you with a relative indication of how aggressively the filter is operating. It "modulates" vertically in proportion to the number of events that are being interpolated by the system. The "Keep Residue" feature allows you to hear what the Narrow Crackle Filter is removing from your source signal. A good way to "tweak" this filter is to highlight a short sector (around 10 seconds) of the .wav file that has the type of narrow crackle impulse noise that you desire to eliminate or attenuate. Preview that sector and adjust the two controls for the best result on that 10 second sector. After you are satisfied with your settings, it should then do a fine job on the entire file in "Run Filter" mode.

Big Click Filter



The Big Click Filter (BCF) handles exactly what the name implies which includes the elimination of very large clicks but also loud thuds. Its response intentionally ignores smaller clicks and ticks. Its response covers the long time interval range of events lasting greater than 2 mSec but less than 200 mSec. Its detector is actually sensitive to the time amplitude product of the applied signal. This parameter is user adjustable by way of the "ratio" control which has a range which can be varied between 0.75 and 2.5. 1.4 is a good starting place with higher values decreasing the filters aggressiveness and lower values increasing aggressiveness.

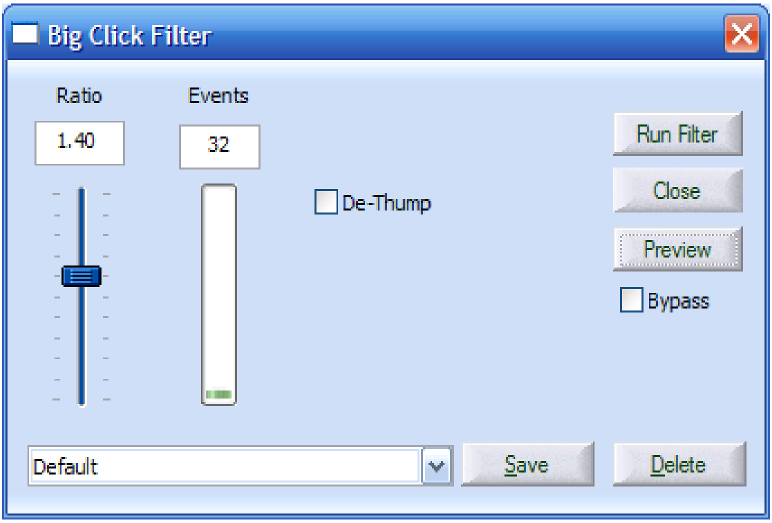


Figure 57 – Big Click Impulse Filter

This filter is very useful for fixing big clicks caused by phonograph records that are cracked or have been glued back together or have major gouges on their surface. It is also very useful for interpolating some of the long lived noises found on 16 inch acetate transcription recordings. Finally, it has a Forensics audio application in that it can attenuate cell phone noise interference from audio recordings. The BCF will not remove nominal clicks, ticks or crackle from a recording; use the EZ Impulse filter to remove those noises after running the BCF. Big clicks should always be removed as the first step of the de-clicking process before attempting to remove smaller impulsive noise using any of the other Impulse filters. The interpolation portion of the algorithm uses a frequency domain technique. Big Click Signals which last up to 200 mSec are able to be handled by this filter. A minimum impulse size of 2 mSec can be handled by the BCF. Very large clicks and thuds sometimes excite the resonance of a tone arm leaving behind a long-lived low frequency "tail" or "thump". This tone arm resonant "thump" can be dramatically reduced by checking the checkbox labeled

"De-Thump" which rejects the de-clicked "tail" for a period of time of 150 mSec past the end of the interpolation portion of the Big Click Filter algorithm. It is important to note that the "De-Thump" function will only remove "Thumps" that are preceded by a large click event; it is not capable of removing stand-alone thumps from recordings.

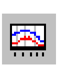
The Big Click filter has the following controls and indicators with the following ranges of adjustment:

- Ratio: 0.75 to 2.5 (Lower Values generally produce more aggressive results)
- Events: Records the number of events found and repaired by the filter
- Events Bar Graph: Indicates the number of events being found and repaired per unit of time.
- De-Thump Checkbox: Attenuates resonant ring-out tails following the repair time interval associated with this filter. Only use this function for the repair of large clicks associated with cracked or broken phonograph records. This function can't be "hot-switched" while previewing.

A demo .wav file is provided in the Wavefiles folder within your Diamond Cut Directory called "BigClickCracked78Demo.wav". Use this file to experiment with the Big Click Filter.

Note: Always use the BCF as the first step in your restoration process (assuming that you have a Big Click Issue). Use the other Impulse Filters as the next step(s).

Continuous Noise Filter



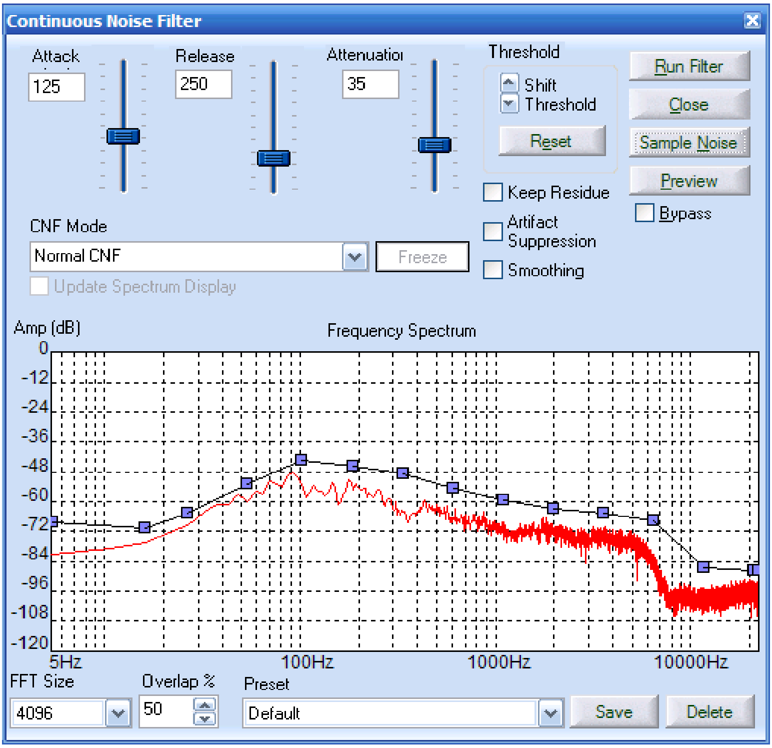
The Continuous Noise Filter is useful for reducing background "Hiss" and other constant noises from a recording or from a noisy AM or FM radio transmission. Magnetic tape recordings and phonograph records all contain a residual background "Hiss" type of noise that can be reduced with this filter. It is referred to as a "Continuous" noise filter (or CNF) because unlike impulse noise, hiss is present at all times. When adjusted properly, this filter can almost completely eliminate all residual continuous noise from a recording. However, it is easy to overuse this filter and leave the recording sounding dead and lifeless, and also introduce digital artifacts into the signal (music or speech).

The Continuous Noise Filter has 4 modes of operation to choose from:

- Normal CNF Mode
- Spectral Subtraction Mode
- Auto Spectrum CNF Mode
- Forensics AFDF Mode

To use this filter in Normal CNF or Spectral Subtraction modes, you must first take a sample of a section of noise from your file. This noise template will then be used by the algorithm to discriminate between what constitutes noise and what constitutes music or speech during the filtering process. It is important to sample a section of the .wav file that does not contain any music/speech so that the filter does not remove signals that contain musical or speech information. Of course, it can be used in the opposite sense too. If you sample on a sound that you do not want other than noise, it will attenuate that sound from the final result.

Note: We strongly recommend trying the Artifact Suppression mode when using this filter; its performance is often much better than standard mode and allows for a more aggressive use of the Attenuation Control without the production of digital artifacts.



The Continuous Noise Filter

The CNF graphically shows a frequency spectrum of the sampled noise in red (sometimes referred to as a noise-print or noise fingerprint). This spectrum represents the amount of noise at each frequency band in the recording. The blue line represents the filter design that has been created by the software program. You can use the mouse to move the blue threshold line to tailor the kind of noise reduction that the filter performs to your taste.

This filter should only be used on recordings that have little or no impulse noise, or on recordings that have already been processed through the Impulse Noise Filter(s) in order to minimize the possibility of producing digital artifacts. When operating this filter in the Auto Spectrum CNF Mode, the system will automatically find and modify its own noise fingerprint on-the-fly. Therefore, there is no need to manually take a noise fingerprint when operating in the Auto Spectrum CNF Mode.

This is one of two different types of non-linear filters that can be used to reduce noise from a signal source. Like the Dynamic Noise Filter, it is useful for reducing "Hiss" from a recording or from a noisy FM radio transmission. However, unlike the Dynamic Noise Filter, it will also reduce lower frequency noise. When adjusted carefully, it can almost completely eliminate all residual noise from a recording. However, when compared to the Dynamic Noise Filter, this filter is a bit trickier to adjust so as to avoid the introduction of digital noise artifacts into the Destination .wav file. It also can have some detrimental effects on the "presence" and the musical transient content of a .wav file when not properly adjusted.

This filter takes a sampling of your file and converts it into the frequency domain utilizing a Fast Fourier transform. Next it marches along to the next time interval and performs another Fourier transform. It keeps repeating this process until the entire .wav file is converted into samples which are no longer representing the time domain, but strictly represented in the frequency domain, with the appropriate Voltage, phase and frequency co-efficients for each window contained in memory. The entire audio spectrum is divided into up to 8,192 bands by a 16,384 point fast Fourier transform (FFT) algorithm (the resolution of which is user selectable). When a signal in a particular band exceeds a threshold (*when operating in the Normal CNF or the Auto Spectrum CNF Modes)* that you can define graphically, then that particular band is allowed to pass its signal from the input of the algorithm to the output of the algorithm. Lastly, the entire file is then re-converted back into the time domain via an inverse fast Fourier Transform. So effectively the only time during which bandwidth is provided at any of the selected number of frequency buckets is when there is a useful signal present in that portion of the audio spectrum. Otherwise, the remaining frequency bands are attenuated to a varying degree (depending on the Attenuation Control setting).

Insert Additional Control Points onto the Continuous Noise Filter

By simply right clicking anywhere on the Frequency Spectrum of the Continuous Noise Filter, you can insert your own additional Control

Point(s) onto the graphical display. This can help increase the flexibility and power of the filter. Upon right clicking of your mouse, the menu that appears allows you to add a point, delete a point, or reset the control point count to the factory default setting of 11.

The following is a summary of the control parameters functionality and range of adjustment provided by the Continuous Noise Filter:

- **Attack Time**

This is the time required for any of the filters to "open up" on the leading edge of a signal that exceeds the threshold line on the spectral graph.

This represents the time constant normalized value at 1 kHz. The time constant for filter frequencies operating above 1 kHz will be shorter than the setting, and the time constant for filter frequencies operating below 1 kHz will be longer.

(The Attack time constant value is weighted with a -1 slope across the audio spectrum.) Small values of attack provide excellent transient response, while long values provide a minimization of digital artifacts produced by the system. A good value to start with for the Attack Time parameter is around 100 Milliseconds. The total range of adjustment for Attack time is 1.0 to 300 milliseconds. Smaller settings will improve transient response but allow more digital artifacts to pass through. Larger values will decrease transient response, but will minimize the production of artifacts during the noise reduction process.

- **Release Time**

This is the time allowed for any of the filters to "close down" or "decay" following a signal that falls below the blue threshold line on the spectral graph. All remaining characteristics of the Release time Constant are the same in nature as the Attack time Constant. A good value to start with for the Release Time parameter is around 200 Milliseconds. If there is too much fast filter "breathing" (a digital artifact that is also sometimes referred to as “pumping”), lengthen this time until you are satisfied with the result. The total range of adjustment for Release time is 1.0 to 1000 milliseconds.

- **Attenuation**

This control sets the degree of attenuation for signals that are present and below the blue threshold line. The greater that one sets the Attenuation control, the greater will be the degree of noise reduction. However, the greater the degree of noise reduction achieved, the greater will be the loss of the sense of "Ambiance" on the resultant recording. So you must make a careful judgment as the correct tradeoff between noise reduction and ambiance for the material you are dealing with. A good value for the Attenuation parameter is around 10 dB to start with. If there is too much loss in signal ambiance, decrease this value. If you desire more noise reduction, increase this value. If you increase the attenuation too much, you will begin to introduce some digital aliasing artifacts into the Destination .wav file. The total range of adjustment for Attenuation is 0 to 100 dB.

- **Threshold (Blue Graphical Threshold Line)**

This feature controls the threshold value above which a signal at a particular frequency must exceed before it is passed through to the output of the algorithm without attenuation. It works in conjunction with the "sample noise" button. Although the continuous noise filter has up to 8,192 discrete frequency bands, it would be inconvenient to have to set each of them. DCArt10/DC Forensics10 provides up to 256 inflection points (shown as blue dots connected by blue lines on the graph of amplitude vs. frequency) that can be moved along both the frequency and the amplitude axis. The software will place these inflection points automatically at approximately 10 dB above the noise floor after you perform the "sample noise" function. Try these settings first to find out if the results are acceptable. Thereafter, if there seems to be some noise that needs more attenuation at a particular frequency, adjust the threshold upwards utilizing the left mouse button at the frequency of interest until you are satisfied with the results. The graphical threshold line is adjustable "live" when preview mode is enabled. If you want more than the 11 inflection points provided by default, double click using the left mouse button while pointing at the desired position on the graph. To remove inflection points, use the right mouse button in a similar manner.

- **Threshold Control Grouping**

    a) Up & Down "Shift Threshold" Control - This feature allows you to globally shift the entire threshold line up or down independent of frequency. The feature consists of an up and down arrow box. It is operated via the left mouse button. The amplitude resolution of the control is 4 dB / click. After clicking on either of the arrows, you will see the entire threshold line shift in the direction of the chosen arrow.

b) "Reset" Control - This button will restore all of the threshold line inflection points to their original default settings.

- **Keep Residue Function**

When enabled, the "Keep Residue" function will allow you to "preview" (hear) or process to the Destination Workspace the algebraic difference between the Source File and the Filters Output. In essence, you will be listening to the noise which would have been removed from the Source File had this function not been enabled. It is sometimes useful (via this function) to be able to hear just how much of the real audio signal along with the noise components that you are removing from the source signal. However, it is only fair to warn the user not to make final adjustments using this feature, as that technique can be quite deceptive. It is always best to optimize your parametric settings for the best results while listening to the actual processed filter output signal (i.e. "Keep Residue" function off).

- **Artifact Suppression Mode**

This mode of operation reduces the level of digital artifacts (sometimes referred to as “the birdies”) produced by the CNF during its noise reduction process, especially when it is being used aggressively. It can also be used to effectively reduce some forms of inter-modulation distortion (IM Distortion) from a recording (like that “raspy” sound found on some distorted 45 RPM records - - - especially on vocals). When operating the CNF in Artifact Suppression Mode, the Attack and Release functions are eliminated in that the routine operates independently of those parameters. The “Attack” control will become grayed out and the Release control will revert to a new mode of operation simply called “Artifacts”. The artifact suppression mode not available when operating in the Auto Spectrum CNF mode. In artifact suppression mode, use the CNF as you normally would, including the taking of a noise print sample in the other two modes of CNF operation (normal mode and spectral subtraction mode). The Artifact Suppression control affects the degree to which the system attenuates digital artifacts with higher settings providing a more aggressive action. You should note that higher levels of "Attenuation" are achievable in Artifact Suppression Mode compared to non-Artifact Suppression mode. 40 to 50 is a good setting to start with for the Attenuation control with an FFT size set to 4096. Experiment with FFT sizes on each side of the recommended value to find the optimal result on any particular file. 200 is a good setting to start with for the Artifact Suppression Control. Adjust this control upwards for an increased artifact reduction effect and downwards for a reduction in the removal of audio material (ambience and transients, etc). Use this control to find the best balance between those two sonic characteristics of the particular material that you are working with. Note that the optimal settings for the mentioned controls are substantially source material dependent. Using the "Keep Residue" mode will allow you to monitor how much audio material is being removed from the signal by the CNF Artifact Suppression system.

The Artifact Suppression system is not functional in Auto Spectrum mode but does operate in Normal mode, Spectral Subtraction mode and Forensics AFDF mode. Also, it is to be noted that fairly high FFT sizes produce more optimal results with this system. FFT sizes below 1024 are not recommended with the Artifact Suppressor because the system is less effective due to the poor frequency resolution associated with reduced FFT count. It is worth noting that the Artifact Suppression mode requires much higher levels of CPU resources and thus takes around 4 times longer to process a given file compared to the normal (non-Artifact Suppression) mode.

- **Smoothing Mode**

An alternative to Artifact Suppression mode is Smoothing Mode. The smoothing checkbox applies some additional signal processing that reduces the digital artifacts or "Musical Noise" that may be heard when applying the Continuous Noise Filter by averaging adjacent frequency bins within an FFT. This feature is most useful on very noisy recordings with high levels of surface noise such as old 78s or extremely noisy Forensics situations. When using the smoothing function, you should be able to increase the level of noise reduction without introducing digital artifacts by about 3 dB or more. The tradeoff (and there are always tradeoffs), is that the frequency selectivity is reduced in this mode (sometimes resulting in reduced bass response). An example of where the smoothing function would not be useful would be when you are trying to remove pure tones or steady state buzzing sounds where you need to maintain a high degree of frequency selectivity. Please note that this function can't be used in conjunction with the Artifact Suppression mode. It is an alternative to Artifact Suppression mode.

- **FFT Size (Resolution)**

Choose between 32, 64, 128, 256, 512, 1,024, 2,048, 4,096, 8,192, and 16,384.

The frequency resolution of the Continuous Noise Filter can be adjusted. This parameter determines the number of frequency bins used by the FFT algorithm. The actual number of frequency bins produced is the FFT Size divided by two (since the FFT produces data for both the real and imaginary planes). Large values of resolution produce the largest degree frequency selectivity and thus high degrees of noise reduction, while the best time domain transient response will be realized with smaller FFT values. Put another way, there is a tradeoff between frequency resolution and time resolution and they are inversely related to one another. You will have to experiment with the various binary weighted values to determine the best resolution for the material that you are dealing with. Listen for the best levels of noise reduction attainable while minimizing any digital artifacts and yet maintaining good musical transient response on things like rim shots on drums, and other percussive instruments.

Important Note:

This parameter cannot be adjusted while the filter is previewing or running.

- **Overlap (Window Overlap – Forensics Version Only)**

This allows you to choose the percentage of overlap between FFT windows. The range for this adjustment is 30% to 50%. Often, reduced values of % overlap will yield better results in terms of digital artifacts produced, but will take longer to process. Conversely, larger values of overlap will produce more digital artifacts, but take less time to process. Start with 50% and lower it order to achieve faster sonic transients tracking. Note that this feature is only available in the DC Forensics10 Audio Laboratory version of the software. The overlap parameter is fixed at 50% in the DCArt10 version of the software and is not visible to the user.

- **Spectral Subtraction Mode In The Continuous Noise Filter**

This changes the mode of the Continuous Noise Filter into a spectral subtraction system making it especially useful for some Forensics noise reduction applications. As with the normal mode of the filter, you will still need to take a fingerprint of the noise when operating in Spectral Subtraction mode. The amplitude signal of the resultant FFT is then subtracted from the entire .wav file signal thereby providing rejection of the highlighted fingerprinted signal. The offending signal could be an air conditioner, city noise, automobile noise, etc. To adjust the depth of effect, use the attenuation control. The attenuation slider sets the maximum amount of attenuation as an absolute number (as in all of the CNF modes) and in Spectral Subtraction mode the threshold shift scales the overall gain of the noise sample. Threshold and attenuation are not equivalent parameters because of the fact that the maximum attenuation value is absolute but it is not a maximum delta value between the signal and the sampled reference.

<u>Important Note:</u>

If the parameters for the Continuous Noise filter are set incorrectly, it has the propensity to produce extremely strange sounds (digital artifacts) - - - some of them quite comical in nature. If you hear these "birdies", back down on the attenuation setting and / or some of the graph inflection points and they should disappear. If not, increase the Attack and Release time values.

- **Auto Spectrum CNF Mode**

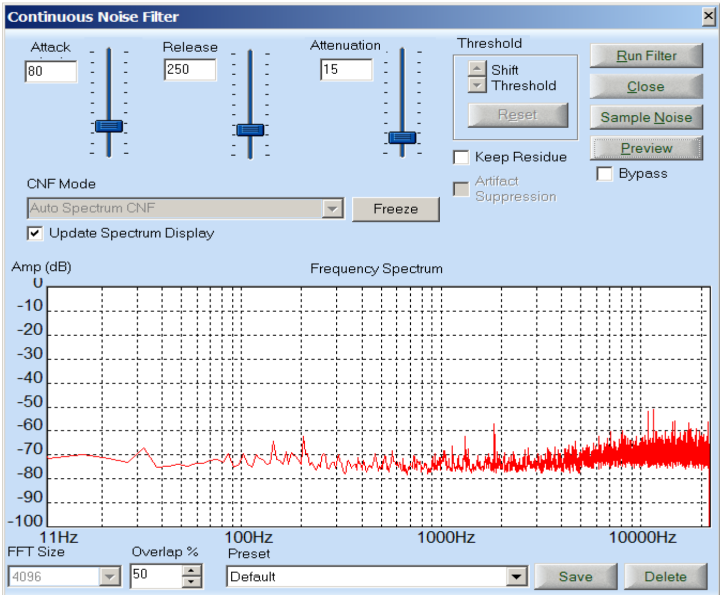
Sometimes, it may be desirable to have the Continuous Noise reduction system determine its own noise fingerprint “on the fly.” There are three possible reasons why you might be interested in using this mode:

  - It simplifies the operation of the Continuous Noise Filter
  - There may be no discernable quiet passage on the source material from which to “Sample the Noise Fingerprint”.

- The Noise Fingerprint may be changing in its noise distribution dramatically throughout the .wav file.

The Auto Spectrum CNF Mode solves these problems by calculating the noise fingerprint “on the fly” and updating it on a continuous basis. It is capable of performing this function even if you can’t see or hear a totally quiet passage in the target .wav file. The mathematical routine utilized to create this fingerprint is more sensitive than your ears and can always find a noise fingerprint. This system may not be quite as effective for removing noise compared to the other two modes of the CNF operation, but its convenience coupled with its ability to adapt to changing noise environments sometimes outweigh its noise reduction performance limitation. To operate the CNF in this mode, merely preview it and adjust (primarily) the attenuation, attack and release times for the best sounding results. As with all of the Diamond Cut filters, the “Keep Residue” mode can be useful as an adjustment tool. In “Keep Residue” mode, tune for maximum noise and minimum signal while operating in “Preview” mode. The Auto Spectrum CNF Mode has the following controls and displays:

- Attenuation (start with a setting of 25)
- Attack Time (start with a setting of 20 mSec)
- Release Time (start with a setting of 50 mSec)
- FFT Size (start with a setting of 8,192)
- Smoothing Checkbox (on)
- Overlap % (50 %)
- Update Spectral Display (on/off selector checkbox - - - your choice)



The Auto Spectrum CNF Screen

Forensics Adaptive Frequency Domain Filter (AFDF) Mode (Forensics Version Only)

The Forensics **A**daptive **F**requency **D**omain **F**ilter (AFDF) is a variation on the basic theme of the Auto Spectrum CNF filter mode. The primary difference is that the Forensics AFDF is optimized for Forensics oriented files and not "High Fidelity" files. It is found as one of the modes under the CNF filter or directly via an item called "Adaptive FD Filter" found under the Forensics Menu*. It has a faster response time compared to the Auto Level function and also a narrower effective bandwidth while producing higher levels of noise reduction at the expense of potentially producing higher levels of digital artifacts. The AFDF Filter is "Adaptive", which means that it will automatically adjust itself to varying noise environments. This filter has a Time Domain twin sister found in the Forensic Menu, which is simply called the "Adaptive Filter". The AFDF can produce more attenuation of noise compared to the Time Domain Adaptive Filter but can also produce more spurious digital artifacts. The Time Domain Adaptive filter does not produce spurious digital artifacts, but can produce an echo delay sound as an artifact. We provide both types of Adaptive filters since they exhibit somewhat differing sonic characteristics. Experimentation is the best way to determine which filter is the most effective on a given Forensics audio file. Also, a combination of the two used together in cascade in the Multi-Filter has been shown to work some magic that otherwise would not be possible on some extremely noisy forensics files. In general operation, the controls and the display graph for the Forensics AFDF is much the same as that of the Auto Spectrum CNF. Refer to it for details.

*Note 1: When you need to use the AFDF in the Multifilter, you must access it via the appropriate CNF mode inside in the CNF.

Note 2: The “Green Zone” spans for the various slider controls will vary depending on the chosen CNF mode and/or whether Artifact Suppression is active or inactive.

Continuous Noise Filter Procedure (Tutorial for Normal CNF Mode)

The Continuous Noise Filter is one of the most mathematically complex of all of the DCArt10/DC Forensics10 algorithms. It will, therefore, take the longest amount of processing time to complete its calculations. This algorithm will benefit the most from the use of a high clock rate computer. This filter is also the most difficult filter to use correctly. Aliasing artifacts can be produced when the settings are not correct for the particular .wav file you are attempting to "de-noise." The first time you use it, it will be worthwhile to spend about an hour playing around with it in order to become familiar with its behavior.

1. Highlight a quiet portion of the Source .wav file. Often, this sector will be found at the beginning or at the end of the file, as with the lead-in or the lead-out groove sector of a record or the lead-in portion of a tape recording. The idea here is to capture a section of noise only, but no signal. This will become the noise floor baseline for the subsequent operation of the Continuous Noise Filter. Only about one or two seconds of noise is needed.
2. With the left mouse button, click on the "Filter" menu.
3. Next, click on "Continuous Noise”.

4. When the Continuous Noise Dialog Box appears, click on "Sample Noise."
5. Some calculations will be made in the ensuing moments. When they are complete, a graph will appear showing the Amplitude (in dB) versus the Frequency of the .wav file noise floor. This graph represents the Noise Print of the file on which you are working.
6. The measured sample noise spectrum is shown in red. The noise threshold value versus frequency is shown in blue. You can set the blue graph threshold value, although DCArt10/DC Forensics10 will automatically choose some settings for the threshold limit line that is a good place to start.
7. If you choose to change the graphical threshold contour, follow the procedure outlined in steps 7 through 10. Using your mouse, place the pointer on the left-most blue threshold marker on the graph (one of ten blue dots).
8. Depress the left mouse button and move the dot either up or down so that it remains somewhere above the red line graph at the bottom end of the spectrum. The higher this line is from the red line, the greater will be the degree of noise reduction at frequencies near this particular dot. If the dot is placed below the red graphical line, no noise reduction will be applied to these frequencies. This is sometimes the preferable setting for the blue threshold line near the bottom end of the audio spectrum (below a few hundred Hertz).
9. Next, move the next blue threshold marker just to the right of the first one, and using the mouse, set it somewhere above that particular frequency on the spectrum graph.
10. Repeat this process until all 11 threshold markers are located somewhere above the "noise floor" graphical representation of your .wav file. Now the blue line should be located above the red line at all frequency locations. Note that the best contour can only be achieved by not only moving the markers along the vertical axis, but along the horizontal (frequency) axis as well.
11. Set the "Attack" time initially to 20 milliseconds.
12. Set the "Release" time initially to 50 or 100 milliseconds. (The "Release" time constant should always be set longer than the "Attack" time constant for a realistic sounding operation of the filter.)

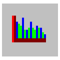
13. Set the "Attenuation" control initially to 10 dB. (Higher numbers results in higher levels of noise reduction.) Too much noise reduction will produce digital artifacts and detract from the "ambiance" of the recording.
14. Highlight the portion of your .wav file on which you desire to apply the Continuous Noise Filter. (You may choose to highlight the entire file or any portion thereof.)
15. Preview (or Run) the Filter.
16. Listen to the result and then determine which parameters need modification. If there is a "lagging" response to audio signals, decrease the "Attack" time. If there is a "swell" of noise following an audio crescendo, decrease the "Release" time. If a portion of the audio spectrum is sounding dull, lower the "threshold" line at the frequency of interest. If a portion of the audio spectrum is sounding noisier compared to the rest, then raise the "threshold" at the frequency range of interest.
17. When you are satisfied with the results, Run the .wav file (in its entirety) to achieve the final processed results in the "Destination" workspace.

Note 1: The threshold line inflection points can be adjusted "live" when you are running the filter in "Preview Mode." You will be able to hear the effects of modification that you make to the threshold line almost immediately.

Note 2: Sometimes, record albums that are in excellent condition exhibit only one type of noise which is called “Rumble”. Rumble comes from the record mastering process as well as from the bearings in your turntable. Generally, one uses a High Pass filter to remediate this problem, but that approach also cuts into the low pass portion of the audio signal. There is a special preset in the CNF called “Dynamic Rumble (Only) Filter” which will remove this type of noise without damaging the deep bass on the recording or affecting any other signal above 90 Hz. This filter is more effective on Rumble noise than the standard High Pass Rumble filters found elsewhere in this software program.

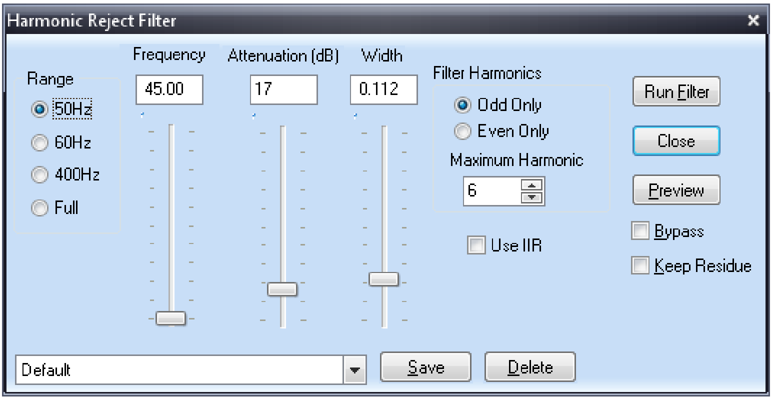
Note 3: The most common issue with the use of the CNF is setting it too aggressively. It’s nice to reduce noise, but not to aim to completely eliminate it, as that approach will produce digital artifacts.

Harmonic Reject



The Harmonic Reject filter, which is sometimes referred to as a "comb" or "multiple notch" filter is used to attenuate periodic noises that contain harmonics. It is capable of attenuating either the odd or even harmonics of the selected fundamental frequency. This filter is very effective for attenuating a form of hum (line frequency related) noise that is rich in harmonics. This type of noise is sometimes referred to as "Buzz." Noise, such as this can be introduced into an audio recording from sources such as light dimmers or switching mode computer power supplies. "Buzz" can contain odd harmonics of the power line frequency, all the way through the entire audio spectrum. A 60-Hertz square wave is, by definition, the fundamental component of the waveform plus all the odd harmonics of that frequency up to infinity. If the waveform is perfectly symmetrical, it will include a fundamental component and only its odd harmonic multiples. The various bands (tines) of the Harmonic Reject filter will attenuate the fundamental as well as all of the harmonic by- products within the audio spectrum. Should you encounter wide-band line frequency related noise that is asymmetrical, it can produce some even harmonics.

An example of noise containing even harmonics would be that produced by an unsymmetrical sine, trapezoidal or square waveform (e.g.- a square wave having non-symmetrical duty cycle values). Non-linear devices in a signal pathway can introduce this effect. The Harmonic Reject filter can be placed in a mode in which it will attenuate the "evens" rather than the "odds" if you should encounter such noise. Before using the Harmonic Reject Filter, it is often useful to identify the fundamental frequency of the noise signal by using the Spectrum Analyzer in a high frequency resolution mode of operation (found under the "View" menu).



The Harmonic Reject Filter

The following is a summary of the control parameters and range of adjustment provided for the Harmonic Reject Filter:

1. Frequency (Fundamental): 20 - 5,000 Hz
2. Attenuation: 1 to 100 dB
3. Filter Harmonics:
   A. Odd Only
   B. Even Only
4. Maximum Harmonic (Number): 0 to 500
5. Width (or Q): 0.005 to 0.5 Octaves
6. Range: (This control optimized the resolution of the Frequency Control) - Choose between 50 Hz, 60 Hz, 400 Hz (aircraft) or Full Range
7. Use IIR Checkbox: Changes the mode from an FFT based system to a Resonant IIR Multiple Notch Filter type

This filter also incorporates a "Keep Residue" feature. This allows you to hear or keep only the noise component of the original signal. This feature is useful for "tuning" the filter to the maximum level of noise, so that when you actually run the filter with the "Keep Residue" feature turned off, the noise left behind will be minimized. Lastly, the keep residue function also allows you to produce "slot" filters, provided the slots that you need are harmonically related. See Appendix 1 for more details about slot filters.

Fine Tuning the Harmonic Reject Filter

Let's assume that you have a .wav file having a buzz which contains roughly a 60 Hz fundamental frequency. This can be determined by applying the spectrum analyzer to the file and making a measurement of the first spike in the series. Set the Range (checkbox) to 60 Hz. Adjust the frequency of the Harmonic Reject filter in the range field to correspond to the reading viewed on the Spectrum Analyzer (direct numerical entry is a good way to do this). Start with a width of 0.2. Set the filter harmonics to 5 or 6 and check the "Keep Residue" mode. Next, click on the preview button and then point your mouse to the frequency slider. You should be hearing mostly buzz and not much signal in this mode. Now, use the up and down keys on your keyboard to move the frequency first above 60 Hz and then below 60 Hz one small increment at a time. Keep on adjusting the frequency in small increments until you hear the loudest buzz coming from your sound system. When you have found the maximum loudness value, switch back to non-keep residue mode. You will hear the signal with the buzz substantially attenuated. The next thing to do is to adjust the Maximum Harmonic number to the proper value required to attenuate the buzz without degrading the quality of your audio signal. Try increasing it until no further improvement is observed. If necessary, try decreasing it too. Do not use maximum harmonic numbers greater than necessary to accomplish noise reduction. The use of too many harmonics will damage your audio signal. Lastly, adjust the Width control to the minimum value that will do the job for you so as to minimize damage to the primary signal of interest.

Sometimes, a better result is achieved using the IIR (resonant notch filter) technique. You can enter that mode via the IIR checkbox. This allows a much higher frequency resolution adjustment capability and so you may be able to more closely hone in on the target signal of interest. However, when using the IIR method, try to minimize the number of harmonics to no more than what is needed to get the job done (Maximum Harmonic). The IIR method is much more math intensive and thus, slower, especially with high values of the Maximum Harmonic settings.

Important Notes:

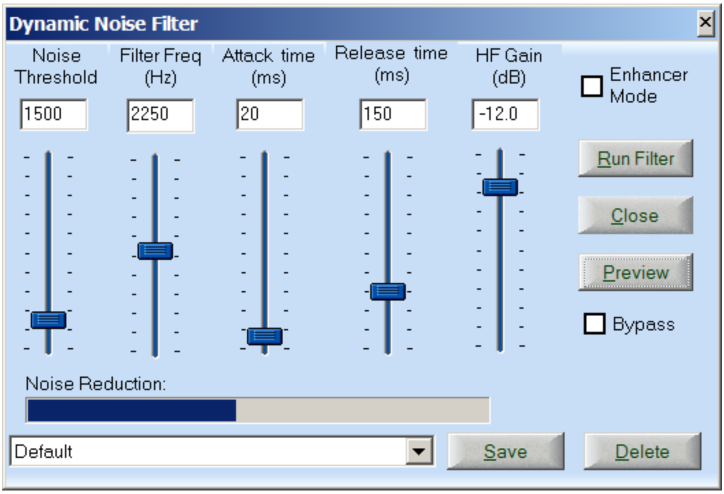
For severe 50 or 60 Hz buzz situations, run at least two passes of the filter. The first pass should be run with a setting of 50 or 60 Hz, odd, and the desired Maximum Harmonics. On the second pass, run with a setting of 25 Hz (for 50 Hz situations) or 30 Hz (for 60 Hz situations), odd, and a lower Maximum Harmonics number. This sequence can be repeated more than once for further buzz reduction. You should also try running the system in “even only” mode after processing the odds first. Don’t forget that you can stack up (cascade) 2 or more of these filters in the Multi-Filter and apply them all at once as a convenience.

Note: Sometimes, when line-frequency related “buzz” is encountered containing only very high frequency components (spikes) the EZ Impulse or Expert Impulse Filter may perform a better job removing buzz and doing less damage to the main audio signal. Experimentation is the only way to know what filter will produce the best result on any given file.

Dynamic Noise Filter

(Analog Noise Filter)
This is a digital simulation of a dynamic analog filter. It is useful for dynamically attenuating "Hiss" from old record recordings or from old magnetic tape recordings. It performs better than a fixed Low-pass filter because it only attenuates high frequencies when there is no high frequency information present above the setting of its "Noise Threshold" slider adjustment. Sometimes this technique is referred to as "single-ended noise reduction". The Dynamic Noise filter's Low-pass corner frequency is frequency modulated by a rectified envelope signal that represents the amplitude of the signal content above a particular low-pass corner. So, normally, the bandwidth of this filter is limited until some high frequency content is measured by its high frequency detector. When this occurs, the bandwidth of the filter is opened up to allow the frequency of interest to pass through. When the high frequency signal diminishes again below a threshold value, the filter closes back down to a smaller bandwidth. The user has the ability to adjust a number of parameters with this filter, including noise threshold, filter frequency, attack time (the time constant associated

with the signal whose job it is to increase the Low-pass filter frequency corner), release time (the time constant associated with the signal whose job it is to decrease the Low-pass filter frequency corner after a high frequency event has ceased) and HF Gain (high frequency gain). This filter should only be used on recordings that contain little or no impulse noise, or on recordings that have already been processed through the Impulse Noise filter first to minimize unnecessary filter “breathing” effects. This Filter can be used either as a dynamic noise reduction tool or as a dynamic high frequency enhancer, or both if two are placed in cascade within the Diamond Cut Multi-Filter.



The Dynamic Noise Filter

The Dynamic Noise Filter provides the following slider controls:

- **Noise Threshold** (0 to 10,000)

The Noise Threshold is the value of rectified and averaged High-pass signal level above which the output of the Dynamic Noise Filter starts to raise its corner frequency. Moving the noise threshold slider control vertically raises its value. This control must be adjusted so that when there are no highs present in the source material, background "Hiss" is attenuated, but when "highs" are present (such as cymbal crashes or the pronunciation of the letter "S"), the filter "opens up" and passes those “highs” to the output of the system.

- **Filter Frequency** (200 – 19,999 Hz)

This is the 1st order High-pass corner filter frequency which drives the Dynamic Noise Filter Detector / Rectifier / Attack & Release Time Constant circuitry. For modern reel-to-reel tapes, this parameter will be operated generally up in the 4 to 6 kHz range. For early 78s, it will be operated in the 1 to 3 kHz range. The Filter Frequency parameter range is from 200 Hz to 19.999 kHz. Operation of this setting above 10,000 Hz is rarely of any value, but it is allowable. Experimentation will be required to determine the best setting and will vary greatly depending on your source material.

Important Note: The entire frequency range of adjustment up to 19,999 Hz is only possible when utilizing a 44.1 kHz sampling rate or higher. At a 22.05 kHz sampling rate, the maximum effective frequency setting will be 10 kHz, and at an 11.025 kHz sampling rate, this value will drop to 5 kHz.

- **Attack Time** (1 to 300 mSec)

The Attack Time slider adjusts the time constant (in milliseconds) associated with the rising edge of a High-pass signal envelope. Fast music will require smaller values of attack time compared to slow music. The range of adjustment for the Attack Time parameter is 1 to 300 milliseconds.

- **Release Time** (1 to 500 mSec)

The Release Time slider adjusts the time constant (in milliseconds) associated with the falling edge of a High-pass signal envelope. This parameter will also require smaller values for fast music compared to the requirements of slow music. Also, the release time will almost always be set to a period of time greater than the Attack time for the algorithm to sound natural. The range of adjustment for the Release Time parameter is 1 to 500 milliseconds.

- **HF Gain**

Gain controls the amount of dynamic High-pass filter signal that is summed back into the output of the filter. This allows you to obtain upward or downward expansion of the high frequency portion of the audio spectrum. The "neutral" setting for this would be 0 dB that represents no expansion or compression. Setting this value greater than 0 dB will be produce a "Spectral Enhancer" function. Values of 0 dB and lower produce a Single Ended Noise Reduction function used to de-Hiss a sound source. This control is calibrated in dB. Its range of adjustment in normal noise reduction mode is from 0 to -90 dB. Its range of control in Spectral Enhancer mode is from 0 to + 10 dB.

- **Enhancer Mode**

Checking this box switches the Dynamic Noise Filter from Noise Reduction mode to Spectral Enhancer Mode. Enhancer mode provides upwards expansion of signals above Filter Frequency setting and its threshold setting. The degree of Enhancement is determined by the HF Gain control setting. Enhancer mode can add some life to dull recordings without introducing a great deal of noise into the signal (as would an ordinary Equalizer).

Important Note:

The controls can be adjusted "live" when the preview mode button is clicked.

Dynamic Noise Filter Operating Procedure (Tutorial)

Most of the parameter settings for this filter will vary considerably depending on the content of your particular .wav file. You will have to experiment to determine the values most to your liking. The values used below in the Procedure Example will get you started.

1. Highlight the portion of your .wav file on which you desire to apply the Dynamic Noise Filter. (You may choose to highlight the entire file or any portion thereof.)
2. Click on the "Filter Menu" with the left mouse button.
3. Click on "Dynamic Noise Filter."
4. Set the Noise Threshold slider all the way down; this is the minimum threshold position of the slider control.
5. Set the Filter Frequency to around 1.5 kHz.
6. Set the Attack Time to about 5 mSec. (Unless you are attempting to obtain some sort of special effect, the Release Time should always be set to a value greater than or equal to the Attack Time.)
7. Set the Release Time to around 50 mSec.
8. Set the Gain Control to - 6 dB as a starting point for noise reduction. (This control should only be set to higher positive

value numbers if a "Spectral Enhancement" effect is desired and Enhancer Mode is checked.) This Gain Control will modify the "effective bandwidth" of your recording by incrementally amplifying the high end of the spectrum when there are enough “highs” present to trip the detector. This can be used for noise reduction or to increase the "presence" of a recording, or to enhance the sound of a vocalist. If greater noise reduction is desired set the gain control to higher negative values. If greater enhancement is required, set the gain control to higher positive values (when in Enhancer Mode).

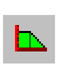
9. Click on "Preview".

When the filter is operating properly, "Hiss" will be reduced, but when there is high frequency content on the recording, the filter should "open up" and pass through more "highs" to its output. If the Threshold is set too high, the filter will never open up, and the .wav file will sound "dull" although "Hiss" may be reduced. If the Threshold is set to low, the filter will always be opened up to full bandwidth, and there will be no noise reduction action. If the Attack time is set too long, there will be a delay heard before the filter changes bandwidth on musical high frequency transients such as cymbal crashes (loss of the leading-edge). If the Release time is set too long, there will be a residual "Hiss" left behind after a high frequency musical event, which will decay out, but too slowly and therefore, unnaturally.
When you determine the best setting of the controls for your particular .wav file, click Run filter. When the filter has completed its operation, the results will appear in the Destination Workspace.

Low Pass, Band Pass and High Pass IIR Filter Sub-Menu

The IIR based Low Pass, Band Pass and High Pass Filters are all in a sub-menu called “LP, BP, HP Filters”.

Low Pass Filter

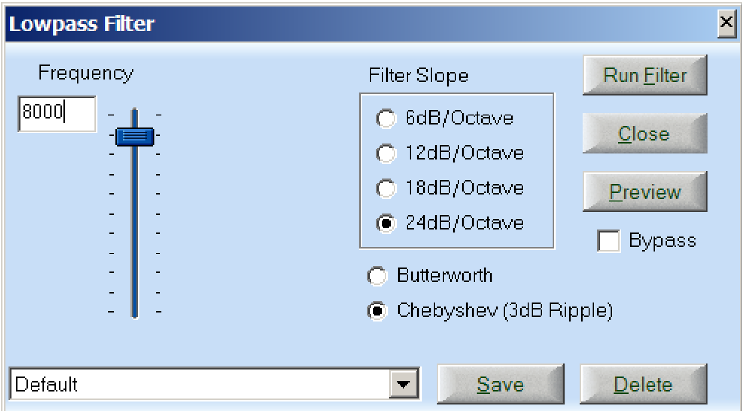


This filter is called a Low-pass filter because it only passes through signals that are lower than its set corner frequency. It attenuates high frequency signals above the corner frequency.

The effect can be similar to turning down the treble control on a home stereo except that the Low-pass filter is much more flexible. This filter can be somewhat useful for reducing hiss in a recording, but care must be taken not to reduce the "presence" of a recording by eliminating too much of the high end musical content at the same time.

Low Pass Filter with Chebyshev or Butterworth Response with up to a 4th Order Response (Slope)

This type of filter is most useful when a recording does not contain any useful sonic information above a certain frequency, and you wish to attenuate that high frequency noise that would otherwise be present.
This is a digital simulation of a conventional Low-pass filter. It is created using an Infinite Impulse Response (IIR) algorithm having a Butterworth or Chebyshev characteristic (for the higher order slopes). Frequencies below the "corner frequency" are passed through to the output, and frequencies above the corner are attenuated. The degree to which higher frequencies are attenuated is determined by the slope (order) of the filter. Four slopes are provided. They are 6dB / Octave, 12 dB / Octave, 18 dB / Octave and 24 dB / Octave. The higher the slope, the more attenuation will occur to frequencies above the corner frequency. The corner frequency is the frequency that you choose, and it is defined as the frequency at which the signal has been attenuated by 3 dB relative to the pass-band. This filter can be somewhat useful for reducing hiss in a recording, but care must be taken not to reduce the presence of the recording by eliminating too much of the high-end musical content at the same time. When used selectively, this filter can also be used either to "De-Ess" an overly sibilant vocal, or reduce harsh harmonic distortion products that may have resulted from occasional master recording overloading (clipping). For forensic recordings, this filter can be used to remove most noise sounds whose frequencies are above the speech spectrum by setting the corner frequency to somewhere between 3,000 to 4,500 Hz. The use of steep slopes (& Chebyshev) are often very useful in these forensics audio situations.



The Low Pass Filter

The higher order (12, 18, & 24 dB / Octave) Low-pass filters are of the Butterworth or Chebyshev types, depending on your choice.

The following is a summary of the control parameters and range of adjustment provided for the Lowpass Filter:

A. Frequency: 5 - 19,999 Hz
B. Filter Slope: 6, 12, 18 & 24dB / Octave
C. Preview Mode Button: On / Off (The slider control can be adjusted "live" when preview mode is on.)
D. Filter Type: Choice of Butterworth or Chebyshev

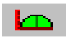
Important Note: The frequency range of adjustment up to 19,999 Hz is only effective when utilizing a 44.1 kHz sampling rate. At a 22.05 kHz sampling rate, the maximum effective frequency setting will be 10 kHz, and at an 11.025 kHz sampling rate, this value will drop to 5 kHz.

Lowpass Filter Operating Procedure (Tutorial)

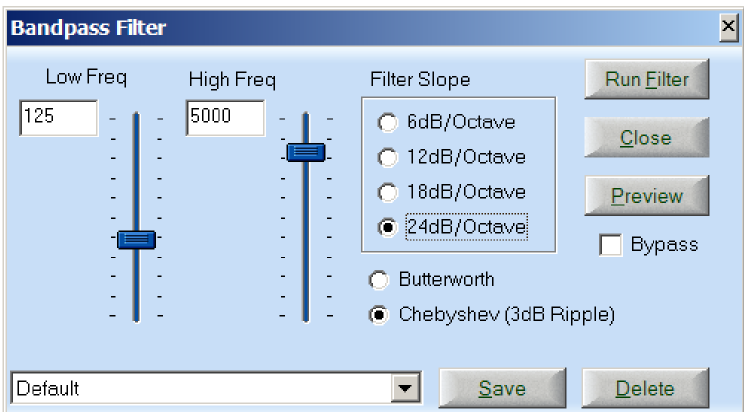
1. Highlight the portion of your .wav file on which you desire to apply the Low-pass filter. (You may choose to highlight the entire file or any portion thereof.)
2. Click on the "Filter Menu" with the left mouse button.

3. Click on "Low-pass."
4. Choose the "Frequency" above which you desire to attenuate all signals, utilizing the left mouse button in conjunction with the "Frequency" slider control. When the control is all the way down, the setting will be 5 Hz, and when it is all the way up, the setting will be 19.999 kHz. (Useful settings will usually fall somewhere within the 3 kHz to 15 kHz range, depending on the source material and the goals of the audio restoration.) If you desire finer frequency resolution, you may also use direct numeric entry of the value.
5. Choose the Filter "Slope" which you desire. Click on 6 dB / Octave, 12 dB / Octave, 18 dB / Octave or 24 dB / Octave. The steeper the slope, the higher will be the degree of attenuation of all frequencies above the "Frequency" setting.
6. If you desire to hear the results of your filter settings before creating a new "Destination" file, click on "preview."
7. After a short delay, you will hear the effect of the settings that you have chosen. (The system may seem to stutter if your computer is too slow to keep up with the algorithm in real time. However, this repeating pattern will not be present in the final Destination processing of the filter.)
8. As the filter is running in either Preview mode or Run mode (Destination File Mode), you will see a dialog box which indicates the "% Done" of the filter algorithm on the selected portion of the Source .wav file. Also, at the top of the Dialog box you will see indicated the "Total Samples to Process:"
9. Keep adjusting the Frequency and Slope parameters and testing the various settings using the "Preview" mode until you are satisfied with your results.
10. When you are satisfied with a group of settings, you will be done with Preview mode.
11. Click on “Run”, and the filter will process your Source .wav file through the filter algorithm, and create a Destination .wav file containing the output of the filter.
12. When this process is complete, you will see the Destination File become highlighted in Yellow, at the same time that the Source File becomes unselected.

Band Pass Filter



Band-pass filters are essentially a combination of a Low-pass filter and a High-pass filter operating together. It attenuates both the high frequency and the low frequency portions of the audio spectrum. It is useful where the recording contains extraneous noise in the low frequency region such as rumble or thumps, and high frequency noise such as hiss. This filter can also be very useful for improving the intelligibility of audio recordings, especially speech, by eliminating the unnecessary portion of the audio spectrum that is not used by speech frequencies to carry useful information to the listener.



The Band Pass Filter

Band Pass Filter with Chebyshev or Butterworth Response with up to 4th Order Slope

This is a digital simulation (IIR based) of a conventional analog Band-pass filter having a Butterworth or Chebyshev response when set to the steeper slope values. Band-pass filters are passive to frequencies within the Band-pass region, but they attenuate frequencies above and below the two corner frequencies. Band-pass filters have both an upper and a lower corner frequency, and like the Low-pass and the High-pass filter, the corner frequencies are defined as the frequencies at which the signals either above the upper corner or below the lower corner are attenuated by 3 dB. Four slopes are provided for the Band-pass Filter, just like the Low-pass and the High-pass. They are 6 dB / Octave, 12 dB / Octave, and 18 dB / Octave and 24 dB / Octave. This filter can be very useful for improving the intelligibility of audio recordings, especially speech, by only passing through signals in the portion of the audio spectrum involved in human speech. The Forensics Menu "Brick Wall" filter has a much steeper version of this filter for dealing with extreme cases of out-of-band noise that needs to be eliminated.

Note:

The higher order (12, 18, & 24 dB / Octave) Band-pass filters are of the Butterworth or Chebyshev type depending on your choice.

Special effects can be produced with the Band-pass filter. These special effects can be useful when producing movies or stage plays or shows and a particular sound producing device and its environment needs to be accurately reproduced through the "House" P. A. System. Here are a few examples:

Simulation	Low Freq. Control	High Freq. Control	Slope
1930's Vintage Table Top Radio:	830 Hz	2000 Hz	18 dB / Octave
Modern cheap Table Top Radio:	265 Hz	6100 Hz	18 dB / Octave
Loud "Walkman" personal stereo as heard by person nearby:	3650 Hz	9800 Hz	18 dB / Octave
Modern Stereo System as heard from the next room:	95 Hz	4100 Hz	12 dB / Octave
1950's Vintage Juke Box:	30 Hz	2700 Hz	12 dB / Octave
AM Transistor Pocket Radio:	1395 Hz	2110 Hz	18 dB / Octave
Telephone Receiver sound from "off the hook":	2700 Hz	2895 Hz	18 dB / Octave
Night Club Band as heard from Parking Lot:	85 Hz	240 Hz	12 dB / Octave
Old Acoustic Phonograph:	870 Hz	2390 Hz	18 dB / Octave
Public Address System at Outdoor Event:	300 Hz	3000 Hz	12 dB / Octave
Modern High End Audio System:	15 Hz	19,999 Hz	6 dB / Octave
Bandpass Filter Response Limits:	5 Hz	19,999 Hz	-

You can create your own simulations of sound devices and acoustic environments through experimentation with the Band-pass filter parameters. Using the above simulations, in conjunction with the DCArt10 / DCForensics10 reverb, you can further enhance various acoustical environments. Once you discover the appropriate values, write them down or store them as presets for future reference.

Cascading this filter (using the Multi-Filter) with others like the Virtual Valve Amplifier to add distortion, and the Reverb to add room acoustical effects can further embellish these sound simulations.

The Band-pass filter can also be used as a tool to determine if any useful audio information exists in a particular portion of the audio spectrum; it becomes sort of an audible wave analyzer when used in this manner. For more information on this mode of operation, refer to the "Using DCArt10/DC Forensics10 as an Audio Waveform Analyzer" portion of the "How Do I" section of this manual or go to the section which explains the operation of the Spectrum Analyzer.

The following is a summary of the control parameters and range of adjustment provided for the Band-pass Filter:

- Low Frequency: 5 - 19,999 Hz.
- High Frequency: 5 - 19,999 Hz.
- Filter Slopes: 6, 12, 18, & 24 dB / Octave.
- Preview Mode Button: On/Off (The slider controls can be adjusted "live" when preview mode is on.)
- Filter Type: Choice of Butterworth or Chebyshev

Note: The frequency range of adjustment up to 19,999 Hz is only effective when utilizing a 44.1 kHz sampling rate. At a 22.05 kHz sampling rate, the maximum effective frequency setting will be 10 kHz, and at a 11.025 kHz sampling rate, this value will drop to 5 kHz.

Band-pass Filter Operating Procedure (Tutorial)

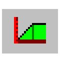
1. Highlight the portion of your .wav file on which you desire to apply the Band-pass filter. (You may choose to highlight the entire file or any portion thereof.)
2. Click on the “Filter Menu” with the left mouse button.
3. Click on "Band-pass."
4. Make an initial determination of what band of frequencies you desire to pass through the Band-pass filter.
5. Utilizing the right mouse button in conjunction with the Low Frequency slider control, select the lower corner frequency of the range that you have chosen. (The range for this control is 5 Hz to 19.999 kHz)
6. Utilizing the right mouse button in conjunction with the High Frequency slider control, select the upper corner frequency of the range that you have chosen. (The range for this control is 5 Hz to 19.999 kHz)
7. If you desire finer frequency resolution for either the lower or the upper corner frequency, you may use direct numeric entry, instead of the slider controls.
8. Choose the “Filter Slope" which you desire. This slope will symmetrically affect both the upper and lower corner roll-off rates. Click on either 6 dB / Octave, 12 dB / Octave, 18 dB / Octave or 24 dB / Octave. The steeper the slope, the higher will be the degree of attenuation of all frequencies outside of the selected pass band range.
9. If you desire to hear the results of your filter settings before creating a new "Destination" file, click on "Preview".
10. After a short delay, you will hear the effect of the settings that you have chosen. (The system may seem to stutter if your computer is too slow to keep up with the algorithm in real-time. However, this repeating pattern will not be present in the final Destination processing of the filter.)
11. Keep adjusting the Low Frequency and High Frequency sliders as well as the Slope parameters until you achieve the effect you desire and are satisfied with the results.
12. When you are satisfied with a group of settings, you will no longer need to invoke the Preview function.

13. Click on “Run”, and the filter will process your Source Wave file through the Band-pass Filter algorithm, and create a Destination .wav file containing the output of the filter.
14. When this process is complete, you will see the Destination File become highlighted in Yellow, at the same time that the Source File becomes unselected.
15. Click on "Close"

Note 1: If the low frequency control is set to a higher frequency than the high frequency control setting, a "no pass" filter will be created. This is of little useful value, but is allowable by DCArt10/DC Forensics10.

Note 2: You can select Chebyshev rather than Butterworth if you want a steeper filter response at the expense of some ripple within the filter pass-band.

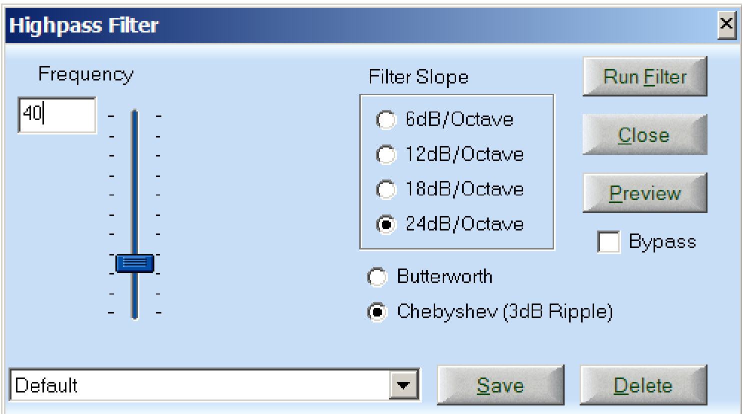
High Pass Filter



A High-pass filter only passes signals that are above or "higher" than the corner frequency. It reduces the level of low frequency signals that are below the corner frequency. The effect can be similar to turning down the bass control on a home stereo. This filter is very useful for reducing turntable rumble, muddiness, and any other extraneous low frequency noise in a recording.

High Pass Filter with Chebyshev or Butterworth Response with up to 4th Order Slope

This is a digital simulation (IIR based) of a conventional analog High-pass filter having a Butterworth or Chebyshev response (for the higher order slopes). This filter attenuates frequencies below the High-pass corner frequency. Just like the Band-pass and the Low-pass filter, this filter has four slopes available. They are 6 dB / Octave, 12 dB / Octave, 18 dB / Octave and 24 dB / Octave. This filter is very useful for reducing the effects of turntable rumble, or microphone seismic effects from inducing "muddiness" onto an audio recording. It can also be used to eliminate any dc (fixed) offset that may have developed on a .wav file (a preset is provided which performs this function). To reduce turntable rumble, start with a setting of 60 Hz and 18 dB / Octave, and then adjust the parameters until you are satisfied. This filter is also useful when selectively applied for reducing microphone "P" popping effects on the vocal track of multi-track recordings wherein an adequate windscreen had not been utilized in the session. Effective settings to attenuate “P” popping typically are 120 Hz with a slope of 18 dB / Octave (selectively applied to the highlighted event).



The High Pass Filter

Note: The higher order (12, 18, & 24 dB / Octave) High-pass filters are of the Butterworth or Chebyshev types, depending upon your selection.

The following is a summary of the control parameters and range of adjustment provided for the High-pass Filter:

- **Frequency**: 5 - 19,999 Hz.
- **Slope**: 6, 12, 18, 24 dB / Octave
- **Preview Mode Button**: On / Off (The slider controls can be adjusted "live" when the preview mode button is activated.)
- **Filter Type**: Choice of Butterworth or Chebyshev

Note: The frequency range of adjustment up to 19,999 Hz is only effective when utilizing a 44.1 kHz sampling rate. At a 22.05 kHz sampling rate, the maximum effective frequency setting will be 10 kHz, and at an 11.025 kHz sampling rate, this value will drop to 5 kHz.

High-pass Filter Operating Procedure (Tutorial)

1. Highlight the portion of your .wav file on which you desire to apply the High-pass filter. (You may choose to highlight the entire file or any portion thereof.)
2. Click on the “High Pass Filter”. (Filter Menu)
3. Choose the "Frequency" below which you desire to attenuate all signals, utilizing the right mouse button in conjunction with the "Frequency" slider control. When the control is all the way down, the setting will be 5 Hz, and when it is all the way up, the setting will be 20 kHz. (Useful settings will usually fall somewhere within the 15 Hz to 500 Hz range, depending on the goals of the audio restoration process.) If you desire finer frequency resolution, you may also use direct numeric entry of the value.
4. Choose the Filter "Slope" which you desire. Click on either 6 dB / Octave, 12 dB / Octave, or 18 dB / Octave or 24 dB / Octave. The steeper the slope, the higher will be the degree of attenuation of all frequencies below the "Frequency" setting.
5. Click on “Preview” to audition the results before processing.
6. After a short delay, you will hear the effect of the settings that you have chosen. (The system may seem to stutter if your computer is too slow to keep up with the algorithm in real time. However, this repeating pattern will not be present in the final Destination processing of the filter.)
7. As the filter is running in either preview mode or normal mode (Destination File Mode), you will see a dialog box that indicates the "% Done" of the filter algorithm on the selected portion of the Source .wav file. Also, at the top of the Dialog box you will see indicated the "Total Samples to Process:"
8. Keep adjusting the Frequency and Slope parameters, and testing the various settings using the "Preview" mode until you are satisfied with what you hear.

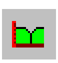
9. When you are satisfied, click on “Run Filter”, and the filter will process your Source .wav file through the filter algorithm, and create a Destination .wav file containing the output of the filter.
10. When this process is complete, you will see the Destination File become highlighted in Yellow, at the same time that the Source File becomes unselected.

<u>Note:</u> For a steeper filter response at the expense of elevated pass-band ripple, choose the Chebyshev rather than the Butterworth response.

Removing DC-Offsets with the High-pass Filter (Tutorial)

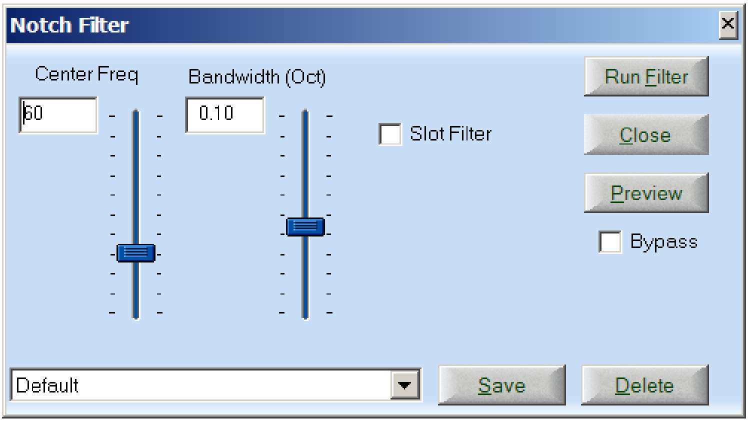
The High-pass Filter can be used to remove any DC-Offset from .wav file. To do so, set the High-pass Filter to 10 Hz and 6 dB per octave, and run it on the .wav file needing correction. This feature can also be found as a preset within the High-pass filter.

<u>Notch Filter</u>



A notch filter attenuates signals that are near its center frequency setting. The degree to which it attenuates frequencies near the center frequency is determined by the bandwidth setting. This filter is useful for removing 50 or 60 Hz “Hum” from a recording. It is also useful for decreasing any sound system acoustic feedback that may be found on some live recordings. A "Slot" filter is also provided within the Notch filter menu item for Forensics applications. Multiple slots can be constructed using the DCArt10/DC Forensics10 Multi-Filter. Slot filters have the reverse response of a Notch Filter; it only passes signals near its center frequency setting.

The Diamond Cut Notch Filter is a digital simulation (IIR based) of a second order notch / slot filter. It attenuates all frequencies near its center frequency setting. The degree to which it attenuates frequencies adjacent to the center frequency is determined by the bandwidth setting. This filter is useful for removing 50 or 60 Hz hum from a recording (or harmonics thereof). In forensics “black-box” recordings, it is useful for reducing 400 Hz aircraft generator noise. It is also useful for decreasing any sound system acoustic resonance that may be found on some live recordings. It can be used to attenuate the heterodyning "whistle" which is sometimes heard on AM broadcast radio reception. Some audio restoration engineers also use this filter to remove some "Hiss" from old 78 RPM recordings. For this application, the filter's center frequency is set somewhere in the 8 to 12 kHz range, with a bandwidth of 0.25 Octave or less. Experimentation is the only way to determine its effectiveness in minimizing "Hiss" from your particular source material. Also, it is important to note that this method is not the most effective for "Hiss" removal. Instead, consider using either the Continuous Noise Filter, Dynamic Noise Filter or the EZ Clean Filter.



The Notch Filter

The following is a summary of the control parameters and range of adjustment provided for the Notch Filter:

- **Center Frequency**: 5 - 18,999 Hz
- **Bandwidth**: 0.01 Octaves to 1.99 Octaves
- **Preview Mode Button**: On/Off (The slider controls can be adjusted "live" when the preview mode is on.)

This filter incorporates the "Slot Filter" feature. Essentially, a slot filter produces a variable band-pass response to its applied signal. This allows you to hear or keep only the residual component of the original signal. This feature is useful for "tuning" the notch filter to the maximum level of noise, so that when you actually run the notch filter with the "slot filter" feature turned off, the noise left behind will be minimized. The slot filter function is also useful in Forensics applications, wherein one is interested in isolating a very particular sound that exists in a very specific and narrow frequency band. If multiple slots are required, use the Multi-Filter with multiple notch/slot filters in the chain, or consider using the Harmonic Reject filter in "Keep Residue" mode, provided the slots, which are required, are harmonically related.

Note: The frequency range of adjustment up to 18,999 Hz is only effective when utilizing a 44.1 kHz sampling rate. At a 22.05 kHz sampling rate, the maximum effective frequency setting will be 10 kHz, and at an 11.025 kHz sampling rate, this value will drop to 5 kHz.

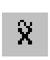
Notch Filter Procedure (Tutorial)

1. Identify the frequency that you desire to reject from the .wav file recording. The determination of the frequency of interest can be simplified by using the Band-pass filter as an Audible Spectrum Analyzer.
2. Highlight the portion of your .wav file on which you desire to apply the Notch filter. (You may choose to highlight the entire file or any portion thereof.)
3. Click on "Notch."
4. Choose the Center Frequency based on the "problem" frequency that you have observed and desire to reject. The Center Frequency range is from 5 Hz. to 18.999 kHz. (The highest useful frequency is around 15 kHz.) If you desire the finest degree of frequency resolution possible, use direct numeric entry rather than the use of the slider controls.
5. Choose the Bandwidth that you find to be the most effective in eliminating the desired frequency. The bandwidth control is calibrated in Octaves, and has a range from 0.01 Octaves to 1.99 Octaves. You should choose the smallest possible bandwidth that still accomplishes the job of rejecting the troublesome frequency. Otherwise, you will start eliminating useful information from your recording. Generally, useful ranges for Bandwidth will be in the 0.5 Octave to 0.1 Octave range.

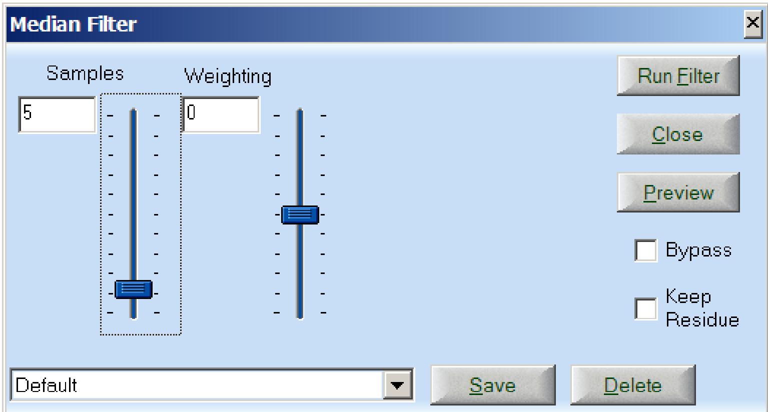
6. If you desire to hear the results of your filter settings before creating a new "Destination File, click on "preview."
7. After a short delay, you will hear the effect of the settings that you have chosen. (The system may seem to stutter if your computer is too slow to keep up with the algorithm in real time. However, this repeating pattern will not be present in the final Destination processing of the filter.)
8. As the filter is running in either preview mode or normal mode (Destination File Mode), you will see a dialog box that indicates the "% Done" of the filter algorithm on the selected portion of the Source .wav file. Also, at the top of the Dialog box you will see indicated the "Total Samples to Process:"
9. Keep adjusting the Frequency and Slope parameters, and testing the various settings using the "Preview" mode until you are satisfied with the results.
10. When you are satisfied with a group of settings, you will no longer need to use Preview mode.
11. Click on Run, and the filter will process your Source .wav file through the filter algorithm, and create a Destination .wav file containing the output of the filter.
12. When this process is complete, you will see the Destination File become highlighted in yellow, at the same time that the Source File becomes unselected.
13. Click on "Close."

<u>Note:</u> If only a small section(s) of your .wav file is in need of Notch filtering, as is often the case when acoustic feedback is encountered on a live recording, you can use "sync mode" and just filter the sector which contains the noise which you’re attempting to reduce. "Sync mode" can be selected under the "View Menu" and is associated with the Diamond Cut Classic Edit mode.

<u>Median Filter</u>



The Median Filter can be used to substantially reduce "crackle" (small impulse noise) from a recording. Use a sample setting of 3 to 7 for this application as your starting point range. A “weighting” control is also provided, which affects the “timbre” of the processed sound.



The Median Filter

There is no analog circuit equivalent to the Median Filter. This filter defines a window of samples, and for that window, determines which sample is the median value within the grouping (median meaning middle amplitude value). That value is the one that is passed along to the destination file, and then the window moves over 1 sample and re-evaluates the median, again passing the new median value to the destination file. This filter is useful for improving the intelligibility of severely distorted signals and it is also useful for pulling signals out of a very poor signal-to-noise ratio situation (pulling signals out of the mud). It is somewhat similar in sound performance to a high-order Low-pass filter. (The median value of a string of sorted numbers is the one in the middle of the string. In other words, if you have seven sorted numbers by amplitude, the fourth number will be the median value.) The DCArt10/DC Forensics10 Median Filter allows you to choose the number of samples over which the median value is determined by the algorithm. The range is from 3 to 20 samples. The higher that you set this value, the greater will be the attenuation of high frequency signals and/or noise. Also, the higher the samples setting, the longer will be the processing time required for the filter. The most useful settings will generally be found to be from 3 to 7 samples. Outside of that range, you may hear a significant degradation of the top end of your recording, depending on its bandwidth. Even at 7 samples, you will notice some "fuzziness" inter-modulating with the upper end of the spectrum on some recordings. So always start with 3 to 5 samples when using the Median filter, and choose the smallest value that produces an effective de-crackling result. It is also important to note that the higher the number of samples selected, the longer it will take your computer to calculate the Median values to process your .wav file. At some high settings, your system may not be able to process the file in "real-time" due to the heavy demands that it places on your CPU.

Weighting Function

A “Weighting” function is provided with this filter that shifts the position on the number line as to which value will be chosen as the “Median.” This feature is particularly useful in forensics audio applications wherein extremely inarticulate speech needs clarification. For example, the absence of consonant or sibilant sounds can render a recording undecipherable. Recording bandwidth limitations can severely distort or eliminate the “hissing-consonants” which render speech understandable - - - a fact that has been known since the earliest ventures by Thomas Edison into the field of recorded sound. With the use of the Median filter in conjunction with the weighting control, this problem can often be corrected. The weighting control will essentially affect the “timbre” of the processed sound. Trial and error will determine the best combination of “Samples” and “Weighting” to cure a particular forensics sound problem.

The following is a summary of the control parameters and range of adjustment provided for the Median Filter:

- **Samples**: 3 - 20

- **Preview Mode Button**: On / Off (The slider control can be adjusted "live" when the preview mode is on.)

- **Weighting**: +/- 100 (%) from the neutral value of 0. (0 provides no weighting to the Median value calculated by the filter.)

- **Bypass Button:** Allows you to compare between the filtered and the raw signal.

- **Keep Residue:** Allows you to hear what the Filter is removing from the raw signal.

Note 1: De-Crackling will generally be best accomplished with a "Samples" setting around 3. Intelligibility improvement of extremely distorted or garbled voice recordings will generally be accomplished with "Samples" settings anywhere between 5 through 17. You will have to experiment to determine the best settings for solving a particular problem.

Note 2: Best results on Forensics files will be had when they are first converted to a 44.1 kHz sample rate.

Median Filter Operating Procedure (Tutorial)

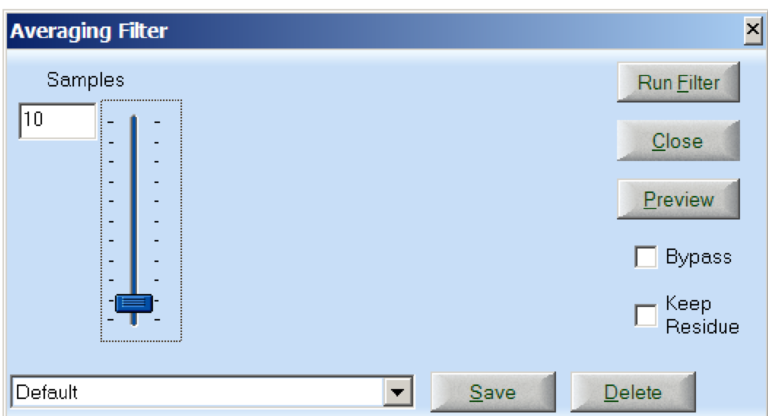
1. Highlight the portion of your .wav file on which you desire to apply the Median Filter.
2. Click on the "Median Filter" (Filter Menu).
3. Choose the number of samples over which you desire the median calculation to be performed. The higher the number of samples selected, the greater will be the attenuation of the higher frequency portion of the audio spectrum. You can choose any integer value from 3 to 20 samples. (The most useful values will be found in the 3 to 7 samples range.) The higher the chosen number of samples, the longer will be the process time requirement for the algorithm. Changes in value are accomplished utilizing the slider control.
4. If you desire to hear the results of your filter settings before creating a new "Destination" file, click on "Preview".
5. You will hear the effect of the calculation of the median value over the chosen number of "samples".
6. As the filter is running in either preview mode or normal mode (Destination File Mode), you will see a dialog box that indicates the "% Done" of the filter algorithm on the selected portion of the Source .wav file. Also, at the top of the Dialog box you will see indicated the "Total Samples to Process:"
7. Keep adjusting the number of samples until you achieve your desired effect.
8. When you are satisfied with a setting, click on "Run", and the filter will process your source .wav file through the filter algorithm, and create a Destination .wav file containing the output of the filter.

9. When this process is complete, you will see the Destination File become highlighted in Yellow, at the same time that the Source File becomes unselected.
10. Click on "Close".

Note: The “Weighting” control is used to affect the Timbre of the resultant sound of the Median filter.

Averaging Filter

This filter sounds similar to that of a Low-pass filter, although it is somewhat more effective than a Low-pass filter in reducing not only "Hiss" but also "Crackle" from a sound source. It is most effective on limited bandwidth sources such as old acoustic recordings made before 1925. This filter is also useful for improving the intelligibility of highly garbled voice communications recordings.



The Averaging Filter

This is another filter that has no analog equivalent. Its interface to the operator is similar to the Median Filter, with the difference being that instead of calculating the median value of a sample window to pass into the Destination workspace, the average value of a group of samples is passed through. You select the number of samples on which the average value is calculated with a slider control in its dialog box. The greater the number of samples, the higher the degree of smoothing

effect on the waveform. The higher that you set the degree of smoothing, the greater will be the loss in the higher end of the audio frequency spectrum.
The following is a summary of the control parameters and the range of adjustment provided for the Average Filter:

- **Samples:** 2 - 100

- **Preview Mode Button:** On / Off (The slider control can be adjusted "live" when the preview mode is being used.)

- **Bypass Button:** Allows you to compare between the filtered and the raw signal.

- **Keep Residue:** Allows you to hear what the Filter is removing from the raw signal.

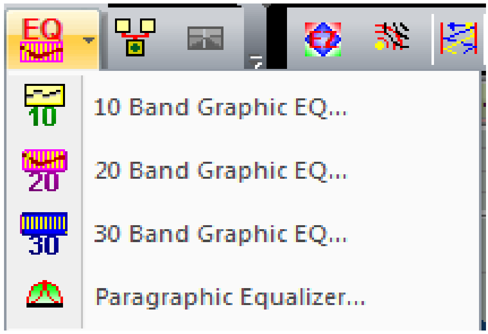
Average Filter Operating Procedure (Tutorial)

1. Highlight the portion of your .wav file on which you desire to apply the Average Filter. (You may choose to highlight the entire file or any portion thereof.)
2. Click on the “Filter Menu” with the left mouse button.
3. Click on "Averaging".
4. Choose the number of Samples over which you desire the moving average calculation to be performed. The higher the number of samples chosen, the greater will be the attenuation of the higher frequency portion of the audio spectrum. You can choose any integer value from 2 to 100 samples. The higher the number of samples selected, the longer will be the processing time requirement for the algorithm. This selection is accomplished utilizing the slider control.
5. If you desire to hear the results of your filter settings before creating a new "Destination" file, click on "preview."
6. You will hear the effect of the averaging over the chosen value of "samples".

7. As the filter is running in either preview mode or normal mode (Destination File Mode), you will see a dialog box that indicates the "**Percent Done**" of the filter algorithm on the selected portion of the Source .wav file. Also, at the top of the Dialog box you will see indicated the "**Total samples to process:**"
8. Keep adjusting the number of samples until you achieve your desired effect.
9. When you are satisfied with a setting, you will no longer use the “Preview” mode button.
10. Click on “Run”, and the filter will process your source .wav file through the filter algorithm, and create a Destination .wav file containing the output of the filter.
11. When this process is complete, you will see the Destination File become highlighted in Yellow, at the same time that the Source File becomes unselected.
12. Click on "Close".

Equalizers

The primary Diamond Cut equalizers are found within a sub-group called “EQ” under the Filter menu. The commercial version of the product includes 4 Equalizers while the Forensics version includes a total of 5. For example, the Forensics version includes a 10, 20 and 30 band graphic Equalizers and a paragraphic Equalizer (a parametric EQ that plots it’s frequency response). Additionally, a 3 band equalizer can be found within the Virtual Phono Preamplifier, which includes bass, mids and treble controls. Its bass and treble controls are of the shelving type (James-Baxandall type) while the mids are of the peaking (2nd order resonant) type. All of the Diamond Cut EQs use IIR techniques to achieve their result. Here is what the EQ menu looks like after the down arrow is clicked and expanded:



The EQ Menu in its expanded state

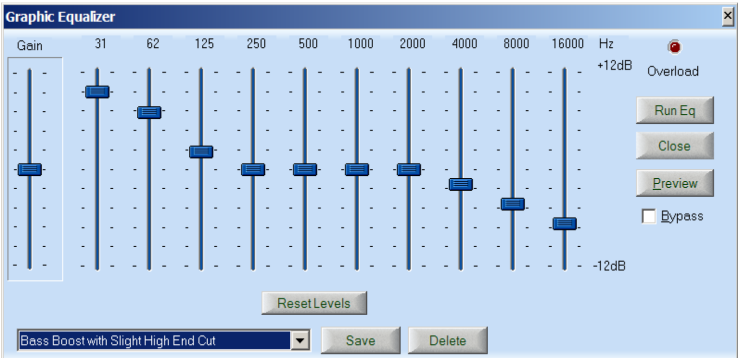
The 10 Band Graphic Equalizer

The 10 Band Graphic Equalizer is a familiar filter that acts like an expanded tone control. The audio spectrum is broken into 10 bands, each being one octave wide. Each band's gain (volume) can be independently adjusted to achieve the desired audio result. This filter is useful for tonal shaping of the finished audio product or to enhance the bass or treble of a recording. It is also useful for improving the intelligibility of recordings or "Bringing Out" a particular instrument or vocal.

The Graphical Equalizer is the digital equivalent (which is IIR based) to the analog Graphical Equalizer found in many sound systems. It uses 2nd order resonant (peaking) techniques to achieve its results. Its primary advantage is that it can be applied to a .wav file without having to resort to adding an analog step in your sound restoration process. This results in decreased noise and distortion on your final product. The equalizer has ten bands containing the following center frequencies:

31 Hz, 62 Hz, 125 Hz, 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz

8 kHz, 16 kHz



The 10 Band Graphic Equalizer

The amplification and attenuation range for each band is + / - 12 dB. A "Reset Levels" feature is provided. Clicking on "Reset Levels" will return all of the Graphical Equalizer slider controls to their 0 dB position. Since the graphic equalizer can add gain to your signal, a latching overload indicator is provided. If, at any time during the processing of a file through the graphic equalizer, the output signal exceeds the dynamic range of the system, the indicator will turn red and latch until the filter is re-run. The top and bottom bands are also peaking type filters and not shelving types. If shelving filters are needed, please refer to the Paragraphic EQ which includes two bands that can be operated in that mode.

Note 1: The top equalizer band of 16,000 Hz is only effective when using a sample rate of 44.1 kHz or higher; it becomes ineffective at sampling rates of 22.05 kHz and 11.025 kHz. The 8,000 Hz band will also be rendered ineffective when using a sampling rate of only 11.025 kHz.

Note 2: The graphic equalizer controls can be adjusted "live" when the preview mode button is clicked.

Note 3: Since the graphic equalizer can actually increase the gain of the system, it is possible to produce clipping which will result in unpleasant distortion products to appear in the Destination Workspace. This usually occurs when excessively boosting the bass portion of the spectrum. The overload indicator will change from green to red, and latch in that condition if there has been an overload. The latch will be reset, following a re-run of the algorithm, provided that the overload condition has been cleared by reducing gain in one or more bands. A good remedy for this is via the overall “Gain” control; reduce its setting if clipping occurs.

Graphic Equalizer Operating Procedure (Tutorial)

1. Click on the Filter Menu.
2. Click on "EQ".
3. Click on “10 Band Graphic EQ”.
4. Using your Mouse, adjust the frequency band slider control(s) up or downwards as desired. This can be accomplished by directly pointing the cursor with the mouse and depressing the left mouse button to move the control.
5. When the Slider control for a particular band is in its center position, the band is neither being attenuated or amplified. Moving the slider upwards produces amplification of frequencies in the band up to 12 dB. Moving the slider downwards produces attenuation of frequencies in the band of up to 12 dB.

The 20 Band Graphic Equalizer

The 20 Band Graphic Equalizer is an IIR based extension of the 10 Band Graphic Equalizer. It exhibits twice the selectivity compared to the 10-band equalizer, since each band is calibrated with half its bandwidth. It is quite useful where greater selectivity is required in order to “tweak” your audio signal frequency distribution, but it is a little more difficult to use compared to the general purpose 10-band equalizer. Its amplification and attenuation range is +/- 12 dB as indicated by the numerical readouts located below each slider control. It has the following frequency band center frequency values:

22Hz, 31Hz, 42Hz, 63Hz, 88Hz, 125Hz, 177Hz, 250Hz, 350Hz, 500Hz, 710Hz, 1.0kHz, 1.4kHz, 2kHz, 2.8kHz, 4kHz, 5.6kHz, 8.0kHz, 11kHz, 16kHz

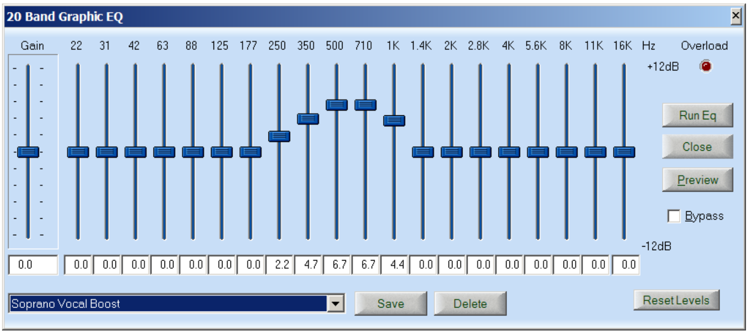


Figure 69 - The 20-Band Graphic EQ

In all other regards, the 20 Band Graphic Equalizer is operated in the same manner as the 10 Band Graphic Equalizer. *Please refer to the description of the Graphic Equalizer above for more details*.

Note: All bands above 45 % of the .wav files sampling frequency are disabled.

The 30 Band Graphic Equalizer

(Forensics Version Only)

30 Band Graphic IIR-Based EQ

The 30 Band 1/3rd Octave Graphic Equalizer is an IIR based extension of the 10 and 20 band Graphic Equalizers. It exhibits three times the selectivity compared to the 10-band equalizer, since each band is calibrated with 1/3rd of its bandwidth. This is very useful in Forensics applications wherein signals in a relative narrow portion of the audio spectrum need to be amplified or attenuated. Its amplification and attenuation range is +/- 12 dB as indicated by the numerical readouts located below each slider control. If this filter does not have sufficient selectivity for your application, then use the FFT based Spectral Filter, which has a choice of from 128 to as high as 32,000 bands. The 30 Band Graphic Equalizer has the following frequency band center frequency values:

25 Hz, 31 Hz, 40 Hz, 50 Hz, 62 Hz, 80 Hz, 100 Hz, 125 Hz, 160 Hz,
200 Hz, 250 Hz, 320 Hz, 400 Hz, 500 Hz, 640 Hz, 800 Hz,1 kHz,
1.3 kHz, 1.6 kHz, 2 kHz, 2.5 kHz, 3.1 kHz, 4 kHz, 5 kHz,
6.2 kHz, 8 kHz, 10 kHz, 13 kHz, 16 kHz, 20 kHz

Note: All bands above 45 % of the .wav files sampling frequency are disabled.

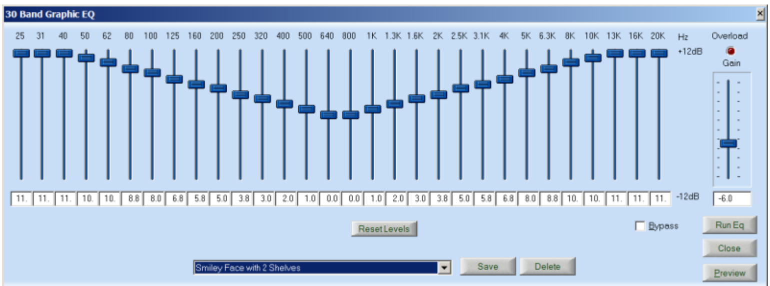
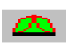
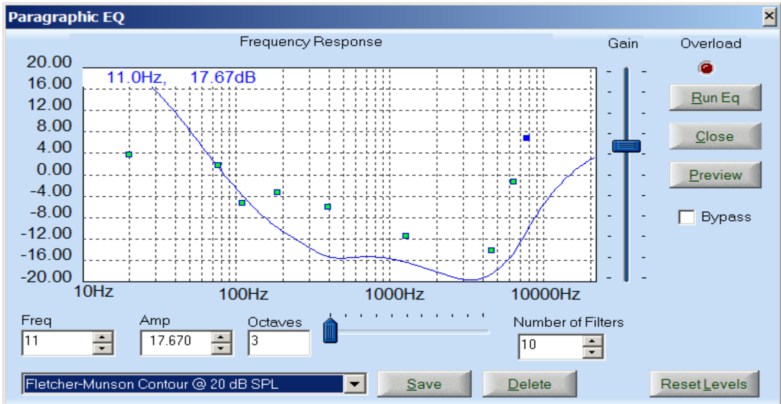


Figure 70 - The 30-Band Graphic EQ

Paragraphic Equalizer



This unique equalizer combines the flexibility of a parametric equalizer with the ease of use associated with a 10 band graphic equalizer. The actual frequency response of the filter is graphically controlled and/or displayed. Virtually any frequency domain transfer function that you can dream of can be created with this filter. Many presets are provided to facilitate many unusual equalization situations.



The Paragraphic Equalizer

The Paragraphic Equalizer is a unique form of a parametric equalizer employing IIR techniques. It combines the graphical representation of its transfer function (frequency response curve) with the versatility of a parametric equalizer. Additionally, it provides you with up to 10 bands of equalization. It differs from the graphic equalizer in that three parameters are adjustable for each band:

- **Frequency:** (Hertz - 10 to 20 kHz)
- **Amplitude:** (Attenuation or Amplification of up to +/- 20 dB)
- **Octaves:** (Q or bandwidth - 0.05 to 3.0)

The following additional controls and displays are provided:

- **Number of filters:** (1 to 10)
- **Output Gain:** (+ / - 20 dB)
- **Reset Levels:** (Resets the Paragraphic EQ to zero dB and factory default frequencies)
- **Overload indicator:** (Illuminates when full-scale output {clipping} occurs.)

The Parametric Equalizer displays its frequency domain transfer characteristic graphically. Therefore, it is referred to as the DCArt10 / DCForensics10 "Paragraphic" equalizer. It can be modified using your mouse by dragging the inflection point dots, and modifying the bandwidth using the octave control. You simply draw the shape of the response, which you desire, and the algorithm adjusts the parameters to match the response curve. Each band is represented by a single square "dot" on the graph. The "active" dot (the one being adjusted) will be the larger one displayed. That is the dot for which the parameters are being displayed numerically on the control panel. You can use the mouse to drag any of the dots, which actually represents a frequency inflection point, wherever you wish. If you want to sharpen or widen the response of any inflection point, use the “Octaves” control to achieve the desired curve for a highlighted band dot. It is often very useful to use the spectrum analyzer, found under the View menu, in conjunction with the Paragraphic Equalizer. You will then be able to see the exact effect that you are imposing on the .wav file signal.

Under the factory presets listing, you will find a number of useful audio restoration functions, including the RIAA curves. Also, various inverse RIAA curves with a variety of turnover frequencies are available. These features enable you to use a standard RIAA pre-amplifier to transfer acoustical and electrically recorded 78 RPM records to your hard drive, and re-compensate at another point in time, without having to purchase specialized hardware. Also, of interest is the family of Fletcher-Munson Equal Loudness Contours at different sound pressure levels. These can be used to compensate for the response of the human ear depending on the loudness level that you expect a particular audio piece to be auditioned.

Shelving Filters: The top band (right-hand side of the graph) always operates as a shelving function (a pole-zero pair). The rest of the bands are second order resonant filter systems (often referred to as peaking filters), except the bottom band (left-hand side of the graph) which can be set either as a shelving transfer function or as a peaking routine. It defaults to a peaking filter, but can be converted to a shelving function by checking the shelving checkbox (“Low Freq Shelf”). So, you can create a system in which you have a low frequency and high frequency shelf with 8 peaking filter bands in-between if you so choose. If you only want to work with two shelving filters, just check the “Low Freq Shelf” checkbox and then reduce the number of filters to 2. One of those two will be the upper frequency shelf and the other one will be the lower frequency shelf. You can then adjust both to achieve the desired upper and lower frequency shelving response. The Amplitude and Frequency controls apply to the shelving function(s), but not the Octaves control. “Octaves” only apply to the various filter bands when they are operating in non-shelving mode(s).

Note 1:
As with all of the DCArt10/DC Forensics10 filters, sample theorem dictates useful bandwidth for the algorithms. The Paragraphic Equalizer will only have a useful bandwidth up to about 10 kHz with a 22.05 kHz sample rate, and about 5 kHz at 11.025 kHz. It will be fully useful with sample rates of 44.1 kHz and above.

Note 2:
Many of the Paragraphic Equalizer presets such as the RIAA, Reverse RIAA and NAB curves are defined over the entire audio spectrum consisting of at least 20 Hz to 20 kHz band spread values. Therefore, the use of any sampling rates less than 40 kHz will invalidate the accuracy of these curves. We recommend using only 44.1 kHz or higher in order to properly realize these equalization curves.

Note 3:
Random white noise can be converted to pink noise by feeding it through the appropriate factory preset(s) within the Paragraphic Equalizer.

Virtual Phono Preamplifier (VPP)

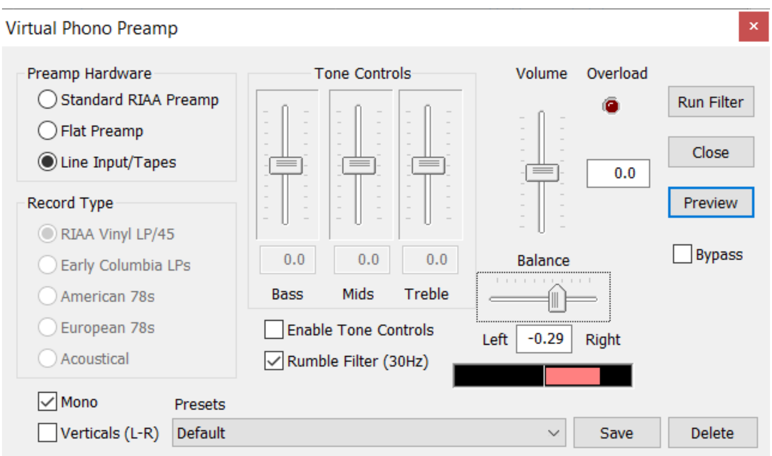
Phono Preamplifier Simulator including Common EQ Curves & Tone Controls

The Diamond Cut **V**irtual **P**hono **P**reamplifier™ (VPP™ or sometimes called the VPA) is an IIR based digital simulation of an analog based hardware pre-amplifier having magnetic phono and line level input circuits. It uses “closed form” mathematical representations of the various phonographic EQ curves resulting in near ideal amplitude and phase responses per the various phonographic EQ standards. For example, the RIAA and Reverse RIAA curve deviations from the ideal are not measureable when using a 48 kHz sample rate or higher (having less than a small fraction of a dB variance over the entire audio spectrum from the ideal curves). In addition to the standard features found on most analog circuit based preamplifiers, the VPP includes the ability to apply various phono equalization and re-equalization curves with near perfect accuracy. It can work in conjunction with three different types of analog input front-end (hardware) sources including standard RIAA preamplifiers, flat phono preamplifiers and line level inputs.

Besides creating the standard RIAA curve from a flat phono preamplifier front-end, the VPP's versatility allows you to use a standard RIAA hardware pre-amplifier front-end to derive from its signal the most common 78 turnover curves. It accomplishes this by reversing the RIAA curve imparted by the RIAA preamplifier front-end and then re-applying the desired 78 turnover characteristic curve. It can also perform the mentioned function when utilizing a flat phono pre-amp front end*. Supported curves also include the Columbia LP EQ curve found on many vintage monophonic LPs as well as two common 78 Turnover curves. Decoding of acoustically mastered recordings is also provided using an internal reverse RIAA curve so that you can use an RIAA preamp to transfer early 78s. Additionally, a 30 Hz, third order (18 dB / Octave) Rumble filter is provided and also a mono selector switch to sum the two channels together (L+R) when transferring vintage monophonic (lateral cut) LPs and 78s in conjunction with a stereo phono cartridge.

The VPP also includes a set of tone controls consisting of a 3 Band Equalizer having 2 shelving type controls (Bass and Treble) and 1 resonant peaking type (Mids). These controls have a wide adjustment range so that you can tune the output of the system to your own taste if desired. The shelving controls response shapes are complimentary with the fixed phono EQ curves in the VPP, and are very useful for fine tuning the phono EQ curves on your system if desired. And of course, the VPP also includes Volume and Balance controls along with a Balance Meter.

Some soundcards produce output signal levels up to roughly 1.5 Volts RMS. And, the input sensitivity of most audio power amplifiers and powered loudspeakers require only around 1 Volt RMS to produce full output power. As a result, you can create a very simple and clean sound system consisting of nothing more than your computer coupled directly into a power amplifier and speakers or powered loudspeakers. This allows you to bypass some of the usual analog hardware preamplifiers or mixers generally found in sound-labs. This reduces the overall distortion introduced into the signal path, because the analog hardware count can be reduced which is often desirable and/or convenient. So a complete playback sound system can consist of nothing more than your computer coupled to a set of powered loudspeakers with the VPP providing a comprehensive set of controls. If the VPP is used within the Multi-Filter context in "Live" mode, you can set up a complete real-time sound-system with your computer, a set of powered speakers and an input source. Furthermore, you can also cascade other filters (like the various Impulse Noise Filters, Continuous Noise Filters or any of the EZ Filters) into the signal pathway after the VPP to remove impulsive and continuous noises from the source and do it in real-time. Of course, the entire plethora of Diamond Cut filters and effects can also be added to the chain making your real-time audio system extremely versatile. It is important to note that if the VPP is used in conjunction with multiple filters and/or effects, the VPP should always be configured as the first item in the signal chain.



The Diamond Cut Virtual Preamplifier (VPP)

Here are the VPP controls defining their functionality and range of adjustability:

Volume Control
On the top right side of the VPP, you will find a Volume control, which performs the obvious function. It has a range of adjustment from 0 to +20 dB or 0 to –20 dB and defaults to a unity gain mode in it's central position (x 1.00, Gain = 0.00 dB).

Balance Control
Directly below the Volume control is the VPP channel Balance control. Normally, this would be set to the center position, which is designated as 0.00. Moving the control to the right moves the stereo soundstage towards the right hand side of your listening space, while moving the control towards the left hand side has the opposite effect. The range of adjustability for this control is 0 to –1 (when moved fully to the right) and 0 to +1 (when moved fully to the left). At either extreme setting of the balance control, the opposite channel is fully attenuated (producing no output). The degree of balance can be seen on the balance meter (balance galvanometer). This meter is very useful to fine tune the vector derived from a stereo cartridge when transferring monophonic records such as old 78 lateral cut records, or 80 rpm verticals. It is also useful for finding the optimum balance for early monophonic LPs. First, you must select the mono checkbox. If it is a lateral cut monophonic record (like an LP or old 78)* then by adjusting the balance control for zero deflection (null) as shown on the balance meter, you will be able to achieve the optimum signal transfer of these records to your computer. This "sweet-spot" is known as the "null-point" as indicated by the balance meter deflection (tune for a minimum deflection of the meter - - - the center position on the meter scale is the "null"). To fine tune the balance control, point the mouse at the slider of the balance control and click on it and then use the left and right arrows on your keyboard to make small changes. After you find that null or sweet spot, you can then run the VPP filter. If you are working with a vertical cut record† (like an early cylinder or an Edison Diamond Disc, or a Pathe' Disc), you will need to assure that both the mono and the Verticals (L-R) checkboxes are checked. After the best signal is obtained via the balance control and the balance meter nulling, run the filter. Next proceed with the remaining steps associated with your restoration processes.

Note 1: On stereophonic LPs and 45s, the balance meter will only show the stereo separation signal of the source recording. You may find the balance control and the balance meter somewhat useful on these types of records for obtaining the best stereo balance of your transfer. This will allow you to obtain the best center sound-stage position for the signal. This can be accomplished by tuning the balance control for equal deflection of the balance meter in both directions from the center (null) position of the pointer.

***Note 2:** A lateral cut monophonic test file can be found under the edit menu / demo files called "Dyna-Bass Demo" for you to experiment with.

†**Note 3:** A vertical cut monophonic test file can be found under the edit menu / demo files called "Preamp Vertical Cut Demo" for you to experiment with.

Tone Controls

The VPP has three tone controls including Bass, Mids and Treble. The tone controls are enabled by placing a checkbox in the "Enable Tone Controls" selector when you desire to use them. To defeat them, simply uncheck the same selector box. The Tone Controls have the following characteristics:

Bass: Shelving type with its corner frequency set to 175 Hz and adjustability of +/- 15 dB (15 dB of Boost or Cut).

Mids: Peaking (band-pass resonant) type with its center frequency set to 900 Hz, 3 Octaves bandwidth. Adjustability is +/- 15 dB (15 dB of Boost or Cut).

Treble: Shelving type with its corner frequency set to 3,000 Hz and adjustability of +/- 15 dB (15 dB of Boost or Cut).

Note concerning all controls: Fine adjustments can be made using the arrow keys on your keyboard after you have used the mouse to select a particular control.

Rumble Filter

The VPP rumble filter is of the 3rd order Butterworth variety having a corner frequency of 30 Hz. Thus, its attenuation at 15 Hz is 18 dB. It can be effective for attenuating rumble sounds due to turntable (or even cutting lathe induced) record rumble, which is a low frequency random noise that often "pumps" sub-woofers. Rumble is most noticeable by the random movement of your sub-woofer's speaker driver cone which wastes power and introduces inter-modulation distortion into the signal if not attenuated. It can also introduce a very annoying "wind" sound coming from a sub-woofer system if there is too much "Rumble" present. You will know if you need this filter simply by experimenting with it. To enable the "Rumble Filter", simply tick off the checkbox with your mouse. If you do not have a sub-woofer, you may not notice turntable rumble even though it is there.

Preamp Hardware (Selector Box)

The top left corner of the VPP provides you with a set of three input selections from which to choose. Your options are:

<u>Standard RIAA Preamp:</u> If you are using a standard magnetic RIAA phono preamplifier, then you need to check this box.

<u>Flat Preamp:</u> If you are using a flat preamplifier (such as the CTP - XXXX family of magnetic phono preamplifiers) you will need to check this box.

<u>Line Input / Tape:</u> If you are feeding your soundcard directly from a tape deck or other high-level signal source, you need to check this box. This selection will bypass the phono equalization and re-equalization systems found in the "Record Type" selector box, feeding the signal directly into the Tone Controls and / or Volume and Balance sub-system.

Please note that any of the above input hardware options need to be connected to the line input of your soundcard; do not connect the output of any of these sources to the soundcards microphone input. Line level outputs will overload the high-gain input of the microphone amplifier of your soundcard creating clipping distortion which results in a very unpleasant sound.

Record Type (Selector Box)
The lower left corner of the VPP provides a group of selections based on the type of record that you are trying to transfer to your computer. This selection, in conjunction with the "Preamp Hardware" goes through a logical truth table in order to determine the proper algorithms to run to produce the correct phono EQ curve for the particular situation that you are working with. These EQ's and re-equalization curves are highly accurate because they are calculated in a closed form mathematical technique. The most accurate transfer results will occur when you use a flat preamplifier as your systems input front end. Analog circuit parameters associated with resistors and capacitors in physical RIAA preamps render them less precise. These inaccuracies are reflected not only in the RIAA curve, but also in any of the re-equalization curves that the VPP can provide.

You have the following record options from which to choose:

<u>RIAA Vinyl LP / 45</u>
This is intended to be used on all 45 RPM records and almost all LPs which were mastered post 1955 and some that were mastered prior to that date as well. The RIAA curve was originally called "The New Orthophonic Recording Characteristic" by RCA Victor which was proposed as an EQ standard in 1953 and later adopted by the RIAA. The VPP RIAA curve is defined as having the following breakpoint frequencies: (F {in Hz} = 1 /2 π (R x C) where R x C = Time Constant)

50 Hz Pullout (3180 uSec Time Constant)
500 Hz Turnover (318 uSec Time Constant)
2120 Hz Rolloff (75 uSec Time Constant)

<u>Early Columbia LPs</u>
This EQ curve is intended for many early LPs mastered during the period between 1948 through 1955. This curve was established as a standard for early Columbia LPs. The VPP Early Columbia LP curve is defined as having the following breakpoint frequencies:

30 Hz Pullout (5310 uSec Time Constant)
300 Hz Turnover (531 uSec Time Constant)
1600 Hz Rolloff (99.4 uSec Time Constant)

American 78s
This EQ curve applies a turnover curve only (and the appropriate pullout) for use with 78 records having a 500 Hz characteristic. It does not implement any Rolloff, since 78s did not encode in that manner. Many American 78s were recorded with this curve. For more details regarding 78 Turnover frequencies by record brand, please refer to the "Turnover Frequency" section found in the Appendix of this user's guide.

European 78s
This EQ curve applies a turnover curve only (and the appropriate pullout) for use with 78 records having a 250 Hz characteristic. It does not implement any Rolloff, since 78s did not encode in that manner. Many European 78s were recorded with this curve. For more details regarding 78 Turnover frequencies by record brand, please refer to the "Turnover Frequency" section found in the Appendix of this user's guide.

Acoustical
This EQ curve reverses the curve introduced by an RIAA front-end preamplifier and is passive for signals sourced from a flat preamplifier. When used in conjunction with a conventional RIAA hardware preamplifier, this setting would often be referred to as "Reverse RIAA Mode". It is to be used on acoustically mastered material. Most recordings mastered before 1925 require this setting, including laterally cut 78s, Edison Diamond Discs, Pathé, and cylinder recordings.

Mono Checkbox
Sometimes it is desirable to convert laterally cut recordings to monophonic (Mono L+R) to reduce surface noise pickup produced by using a stereo phono cartridge. Also, you may prefer to convert a stereo recording to monophonic which can be accomplished with this feature. Alternatively, you may be working with vertical cut records and need to decode the vertical signal from a stereo transfer. To accomplish that task, you can use the Mono feature coupled with the "Verticals (L-R)" checkbox*. Always use the balance control in conjunction with the balance meter first to achieve the optimal signal balance. However, it is often better to de-click your recordings first before converting your file to Mono. Experiment to see if that is the

case with the file you are working with. Nonetheless, this feature is provided in the VPP as a convenience which is especially useful when using the system as a real-time phonograph playback system. It is important to note that Hill and Dale (vertically cut) records will not play correctly unless the Verticals (L-R) mode is checked. Alternatively, to play Hill and Dale records, please refer to the Diamond Cut File Conversion Filter (Mono L-R) to properly decode these recordings. So, when you are dealing with vertically cut records, be sure to leave the VPP Mono Checkbox turned to the proper setting so that the proper decoding can be applied. This is especially important when using complex filters further down the chain of events or further down the sequence of Multi-Filter filters.

*Note: A vertical cut demo file is provided called “Preamp Vertical Cut Demo” which is described at the end of this section of the documentation.

Overload
Since the VPP volume control and / or its tone controls have the ability to add gain to the system, overload (sometimes known as clipping) can occur. The red overload “LED” located in the upper right hand corner of the VPP will flash when this condition occurs. If it does, lower the offending control until the Overload indicator no longer illuminates.

Presets
As with most of the Diamond Cut filters and routines, you can store your favorite settings by using the preset feature in conjunction with the “Save” button. The system will prompt you to name the state of the VPP after clicking on “Save”. To recall a preset, scroll down and click on the desired setting. To delete a particular preset, highlight it and then hit the “Delete” button.

Factory Presets
You will see that there are around 60 factory presets provided in the VPP. Some presets are EQ Curves specified by their industry standard EQ Curve designation. For example, you can find the AES, NAB and FFRR LP EQ curves under those acronyms. You will also find EQ Curves that are specified by LP Label brand-name. Presets that are approximations have "Approximate" in the front of the name and finally, presets that are mathematically exact (closed form) have

"Exact" in front of the name. These various LP preset curves apply to records recorded prior to the spring of 1954. Almost all LP labels switched over to the RIAA standard curve after that date. For LP records recorded later than the spring of 1954, simply use the RIAA curve found in the VPP. A complete set of VPP presets can be found in the Preset Listing section of this users guide.

VPP Application Example #1
You have an RIAA front-end preamplifier which you are using to play or transfer a European 78 RPM record. You want to decode it properly. To do so, set up the VPP in the following manner:

Preamp Hardware: Check the “Standard RIAA Preamp” setting.
Record Type: Check the “European 78s” setting.

Optionally, you can also tick off the Mono checkbox.

VPP Application Example #2
You are using a Flat preamplifier front-end to play Vinyl LP Stereophonic Records. To properly decode them, set up the VPP in the following manner:

Preamp Hardware: Check the “Flat Preamp” setting.

Record Type: Check the “RIAA Vinyl LP/45” setting.

VPP Application Example #3
You are using an RIAA front-end preamplifier to play early Columbia Vinyl LPs and you desire to decode them properly. To do so, set up the VPP in the following manner:

Preamp Hardware: Check the “Standard RIAA Preamp” setting.

Record Type: Check the “Early Columbia LPs” setting.

Optionally, you can also tick off the Mono checkbox.

VPP Application Example #4

You are using an RIAA front-end preamplifier to play Acoustical 78s. To properly decode them, set up the VPP in the following manner:

Preamp Hardware: Check the “Standard RIAA Preamp” setting.

Record Type: Check the “Acoustical” setting.

Optionally, you can also tick off the Mono checkbox.

VPP Application Example #5

You are using a Flat Preamplifier front-end to play American 78s having a 500 Hz turnover curve. To properly decode them, set up the VPP in the following manner:

Preamp Hardware: Check the “Flat Preamp” setting.

Record Type: Check the “American 78s” setting.

Optionally, you can also tick off the Mono checkbox.

Important Note: Never choose settings for the wrong type of Preamplifier hardware being used because the balance of the frequency spectrum of energy distribution will be extremely unbalanced and / or distorted.

VPP Application Example #6

You are using a Flat Preamplifier front-end and you desire to play some LPs in real time through your computer sound system. You do not desire to transfer them to your hard drive, but simply to listen to them. To play them, do the following:

Bring up the Virtual Preamplifier in the Multi-Filter.

Preamp Hardware: Check the “Flat Preamp” setting.

Record Type: Check the “RIAA Vinyl LP/45” setting.

To play the LP in real time, click on “Live Preview” in the Multi-Filter.

Adjust the tone controls, volume and balance for the most pleasing sound.

Note: If you want to remove noise in real time while playing your LP, drag the appropriate noise reduction filters into the Multi-Filter path after (to the right side) of the VPP and adjust them appropriately.

VPP Application Example #7
You are performing a transfer using either fractional speed mastering or high speed dubbing techniques on an electrical recording and you want to decode its EQ properly. Your front-end preamplifier is of the flat variety*. To properly re-create the appropriate EQ, you will need to do the following:

Transfer the recording and then correct the speed using the Change Speed Effect. After the speed has been corrected, run the file through the VPP using the following settings:

Preamp Hardware: Flat Preamp.

Record Type: Use the setting which best fits the description of the transferred recording.

*Note: The CTP Series of Flat Phono Preamps can be purchased from Diamond Cut Productions, Inc.

VPP Application Example #8
You are performing a 78 RPM transfer using a magnetic RIAA pre-amplifier and you are also using fractional speed mastering techniques and you want to decode its EQ properly. To re-create the appropriate playback EQ, you will need to perform the following steps in this exact sequence:

Transfer the recording to your computer and then run the file through the VPP with it set for the following:

Preamp Hardware: Standard RIAA.

Record Type: Acoustical (even though the record may have been of the electrical variety; this step reverses the RIAA curve that was imparted onto the signal by your RIAA preamplifier).

Next, correct the speed of the transfer using the Change Speed Effect. Then, re-set up the VPP as follows and then "Run" the Filter:

Preamp Hardware: Flat Preamp.

Record Type: Use the setting which best fits the description of the transferred 78 RPM recording.

VPP Application Example #9
You are using the Virtual Phono Preamp to cut vinyl masters (vinyl has come back in vogue in some circles lately). To properly encode the master with the RIAA curve, set up the VPP in the following manner:

Preamp Hardware: Check the "Standard RIAA Preamp" setting.

Record Type: Check the "Acoustical" setting.

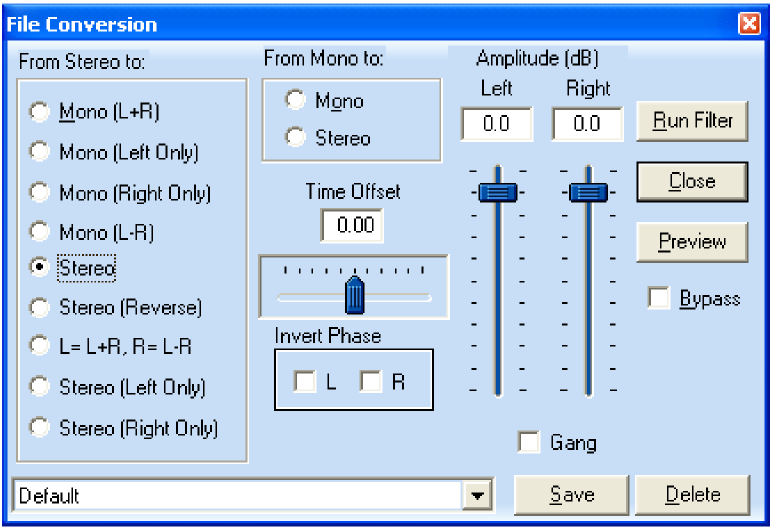
This process will allow you to take digitally recorded material and directly create the encoded RIAA master to a very high degree of accuracy (within a fraction of a dB of the theoretical curve). This master can then be used directly to drive the cutter lathe power amplifier to create your RIAA encoded vinyl master.

Preamp Vertical Cut Demo file:

This demo file is a stereo recording of a vertical cut recording made around 110 years ago by Pathe' (Skies of Normandy). If you try to play it in its raw state, the sound is overwhelmed by noise. However, if you bring up the Virtual Phono Preamp and check the Mono and Verticals (L-R) check-boxes and then preview the file, it will become substantially clearer. The reason for this is because this combination of Virtual Phono Preamp check-boxes allows the system to resolve (decode) the vertical component of the recording while rejecting the lateral (or horizontal) component of the signal leaving behind a relatively clear signal. After this step has been performed on a vertical cut recording, use the normal steps you would use to remove the remaining impulse and continuous noises.

File Conversion

The File Conversion filter is not really a filter at all, but a way to convert mono files to stereo and visa-versa. It can also be used to adjust the channel balance or reverse the channels of a stereo recording, or convert a mono source into a stereo file. It is useful in converting stereo recordings made out of phase (such as old vertically recorded acoustic discs) into a stereo or mono file that is compatible with modern systems. The Time Offset feature can be used to correct for group delay problems, correct azimuth errors from tape recordings, or to create special effects. The phase inversion function can be used to correct files that contain an inadvertent 180 degree phase shift of one channel with respect to the other (only one of the phase inversion checkboxes need to be checked to accomplish this correction on a relative basis). Phase inversion sometimes occurs when a balanced microphone (XLR or Cannon type connector) had been wired backwards. When paired with a properly wired microphone, the out of phase microphone can produce unusual phasing issues when both are mixed together. The phase inversion feature will correct this problem.



The File Conversion Window

The "File Conversion" Filter includes three slider controls. One set of two controls allows you to adjust the Amplitude (gain) levels for each channel. The third control, called "Time Offset", provides a means for azimuth correction, Forensics audio enhancement, and stereo simulation. A final important use of the file conversion filter is to simply copy parts of the source file over the destination file. This is one way to revert back to the original source file (undo) following a bad filter application.

This sometimes will be the first step that you will perform in a sound restoration project. This File Conversion Filter provides a number of file conversion options that can provide a certain degree of noise reduction in and of themselves. Gain adjustments can also be performed during the file conversion process. The following File Conversion options are available:

From Stereo to -

- **Mono (L + R):** This option adds the .wav file Left Channel Input to its Right Channel Input before feeding it into the Destination workspace. It produces a single channel output signal.

- **Mono (Left Only) & Mono (Right Only):** These options choose only one of the two .wav file inputs to be used in the file conversion-processing step and yields a single channel output.

- **Mono (L - R):** This option subtracts the Right .wav file signal from the Left signal before feeding it into the Destination workspace as a single channel signal.

- **Stereo:** This option maintains the .wav file in stereo through the file conversion-processing step. It can be used to adjust the gains of the two signals if they are incorrect while maintaining their independence.

- **Stereo Reverse:** This option will reverse the left and right channels during the .wav file conversion-processing step. It can also be used to adjust the gains of the two signals at the same time, if they are incorrect.

- **L = L+R R = L-R:** This option provides monophonic mix to the left channel, and provides the ambient signal from a stereo recording to the right channel.

- **Stereo (Left Only):** This option takes the Left Channel input and applies it to both (stereo) output channels and yields a dual channel output signal.

- **Stereo (Right Only):** This option takes the Right Channel input and applies it to both (stereo) output channels and yields a dual channel output signal.

From Mono to -

- **Mono:** This option merely provides a clone of the original .wav file.

- **Stereo:** This option converts a monophonic single-track file into two single-track monophonic files.

Here are more details on the various File conversion options and their specific application:

- **Mono (L + R)**

This is generally used to convert a lateral cut record (like a typical 78, or a monophonic LP) which is monophonic to start with, but which has been transferred to the hard drive with a stereo cartridge, and convert these two signals into one signal on which you will perform further processing. The advantage of this simple conversion is that some of the noise content of the record will cancel out during this process, in particular, low end rumble, and even some higher frequency surface noise. This process alone can provide up to 6 dB of signal-to-noise improvement (depending on the condition of your source) compared to the use of only one of the lateral groove walls (i.e. using the left only or the right only signal).

Important Note:

It is advisable to set both gain controls to - 6 dB to avoid overloading of the Destination channel during this mixing process, unless your recording is extremely under-recorded to start with. Minus 6 dB is the default value for the two gain settings in the Mono (L + R) File Conversion feature.

- **Mono (Left Only) & Mono (Right Only)**

Sometimes, 78-rpm laterals are worn unevenly due to years of improper tracking of the tone arm that played the particular record. Therefore, it is sometimes useful to compare the Left Only groove wall with the Right Only groove wall to hear if that is the case. If you hear a significant difference between one of the two groove walls, you should then compare the quieter of the two with the Mono (L + R) signal for comparison. Choose the quietest of the three possibilities for your Destination file.

- **Mono (L - R)**

This feature takes the algebraic difference between the left channel and the right channel audio signals and feeds it into the destination file. It has four significant applications:

1. If you have transferred vertical cut records such as cylinders or Edison Diamond Discs utilizing a stereophonic cartridge, and haven't previously extracted the vertical signal component from that signal, this feature will enable you to do so. Just as you would have done with the laterals, it is useful to listen to

the Left Only signal and compare it with the Right Only signal to make sure that no significant tracking damage has been done to the record over the years. Choose the quieter of the two for subsequent comparison to the Mono (L - R) signal. Generally you will find that the Mono (L - R) signal has the best signal- to-noise ratio for vertical (hill and dale) recordings. Pathé (groove width modulated 78s) recordings should also be converted to monophonic utilizing the Mono (L - R) feature.

2. This feature can also be used to compensate for gain imbalances between the left channel and the right channel of the analog equipment used to make the transfers into your computer system. When listening to a lateral monophonic recording in Mono (L - R), you can adjust the gain control sliders until you hear a maximization of the noise and garble on the recording, and a minimization of the useful information content of the recording. This will provide the best setting of the gain controls when you finally make the file transfer utilizing the Mono (A + B) file conversion feature.
3. It can be used to "cancel" a television or radio broadcast out of a surveillance recording. The surveillance recording would have had to be recorded in stereo, with the surveillance signal on one track and also with a "reference" track containing the broadcast. This technique will not completely cancel out the source radio or television source, but will attenuate it somewhat. Use the gain controls and preview to obtain the most effective degree of cancellation.
4. It can be used to attenuate some lead vocals on stereophonic recordings. A more comprehensive and more effective system for this purpose can be found under the Channel Blender presets listing.

- **Stereo**

This algorithm preserves a truly stereophonic Wave file in dual-channel format through the sound editing and sound restoration process. It allows you to adjust the channel levels to bring them more into loudness balance if they are imbalanced to begin with.

- **Stereo Reverse**

This algorithm transposes the left and right channels from the source file before it is transferred into the destination file. This algorithm also allows you to, while reversing the channels, adjust the channel levels to bring them more into balance if they are not balanced to begin with.

- **Time Offset Feature / Azimuth Correction** (0 mSec to +/- 20 mSec with 10 uSec resolution)**:**

The file conversion routine includes a "Time Offset" feature. It is mounted horizontally on the file conversion control panel. For normal file conversion operations, this control MUST be set to zero. The time-offset algorithm provides you with the ability to retard or advance the timing between two stereo tracks. This will work with a stereo-to-stereo conversion or a mono to stereo conversion. The range of adjustment is + / - 20 milliseconds; when the control is set to its center (zero), the time offset between the two stereo channels will be zero milliseconds. To fine-tune this parameter (as with any of the parameters in the program) Use the UP ARROW key to increment and the DOWN ARROW key to decrement the time offset value. This is especially useful when performing the Analog Magnetic tape recording azimuth correction procedure.

This feature has three applications:

1. **Analog Magnetic tape recording azimuth correction:** When analog magnetic tapes are recorded or reproduced, the gap of the respective head (recording or playback) should ideally be perfectly normal (perpendicular) to the direction of the tape movement. If, in either of the two mentioned processes, the respective head gap is off-normal (off-azimuth), two types of signal degradation will occur. The first phenomenon results in the loss of the high-end of the audio spectrum frequency response. The second effect produces a phase shift of one channel with respect to another thereby "smearing" a stereophonic image. If you are reproducing a monophonically recorded cassette tape via a stereophonic playback machine, the effect of azimuth misalignment on high frequency loss can be somewhat improved by compensating using the "Time Offset" feature (the same applies when a monophonic half-track reel-to-reel tape is reproduced on a quarter track

machine.) To compensate for the effect of azimuth misalignment, adjust the "Time Offset" control until the best high frequency response is heard with your stereo system placed in monophonic playback mode while previewing. If you are dealing with stereophonic source materials, it is hard to determine the correct phasing by merely adjusting the "Time Offset" for the best image. But, if you place your stereo system in monophonic mode with a stereo tape source, and follow the same procedure just described (use the "Time Offset" feature to adjust for the optimal high frequency response), this will also correspond to proper left channel to right channel phasing, and will therefore produce the best stereo image. Of course, you must place the system back in stereo mode after the file conversion has been completed in order to appreciate the results.

2. **Improving the Intelligibility of Forensics recordings:** It has been shown that audio signals, which are very difficult to discern, can be made more intelligible when the brain is presented with the same signal twice with a short time interval in-between. The human brain processes the information that arrives at each ear independently by the left and right brain. The information is then shared and compared between the left and right hemispheres. Comprehension of the information results from the interaction of both hemispheres communicating with one another. When a delay is injected between the information heard by the left ear and the right ear, the intelligibility factor is improved. This phenomenon was discovered by one of the British intelligence agencies (MI-5 or MI-6) in the late 1950's. The technique involves the use of stereophonic headphones (so that each ear is acoustically isolated from the other), and an adjustable delay inserted between the two reproducers. The Time Offset feature can be used for this application. First, use the standard techniques for cleaning up the Forensics recording. Then, apply the monophonic signal to the file conversion algorithm using the "Time Offset" in preview mode with headsets to adjust for the best intelligibility. This may improve your ability to transcribe conversations, which would otherwise be

impossible to discern. This technique does not work with loudspeaker reproduction.

3. **Stereo Simulation:** The "Time Offset" feature is one of several methods provided by DCArt10/DC Forensics10 that will produce a stereophonic effect. Merely start with a monophonic file, and convert it to a stereophonic file with some value of "Time Offset" applied. Adjust the "Time Offset" control to produce the spatial effect that you desire while using preview.

- **Phase Inversion:**

180-degree phase inversion can be applied to either channel by checking the appropriate box. This feature can be used to correct for out-of-phase stereo masters wherein the cutting head was connected incorrectly to the cutting head power amplifier, or one microphone was wired backwards on a stereo recording situation or other similar defects. It can also happen if one channel of your stereo phono cartridge had been wired backwards creating a 180 degree phase shift and a distortion of the stereo image. Please note that if both boxes are checked, you are right back where you started - - - phase inversion correction will not occur. You must only check one of the two boxes to correct for one out of phase channel.

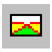
- **Amplitude dB (gain)** – 96 dB to + 9.9 dB**:**
  These two controls (one for the left channel and another for the right channel) are used to set the amplitude of each channel in terms of dB. Sometimes, these are referred to as “gain” controls. The amplitude level is shown in dB directly above each Amplitude slider control.

- **Gang:**
  This checkbox will lock the Amplitude dB (gain) controls together so that they move simultaneously.

Note:

The gain and the Time Offset controls on the File Converter routines can be adjusted "live" after the Preview mode button has been clicked.

Cross Fade Filter



The Cross-fade filter is used to join sections of different .wav files into a single .wav file in a special way. Rather than just abruptly ending one file and starting another, the Cross-fade filter will smoothly fade from one file to another. During the time that the files overlap, the destination file is gradually faded to silence, while the source file fades from silence to full volume. This filter is also available via its icon, or from the Filter Menu or from the Edit menu as one of the “Paste” functions.

Note:

Crossfade can only be performed in Classic Edit Mode. Therefore, this menu item and the corresponding Crossfade icon will only come alive after you have files situated in both the Source and Destination workspace. One file can then be cross-faded into the other.

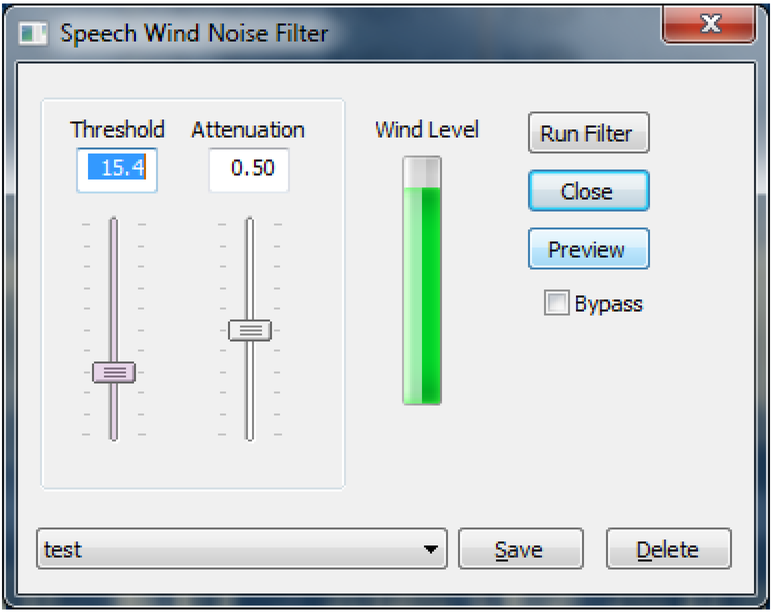
Using the Filter Menu Cross-fader (Tutorial)
Warning: This is not undo-able

1. Open a Source File which you desire to be the next segue song in a cross-fade sequence.
2. Open a Destination File that you desire to be the first number in the cross-fade sequence.
3. Highlight the Source File up to the end of the file.
4. Highlight the Destination File segment at its ending where you desire the cross-fade to occur (the Source File will become un-highlighted when you do this).
5. Click on the Cross-fade feature under the Filter Menu.
6. Set the Gain Controls as follows: (Default settings are correct)
   - **File # 1 Level**
   - Start = -100 dB
   - Stop = 0.00 dB
   - **File #2 Level**
   - Start = 0.00 dB
   - Stop = -100 dB

7. Choose the Cross-fade timing that you desire. Linear produces a pleasing result on most material.
8. Click on "Do Cross-fade."
9. The final results will reside in the Destination Workspace following processing. The entire Source File will be appended to the Destination file following the cross-fade sequence. You may have to “Zoom-Out” to see the entire result in the Destination Window.

Wind Noise Filter (speech)

High levels of wind can interfere with outdoors audio recordings. Often, audio professionals will apply a fixed High Pass Filter to the recording to reduce this annoyance. But, this approach also damages the good audio that one is trying to capture. The Diamond Cut Wind Noise Filter (speech) is a special IIR based dynamic filter using multiple bands across the lower end of the audio spectrum (ranging from 20 Hz to 500 Hz). It dynamically activates these filters depending on the measured presence of random noise within any of those bands leaving behind the good audio while attenuating the random wind noise. Although this is not a perfect filter, it is very useful in spoken word outdoor situations and can produce a greatly improved result compared to the fixed High Pass Filter approach to the problem.



Wind Noise Filter

Threshold Control: (1.0 to 50.0) Increased aggressive filtering action occurs at lower threshold control settings. The least aggressive setting is 50.0 (all the way up). Adjust this control until the "Wind Level" meter is fluctuating in sync with the wind noise, and the wind noise is dynamically being reduced by the filter.

Attenuation Control: (0.01 to 1.00) This slider control allows you to set the amount of attenuation provided by the Wind Noise Filter. The maximum degree of attenuation occurs at a setting of 1.00. Adjust this for the best balance of the sound quality vs. low frequency artifacts.

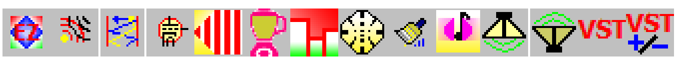
Wind Level Bar Graph: This indicator shows the dynamic activity of the wind noise filter. The higher the meter graph deflection, the greater is the action of the wind noise filter.

Operation: Bring up the file with wind noise interference. Bring up the Wind Noise Filter. Set the Threshold control to 15.0 (initially). Set the initial Attenuation value to 0.50. Next, adjust the Threshold control upwards or downwards until the Wind Level Bar Graph modulates (deflects). Keep adjusting the Threshold control until you perceive the average deflection to be around the center of the bar graph display. It may occasionally top out or bottom out - - - that is ok. Finally, adjust the Attenuation control for the optimal level of noise reduction and a minimal amount of artifacts.

Note: A Wind Noise Demo file is provided to help you learn how to use this filter. It is called “Wind Noise Filter Demo.mp3”.

The Effects Menu

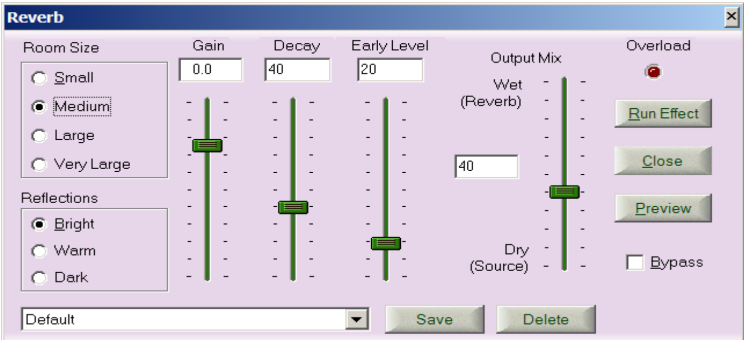
The Effects Menu houses a good portion of the audio enhancement tools located in DCArt10/DC Forensics10. This is the place to head when you want to breathe new life into an old or poorly recorded file.



The Effects Toolbar

Reverb

The Reverb effect is used to add a realistic room sound to a recording. The reverb is capable of simulating different size rooms, with different types of reflective surfaces and decay times. The reverb effect lets you control the overall room size, decay time, early reflection level, and the mixture between the original material and reverb sound.



The Reverb Effect Window

Reverb can be useful when dealing with recordings, which are completely "dead" as originally mastered. As with the various other filters, the reverb effect can be applied globally or selectively (using sync mode) to a .wav file. The reverb effect can also be used to convert a monophonic recording to a simulated stereophonic recording. The following controls are provided on the DCArt10/DC Forensics10 Reverb:

- **Room Size:** (check box)
  - Small (Club)
  - Medium (Auditorium)
  - Large (Concert Hall)
  - Very Large (Stadium)

- **Reflections:** (check box)
  - Bright: (Simulation of a very "hard" acoustical environment, as in a stone building)
  - Warm: (Simulation of a typical auditorium or theater)
  - Dark: (Simulation of a heavily draped auditorium)
- **Decay:** Control Range 1 to 99 in relative units.

The decay control affects the dampening effect of the algorithm on the reverberated signal. The higher this control is set, the longer the reverberation "dwell-time." The lower that this control is set, the quicker will be the decay of the reverberated waveforms.

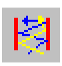
- **Output Mix:** (Slider Control) Control Range: 0 to 100 in percentage units.

The Output mix determines the amount of the reverb effect that is fed into the system output. When the control is set to zero (dry), there will be no reverb effect. When the control is set to 99, there will only be the reverb effect, with the source signal bypassed. Useful ranges of control are usually in the 5 to 25 range, but if you are looking for extreme effects, you can get them if desired.

- **Reverb Presets:**

The Reverb is equipped with a number of descriptive presets. This is a good place to start from when using the reverb effect. Choose the desired acoustical environment (which can be selected and previewed “on-the-fly”). After you have found something close to the sound you desire, revert to the various controls to fine “tweak’” the reverb for the exact sound you are looking for.

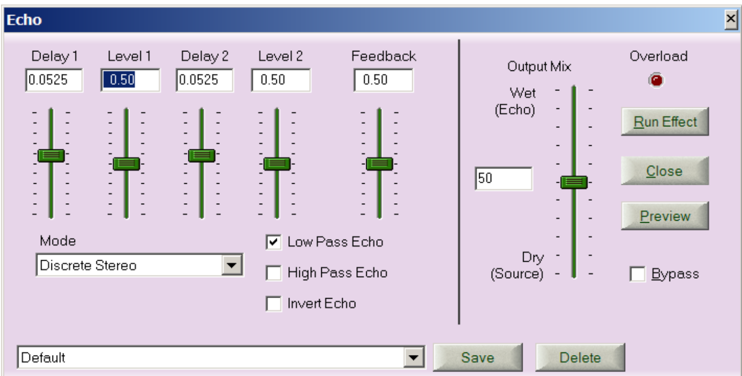
Echo Effect



The Echo Effect is a digital simulation of the old-fashioned magnetic tape delay lines (sometimes called “Tape Delay Echo Chambers”) that were popular in the early 1950s and throughout the 1960s. These devices worked on either of the following basic principles. One method varied the distance between the recording and the playback head producing a variable time delay, a portion of which could be fed back to the system’s input. A second method used a variable speed motor driving the tape media with the recording and playback head positions being fixed, producing similar overall results. The Echo Effect contains two independent delay lines, which can be used in a number of different modes providing it with a great deal of flexibility compared to the old analog systems. Because it performs its function utilizing digital techniques, it does not suffer from the noise and distortion buildup associated with the older analog systems. However, if the so-called “retro” sounding echo chamber is desired, you can use the Echo Effect in conjunction with the Virtual Valve amplifier in the Multi-Filter. The Echo Effect is useful in any of the following applications:

- Adding special effects to speech or music.
- Creating certain acoustical simulations.

- Enhancing the articulation sounds on Forensics Audio Wave Files.
- Simulating Stereo from Monophonic sources.
- Creation of "Comb" Filters.
- Adding a simple time delay to one channel of a file.
- Double-Tracking a Vocal to give it more depth.



DCArt10's Echo Effect

The Echo Effect has the following controls:

1. Delay 1: Range: 0.0001 to 5.0000 Seconds (10 uSec to 5 Seconds)
2. Level 1: This control varies the level of the effect produced by Delay 1.
3. Delay 2: Range: 0.0001 to 5.0000 Seconds (10 uSec to 5 Seconds)
4. Level 2: This control varies the level of the effect produced by Delay 2.
5. Feedback: (0.00 to 1.00) This controls the amount of signal fed from the Delay Line outputs back to its input. When this control is set to zero, you will only hear a single delay associated with each of the two delay lines. The higher that this control is set, the greater will be the reverberation sound due to feedback.
6. Output Mix: Range 0 to 100 (0 = Source Only Signal {Dry}) & (100 = Echo Only signal {Wet}). Output Mix controls the

ratio of the Source signal with the processed signal. A good nominal starting point for this control is 50.

7. Mode (Modes of Operation Selection Box)
    - Discrete Stereo: (Delay 1, Level 1 = Left Channel & Delay 2 Effect & Level 2 = Right Channel Effect)
    - Discrete Stereo Reverse: (Delay 1, Level 1 = Right Channel & Delay 2 Effect & Level 2 = Left Channel Effect)
    - Summed Mono – Stereo Out (Output): This mode is the same as Discrete Stereo except the input signal is summed to Monophonic before being applied to the Delay Lines. The "Dry" signal remains Stereophonic in this mode.
    - Summed Mono – Mono Out (Output): In this mode, the input signal is summed to Mono and then fed to Delay 1 Level 1. From there the signal is fed to Delay 2 Level 2 before being fed to the Output. In other words, the two delay lines are cascaded. The "Dry" signal remains Stereophonic in this mode.

      Important Note: This mode can produce sustained oscillations if the feedback control is set too high.

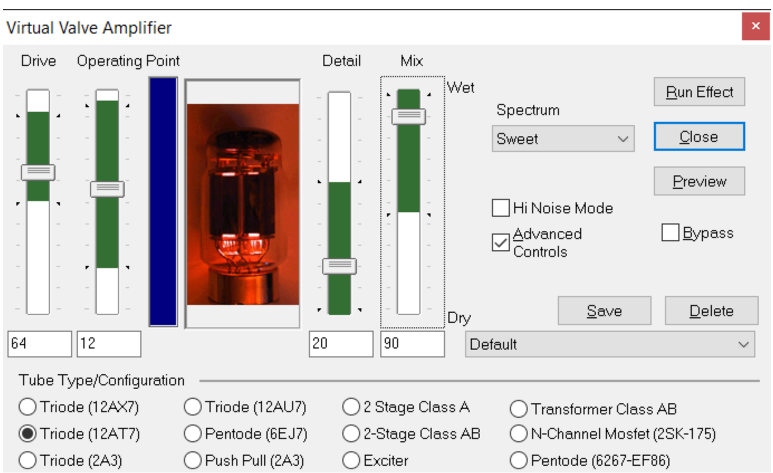
    - Overload Indicator: This indicator lights up in red when the Echo Effect system is overdriven. If this occurs, back down on the level or mix controls to prevent clipping distortion from occurring.

Ancillary Echo Effect Controls:

1. Low Pass Echo: This inserts a 7 kHz Low Pass Filter in the Effect Signal Path.
2. High Pass Echo: This inserts a 5 kHz High Pass Filter in the Effect Signal Path.
3. Invert Echo: This phase inverts the Effect Signal.
4. Bandpass Echo: This is achieved by checking both the Low Pass and High Pass Echo checkboxes. It produces a Bandpass response having corner frequencies of 2 kHz to 8 kHz and is inserted into the Effect Signal Path.

<u>Virtual Valve Amplifier</u>

The Virtual Valve Amplifier™ (or VVA™) is a computer simulation of a number of vacuum tube amplifier circuits. (Valve is the British term for electron tube.) We call it the "Virtual Valve Amplifier" because that sounds cooler than "Virtual Tube Amplifier". Its effect is to add "tube-warmth" to the sound of a recording. This is sometimes desirable to apply to DDD (purely digital) recordings. It can also be used to add subtle harmonics to very old recordings. A harmonic exciter is also included with the Virtual Valve Amplifier. It is important to note that the Virtual Valve amplifier is using real tube circuits, and real tube non-linear device characteristics to produce its effect. The wide range of adjustability of this algorithm will allow you to create an amplifier that runs the gamut in sonic performance from "grit-guitar" to "high-end audiophile."



The Virtual Valve Amplifier

The Virtual Valve Amplifier (VVA) produces a variety of sounds associated with valve (electron tube) based amplifiers. The effects run the range from a nuanced "tube warmth" sound to extreme effects like "guitar amplifier overload" or "fuzz box." The DCArt10/DC Forensics10 VVA accomplishes these effects through the use of actual electron tube circuits, which are simulated by your computer. The electronic models of the various tube amplifier circuits have been derived from the "large-signal" transfer functions of the various tubes and output transformers you can choose from. This data has been derived from extensive bench measurements of tube amplifier circuits under varying operating conditions. The circuits are not "idealized", but real circuits. As such, the effects will sound literally as would be heard if you were to process a signal through a physical electron tube amplifier having the devices or configuration that you have chosen.

However, with the VVA, you have a great deal more control over the various sounds that can be produced, since controls, which are not normally found on electron tube equipment, have been provided. Parameters such as "Operating Point" (sometimes referred to as "Q" point by engineers) are usually fixed by the amplifier manufacturer. "Drive" is determined by how loud you play a "physical" amplifier, but with the VVA, the output level remains constant independent of drive due to an internal gain compensation algorithm. The following is a listing of the controls that are provided on the DCArt10/DC Forensics10 VVA:

- **Drive Slider:** 1 to 100

This control effects the degree of modulation applied to a given tube amplifier circuit and centered about the operating point setting. The higher the drive level setting, the greater will be the production of predominantly even order harmonics due to the circuit's asymmetrical non-linearity. As a result, there will be more "effect" as this control is increased. Also, the "depth" of the effect is determined in part by the degree of drive applied.

- **Operating Point (or Harmonic Control) Slider:** -100 to zero (in the middle) to +100

- **VVA Mode:**

The operating point control performs two different functions, depending on the Tube Type / Configuration selected. When a triode or class A amplifier is chosen, it sets the operating point for the particular tube or amplifier configuration that you have chosen. Operating point also determines the device's bias value at zero signal input. The distribution pattern of harmonics, which are introduced into the output of the amplifier, are determined to a large degree by the location of the operating point. When the control is set to + 100, (all the way up) the devices are operating close to “saturation,” and when the control is set to – 100 (down), the devices are operating close to “cutoff.” The non-linearity distortion distribution is different near cutoff as compared to operation near saturation. You can use the operating point control to achieve variations in the desired “tube effect.” Most audio preamplifier tubes such as the 12AX7 are the most linear in the middle of their dynamic operating curve (control set to the middle “0” position).

- **Harmonic Exciter Mode:**

When the system is placed into harmonic “Exciter” mode, the operating point control reverts to a “Harmonics Control” which varies the distribution of harmonics that are produced by the VVA. The Harmonic Exciter is designed to provide the following audio enhancements:

A. Synthesize the upper register harmonics that may have become lost through “generation loss” or due to the poor frequency response of the master recording.
B. Add “presence” to a vocal recording.
C. Create a more “up-front” sound on any modern recording.

When the control is set to +100, both even and odd harmonics are produced. When the control is set to –100, only the first 3 to 4 even harmonics of the fundamental are produced. Settings in between will produce varying combinations of the two extreme settings. The system is placed into harmonic Exciter mode by checking the “Exciter” box listed under Tube Type / Configuration, located at the bottom of the VVA window. (You must be in “Advanced Controls” mode to see the Exciter checkbox.) The magnitude of the inserted Exciter effect is controlled by the “Mix” control.

- **Operating Point Indicator:**

Vertical undulations are graphically presented proportional to signal level, drive, and operating point. The Operating point indicator will have a blue background in standby modes of operation and a blue background with vertical undulations appearing in any of the operational modes of the VVA. An orange modulating vertical bar during operation indicates the modulation adding to and subtracting from the operating point center value. The magnitude of the drive level to the amplifier is indicated by orange undulations plus and minus about the operation point. So both the effects of the drive and the operating point slider are presented via the same graphic display in an intuitive manner. (Operating point is usually referred to as the "Q" point in engineering terms which stands for quiescent point.)

- **Detail:** 0 to 100

The detail control allows you to control the sensitivity of the VVA to the more delicate nuances of the musical material presented and processed. The higher the setting, the greater the effect will be on the material.

- **Mix:** 1 to 100

The Mix control affects the degree of VVA signal, which is re-inserted into the signal path. At its maximum setting of 100 (wet), the dominant signal pathway is exclusively through the VVA, and when the control is set to 0 (dry), only the non-processed signal is fed through the system. You can choose any level in between which appeals to your taste.

- **Spectrum (Drop-Down Menu):** -Silk -Sweet -Warm Full Range -Fat Bass

The "Spectrum" range control effects the spectral distribution of the harmonic by-products, which are passed through to the systems output. The most desirable setting is very much a function of the musical material which is being processed and the desired tube sound.

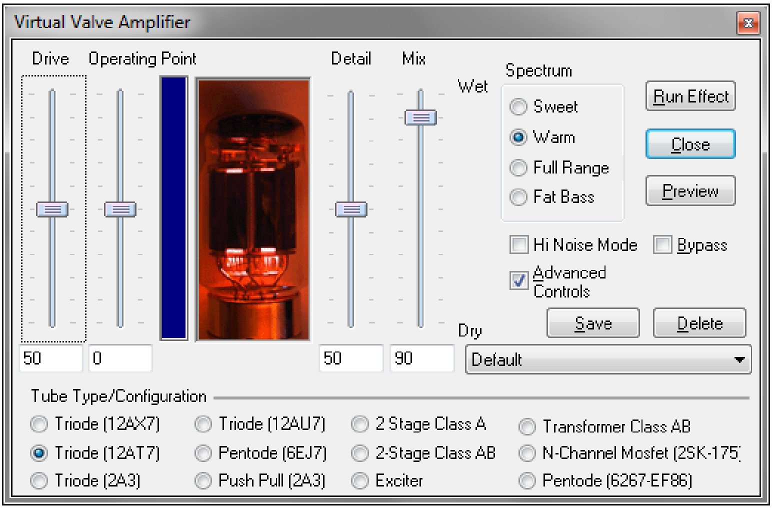
"Full Range" mode is generally used to "round out" the entire audio spectrum of a recording. It is important to note that when you are operating a tube in Full Range mode, there is a dramatic increase in the propensity for the production of inter-modulation distortion, which is not particularly desirable (except for heavy metal rock). To minimize this effect, you will have to use much smaller values of "Drive" to obtain reasonable results when compared to the "Silk", "Sweet", "Warm", or "Fat Bass" modes. Sweet and Warm modes are preferable to use over Full Range mode because they dramatically reduce the propensity of the electronic valves from producing inter-modulation (IM) distortion, leaving behind only the more pleasing harmonic distortion type. Silk mode is primarily useful on modern high quality recordings and less effective on old 78s or similar older types of recordings.

Fat Bass Mode

"Fat Bass" mode is used to add some dimension to sterile sounding bass. It restricts the VVA's action to the excitation band of frequencies below 175 Hz. However, it allows harmonics to be produced that are higher adding that "Fat Bass" sound typically associated with tube amplifier circuits. Push-pull circuit configurations will produce dominantly Odd harmonic bass effects while Class A circuits will produce both Even and Odd harmonic effects. "Sweet" and "Warm" settings restrict the operation of the VVA to the upper portions of the audio spectrum, producing pleasing tube generated harmonic and compression effects.

- **Advanced Controls Checkbox:** On/Off

This enables the more advanced controls of the VVA to be displayed to the user, if desired. When this control is not checked, default values will be chosen for some of the control settings, tube types, amplifier configuration, operating point and detail controls based on the selected VVA preset. When the Advanced Controls are turned off, you will still have control over the VVA Drive and Mix settings.



The Virtual Valve Amplifier
(with advanced controls showing)

- **Hi Noise Mode Checkbox:** On/Off

Some extremely noisy recordings (like certain 78 RPM records) can become even noisier when the VVA is applied to their signal. So, not only will you obtain the pleasing harmonics that you desire, but you will also obtain the undesirable elevation of the recordings already high noise floor. The "Hi Noise" mode is designed to alleviate this problem. When this checkbox is activated, the system changes into a noise suppression mode. The "Detail" control reverts into what is redefined as a "Threshold" control. Harmonics are only generated when the average harmonic signal level exceeds the set threshold which can be varied by this control. The green indicator (a virtual LED which becomes visible directly above the threshold control when operating in Hi Noise Mode) will illuminate when harmonics are being generated by the system that are above the set threshold level. Moving the threshold control upwards increases the noise reduction effect, but also reduces the VVA effect. Moving the control all the way down will keep the VVA system active all the time with no noise suppression effect at all. Moving the control all the way up will keep the VVA effect off all of the time. Adjust the threshold control somewhere in the middle so that quiet passages squelch the system as indicated by an extinguished LED

indicator. Reasonable levels of noise suppression are had when the green indicator occasionally flashes while previewing a track. This “Hi Noise” mode will reduce the incremental noise contributed by the VVA to an already noisy recording. Thus, you can enjoy harmonic generation and increased brilliance on dull & noisy recordings without suffering an elevation of their noise floors.
Please note that the Hi Noise Mode is designed to be used in conjunction with the VVA’s “Sweet” or “Warm” settings only.

- **Bypass:** On/Off

This control allows you to quickly compare the effects of the processed signal produced by the VVA to the unprocessed signal, while the program is in “Preview” mode.

- **Preset Settings:** Listing

The VVA has a list of pre-sets, which will be a valuable starting point from which to fine tweak the adjustment controls to your desired taste. These presets are somewhat descriptive to help you in making a choice. The choices can be changed in real-time while running the program in Preview mode, so that you may compare the various presets.

- **Tube Type Checkbox:** Checkboxes for the following Valves (tubes) or circuit configurations:

    A. **Triode (12AX7)** - This configuration incorporates this high-mu dual triode into a typical RC coupled class A audio pre-amplifier configuration. This tube was chosen, because it had been and still is the industry standard pre-amplifier valve. It has a relatively flat linear operating region in the middle of its dynamic operating range, producing relatively lower levels of distortion compared to some of the other devices offered in the VVA. But, by moving the Operating Point to either the saturation of cutoff extreme, more “tube-warmth” effect can be produced by this device. This is the same device as the European type ECC83.

    B. **Triode (12AT7)** - This amplifier configuration utilizes the same type of RC coupled pre-amplifier circuit described above, but using a 12AT7 high-mu dual triode. The primary difference is that the 12AT7 was designed primarily for RF mixing applications. As a result, it has a large degree of non-linearity throughout its entire dynamic operating range, including the middle. As a result, you will be able to obtain a higher level of “evens” (even order) harmonic distortion (the most pleasing harmonic distortion) in which to add back into the signal path of the VVA. It also produces some odd order distortion products. This is the same device as the European type ECC81.

C. **Triode (12AU7)** - This amplifier configuration is simulating the driver / phase inverter stage of a push-pull power amplifier. It utilizes the 12AU7 medium-mu dual triode, and, like the previously described circuits, is biased class A and is RC coupled. This device also has a significant non-linearity in the middle of its dynamic operating curve. (In power amplifiers, some of this non-linearity is removed via the use of negative feedback, and decreasing the mix control level on the VVA simulates this phenomenon.)

D. **Pentode (6EJ7)** - This single stage, high-gain microphone amplifier configuration utilizes a sharp-cutoff pentode. It can produce a very pleasant “tube-warmth” effect when the operating point is properly set. This device is the same as the European type EF183.

E. **2 Stage Class A** - This is an 8 Watt class A power amplifier, consisting of a 12AU7 medium-mu triode driving a single 6L6GC beam power pentode audio output valve. Its effects are distinctive due to the convolution of the non-linearity of the triode interacting with those of the pentode, with both devices operating in class-A mode. The 6L6GC is similar in performance to the industrial type 5881, and also the European tetrode, type KT-66 (KT = Kink-less Tetrode).

F. **2 Stage Class AB** - This is a 25 Watt class AB power amplifier, consisting of a 12AU7 phase inverter / driver, pushing a pair of 6L6GC beam power pentodes. Because the circuit is push pull, the output devices produce a more symmetrical and reduced even-order distortion characteristic distribution. The more dominant distortion products are the odds with this circuit configuration. The operating point is fixed at the factory, and cannot be adjusted for this amplifier configuration.

G. **2A3 Push-Pull** - The 2A3 is what some people refer to as a "retro – triode". It was invented in the 1930's, had a directly heated cathode, and produced a high power output at its time of development. It was often found used in theatrical applications and public address systems. The "Push-Pull 2A3" VVA setting uses the 2A3 triode implemented in a "push-pull" class AB_1 power amplifier circuit designed to produce 15 Watts of output power. This configuration exhibits a more linear output transfer characteristic compared to its Pentode push-pull counterpart. We have included the 2A3 tube in this particular configuration in the VVA because a musician friend of ours (Les Paul) recommended that we do so because of its unique sonic characteristics. He explained to us that he used a push-pull pair of these devices as the power amplifier to "cut" all of the records that he released from his own home studios. The reason that he used these was the extremely clean sound that they produced. The particular devices that we used to create the 2A3 VVA models were of the "dual – plate" variety. The devices used in the characterization process for the 2A3 based VVA were taken from new (unused) but old stock (NOS) and were manufactured for the military by RCA Victor in 1953.

H. **2A3 Single-Ended** - This is a single ended class A power amplifier implemented using the 2A3 power triode. It exhibits reasonably good linearity and about 4 Watts of audio in a “single-ended” class-A configuration. Its dominant distortion products are “evens.” This is the only power triode in the VVA suite of tubes.

I. **Exciter** - This check box enables the Harmonic Exciter feature of the DCArt10/DC Forensics10 VVA. The exciter uses a simulation of a vacuum tube rectifier (6X4) to produce harmonics. Asymmetry between the positive and negative going transfer function establishes the relationship between the degree of even and odd harmonics produced. For more details on its performance, please refer to the Harmonic Exciter description under the Operating Point Control description.

J. **Transformer Class AB** - This check-box enables a push-pull, transformer coupled, 6L6GC based, class AB, 20 Watt power amplifier having a 12AU7 based driver / phase inverter stage. It produces a distortion dominated by odd order components since most even order products cancel out in push-pull circuits.

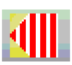
K. **6267 / EF 86 Pentode** - The 6267 / EF86 pentode was suited well for use in low-level preamplifier service where low noise and minimal microphonics were important. It was often found used as the first-stage amplifier in tape decks. Its high-gain characteristic provide it with an interesting family of operating curves that provide useful harmonic distortion and signal compression in the VVA.

L. **2SK-175 MOSFET** - This device is not a tube, but rather it is an N-Channel Audio Power MOSFET (the P-Channel compliment of which is the 2SJ-55). It was commonly found in high power, high quality audio power amplifiers and is included here because it has a set of operating curves which differ in shape somewhat from the various VVA electron tubes. It will provide you with a different distribution of distortion harmonics which you may find pleasing in some circumstances.

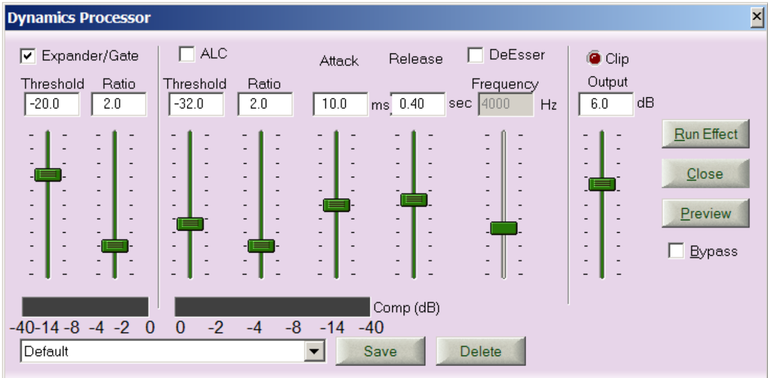
Just like the other filters and effects, the VVA is equipped with a set of descriptive presets. This is always a good place to start from when using the VVA. After you have found the preset, which most closely resembles the sound you are looking to achieve, you can go back and "tweak" the controls more precisely. After you have established a group of settings that you would like to keep, use the "Save" settings feature to give your preset a name so that you can recall it in the future.

Note: Voltage Amplification stages usually produce a phase inversion of the applied signal (180 degrees). All of the VVA devices and/or systems are phase corrected to zero degrees for user convenience.

Dynamics Processor



The dynamics processor provides you with the ability to control the dynamic signal content of the audio envelope of a .wav file. Included are Compression and Limiting, downward Expansion/Noise Gating, ALC, and De-Essing.



The Dynamics Processor

The Dynamics Processor provides you with three functions related to the control of the dynamic range of an audio signal. The functions are as follows:

- **Expander/Gate**

(Wide Attack & Release Time Ranges are provided in version 7 on Forward)

This system is a downward dynamic expander. When the applied signal is below the threshold setting, the dynamic range of the signal is increased depending on the value of the ratio setting. In other words, the incremental attenuation of the .wav file signal is proportional to the ratio setting when it is below the threshold value. The higher that you set the value of this ratio, the greater will be the degree of downward expansion applied to the signal. Signals above the threshold value are passed through the system with no processing applied. When the ratio control is set to its maximum value (control set all the way up), the system will behave like a Noise Gate (several noise gate presets are available in the Presets menu).

Speeches, sermons, lectures and other similar audio situations often produce undesirable echoes or reverberations in large venues. The Diamond Cut Productions Dynamics Processor when used in Expander/Gate mode can de-verb (reduce) this problem. There are five presets that you can chose from if you encounter this audio problem, having the pre-fix of either reverb reduction or echo reduction. These presets are a good starting point to fix this type of audio problem.

When more modest values of the ratio control are used, the system can produce some improvement in the dynamic range and also the average signal-to-noise ratio of a .wav file. The Expander has the following controls available:

A. Expander/Gate Checkbox: On/Off

Checking this box will enable or disable the Expander/Gate function of the Dynamics Processor.

B. Threshold: -70 dB (control down) to 0.00 dB (control up)

This control establishes the signal level below which the Expander performs its process on the .wav file signal.

C. Ratio: 1.00 (control down) to 29.99 (control up)

This control determines the degree of downward expansion applied to the .wav file for signals that are below the threshold value setting. The higher the number chosen, the greater will be the signal expansion effect.

D. Expander bar graph: Horizontal meter indicating from 0 to –40 dB.

This meter indicates the actual value of downward compression in dB, which is being applied to the .wav file signal.

E. Attack: 0.1 mSec to 1000 mSec

This control is used for the Expander/Gate, Compressor, ALC and De-Esser functions of the Dynamics Processor. It determines the time constant associated with the onset (delay) of any of the Dynamic Processor effects.

F. Release: 10 Seconds to 0.01 Seconds

This control is also used for the Expander/Gate, Compressor, ALC and De-Esser functions of the Dynamics Processor. Its setting determines the delay time associated with the decay of the particular process chosen.

- **Compressor**

This system is an upward compressor and is the only thing active when the Expander/Gate is turned off. When a .wav file signal level is above the threshold setting, the dynamic range of the signal is decreased, the degree of which depends on the value of the ratio setting. In other words, the incremental attenuation of the signal is proportional to the ratio setting when it is above the threshold value. The higher that one sets the ratio value, the greater will be the degree of compression. When this ratio value is set to its maximum, the system will behave like a Limiter. Signals below the threshold value are passed through the system with no processing applied. When the ratio control is set to its maximum value (control set all the way up), the system will produce the largest degree of compression. The Compressor has the following controls available:

A. Threshold: -60 dB (control down) to 0.00 dB (control up)

This control is similar to the threshold control for the expander, but establishes the signal level above which the compressor performs its process on the .wav file signal.

B. Ratio: 1.00 (control down) to 29.99 (control up)

This control determines the degree of compression, which is applied to the .wav file for signals that are above the threshold value setting. The higher the number that is chosen, the greater will be the effect on the signal.

C. Expander bar graph: Horizontal meter indicating

from 0 dB to 40 dB. This meter indicates the actual value of compression in dB that is being applied to the .wav file signal.

D. Attack: 0.1 mSec to 1000 mSec

This control is used for the Expander/Gate, Compressor, ALC and De-Esser functions of the Dynamics Processor. It determines the time constant associated with the onset (delay) of any of the Dynamic Processor effects.

E. Release: 10.0 Seconds to 0.01 Seconds

This control is also used for the Expander/Gate, Compressor, ALC and De-Esser functions of the Dynamics Processor. Its setting determines the delay time associated with the decay of the particular process chosen.

Important Note: The Punch and Crunch Effect provides you with a somewhat different approach to compression and expansion using multiple frequency bands.

De-Esser

A de-esser is a form of compressor, which is only reactive to the frequencies associated with the pronunciation of the letter "s" ("ess"). It is necessary to perform this function on over-modulated signals in the "s" frequency range and is adjustable from 1,000 Hz to 10,000 Hz. This occurs due to poor mike technique, a poor initial mix, improper mic channel equalization, or insufficient "padding" of the mic input circuit during the recording session. When the frequencies in the sensitive band are detected and are above the threshold setting, compression will be applied to the degree determined by the compressor ratio control. To place the compressor in the De-esser mode, click on the box by the same name.

One global control is provided in addition to all of those mentioned above. The output gain allows you to correct for overall amplitude effects (attenuation or gain) that any of the dynamic processor functions may have on the overall output signal level. Presets have also been provided to get you started with reasonable setup parameters for the various dynamic processor functions.

- **Automatic Level Control (ALC or AGC)**

The Dynamics Processor includes an automatic level control feature (ALC). Sometimes, these algorithms or systems are referred to as automatic gain controls or AGC's. This feature provides upward expansion of signals below the threshold line and downward compression of signals above the same threshold. This feature is useful in Forensics applications where there is a large variation in signal levels between several different parties that may be communicating with one another. It is also useful for the broadcast of live sporting events (if you have the DC Forensics10 version of the product) in which the crowd reaction is of interest when the announcer is not speaking. Simply clicking on the "ALC" box in the Dynamics Processor activates this feature. The threshold, attack, and release controls are still active when this function is invoked.

Output Control and Clip LED

Since the Dynamics Processor can add considerable gain to your system (especially when operating in Expander mode), an output level control and a red LED clip indicator are provided. If you observe that the red LED indicator is flashing, then the system is clipping your signal. For the cleanest results, use the Output Control to decrease the level until this no longer occurs.

Reverse File

The Reverse file feature does just that - - - it converts a .wav file so that it will play in reverse. This has several uses:

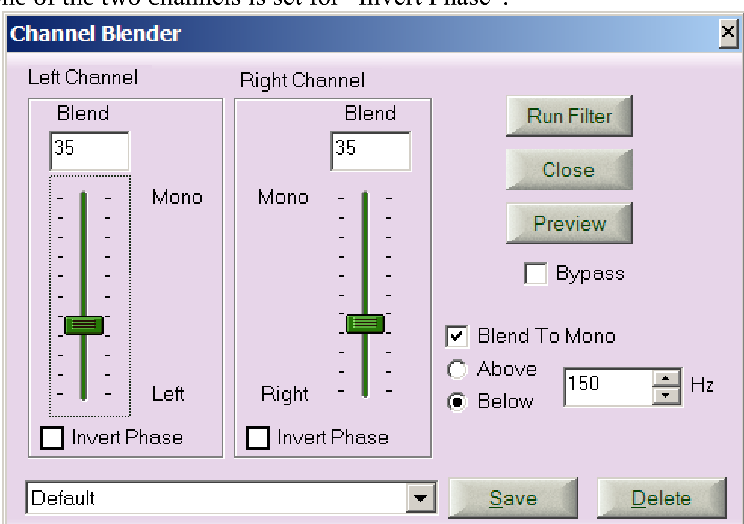
a. Sometimes, file reversal is beneficial for removing stubborn ticks or pops with any of the Impulse Filters. By running a reversed file through an impulse filter, sometimes clicks that were otherwise too difficult to detect, may be found and removed. Of course, when the process has been completed, you must reverse the file again, so that it may be heard in the normal forward direction.
b. The Reverse File feature may be used for reversing metal stamper recordings that have been transferred on standard turntables that were not capable of running in reverse. Simply record the metal stamper which is reversed (using the appropriate bi-radial stylus), and then run the "Reverse File" feature to correct the transfer for forward playback.
c. Professionally recorded master tapes are sometimes stored with "tails out". If they have been in storage for very long periods of time, there is a risk of losing oxide during the rewind process. So, it is of benefit not to rewind the tape, but to play the tape backwards while transferring it

to your hard drive. It will sound reversed, but can be restored by using the reverse file feature.

d. It provides an interesting effect. Of course, it can also be used to make sure that there are no demonic subliminal messages recorded in reverse on the music that your children love to listen to!

Channel Blender

The Channel Blender provides the ability to reduce muddy bass from vinyl recordings, decrease the "ping-pong" effect from early stereophonic recordings, minimize multi-path distortion from FM stereophonic broadcast recordings and even remove vocals from stereo recordings. It can also be used as a spatial enhancer (spatializer) when one of the two channels is set for “Invert Phase”.



The Channel Blender

The Channel Blender serves (at least) 7 purposes:

1. In the early days of stereo, channel separation was the rage. Sound engineers often literally segregated the recording artists into separate recording studios or booths in order to maximize the channel separation. This later became known as the "ping-pong" effect. In other words, with stereo separation, if a little was good, and more was better, then too much was thought to be just enough. The Channel Blender can be used to reduce the extreme stereo separation found on some of these early stereophonic recordings, restoring them to a more natural sound.

2. Rumble on Vinyl recordings is dominated by the vertical displacement component of the master recording and playback stylus. Since bass is acoustically non-left or right below about a hundred Hertz, this rumble can be reduced by summing the low frequency signals (to mono) below a certain crossover frequency. This monophonic bass signal can then added back into the main stereophonic signal. The "Blend to Mono" feature performs this function when it and the "below" function are checked. This can add clarity and improved bass definition to vinyl recordings, which sound muddy due to excessive rumble. Keep in mind that rumble is not just a by-product of the turntable from which you are playing a record, but also involves the system which mastered it in the first place. Even though you may have a very expensive turntable, you will still encounter recordings that are laden with rumble. The recommended frequency for this feature is around 125 Hertz with the "below" box checked. Experiment to determine the best results for the material that you are dealing with.
3. FM stereo multi-path distortion, when it occurs, is dominant in the top two octaves of the audio spectrum. By placing the channel blender in Blend to Mono above the corner frequency setting, you can reduce this distortion with a tradeoff of channel separation at the upper end of the audio spectrum. Try corner frequency settings starting at around 5 kHz with the "above" box checked.

4. Lastly, ambience can be enhanced on a stereo recording by phase inverting one of the channels and summing by the L-R rather than the L+R information back into the main signal path. This is accomplished by phase inverting one of the two channels.
5. The Channel Blender can reduce the "thump" that you hear when you play a cracked record by blending both channels to 100 via the Left & Right "Blend" controls.
6. The Channel Blender can also be used as a Spatial Enhancer (Spatializer) or room acoustic dimension expander. This feature can be used to add more of a "live" feel to recordings that have most of the activity occurring in the middle of the sound-stage and where it would be desirable to extend some of those sounds more to the left and right side. This is accomplished by setting one of the channels for "Invert Phase" and the "Blend to Mono" checkbox being checked. You must also set the "Blend to Mono" feature for a crossover frequency less than around 200 to 500 Hz. Four factory presets are provided to help you get started using the spatializer functionality.
7. The Channel Blender can be used as a "Vocal Remover" using several special presets that it contains. Vocal removal (vocal attenuation) requires a Stereo recording source wherein the lead vocalist is contained within both channels.

   There are 4 presets to choose from including:

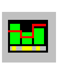
   Lead Vocal Attenuator 1
   Lead Vocal Attenuator 2
   Lead Vocal Attenuator 3
   Lead Vocal Attenuator 4

The Channel Blender has the following unique controls:

- **Left and Right Channel Blend Controls:** These two controls take the summed or differenced signal and add it back into the respective left and / or right channels. When these controls are set to 0, there is no blending effect. At a setting of 100, the blending is maximized. An Invert Phase check box is located in the Left and Right blend control panels. This produces a 180-degree phase inversion of either of the channels before the summation takes place. Therefore, you can blend in L+R (with the phase inversion boxes not checked) or you can blend in the L-R signal (with ONE of the two Invert Phase boxes checked). The L-R signal contains the ambience information on most stereophonic recordings. If both Invert Phase boxes are checked, the signal reverts back to L+R, so if ambience enhancement is desired, only check one box.

- **Blend to Mono Checkbox:** This checkbox sums the signal to monophonic above or below the indicated frequency. You can select a crossover frequency (corner frequency) anywhere between 10 and 10,000 Hertz.
- **Above:** This blends to mono all frequencies above the corner frequency setting. This is used to reduce multi-path distortion from FM broadcasts.
- **Below:** This blends to mono all frequencies below the corner frequency setting. This is used to reduce rumble and muddy bass on vinyl stereophonic recordings.
- **Invert Phase:** This changes (shifts) the phase relationship of the chosen channel by 180 degrees with respect to the other channel. (If both boxes are checked, no net phase shift will occur.) This function is often associated with the spatializer effects (found in the presets area).
- **Presets:** Over 15 presets are provided for you to use as starting points.

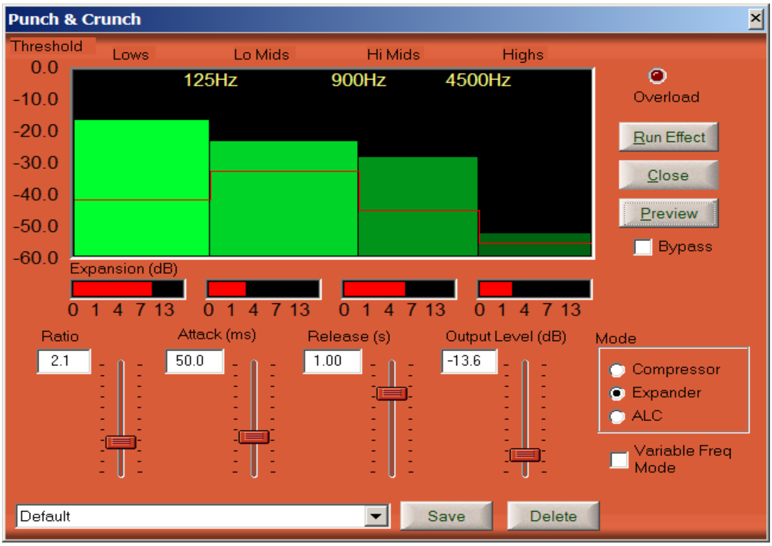
Punch and Crunch



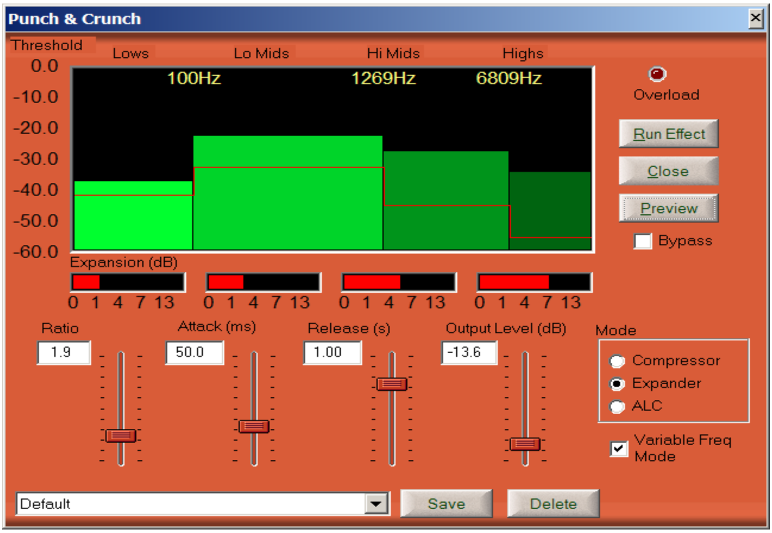
Punch & Crunch™ is a multi-band Dynamics Expander, Compressor or **A**utomatic **L**evel **C**ontrol (ALC). It allows you to modify the dynamics of a piece of audio in a very natural sounding manner. In some cases, you may want to restore lost signal transient response by applying the expander mode, but in other cases you may want to compress the signal in order to produce more "broadcast presence" (or perceived loudness).

Punch and Crunch has a four band Dynamics Expander (Punch) or Compressor (Crunch) or an Automatic Level Control (ALC). Each band is fully adjustable in terms of dynamic amplitude and frequency response. It is useful for a number of applications such as the following:

1. Adding dynamic range or "Punch" back into severely compressed radio broadcast or vinyl recordings.
2. Adding "dial presence" or "Crunch" (compression) to radio broadcasts, without suffering the "pumping" effect found in conventional wide-band dynamic compressors.
3. Decreasing the dynamic range of classical music so that it can be more "listen-able" in restaurant or automotive environments by applying the compressor function.
4. Improving the signal-to-noise ratio and dynamic range of old 78-RPM recordings by using Expander mode.
5. Improving the intelligibility of forensics recordings.
6. Elevating the effective loudness of cable TV broadcast commercials.
7. Compensating for "near-party, far-party" telephone conversation levels via the ALC mode.
8. Special Effects creation.



The Punch and Crunch Effect in Fixed Frequency Mode



The Punch and Crunch Effect in Variable Frequency Mode

It works by breaking the audio spectrum into four separate bands. Each band is independently processed when its signal exceeds the graphical display of its particular threshold line. The ratio control modifies the degree to which the bands are expanded, compressed or ALCed. The actual compression or expansion of any particular band is shown by horizontal bar graphs for each band that are calibrated in dB. The bands are broken into the following “buckets” when operated in the non “Variable Freq Mode”:

- **Band 1:** 0 to 125 Hz
- **Band 2:** 125 Hz to 900 Hz
- **Band 3:** 900 Hz to 4,000 Hz
- **Band 4:** 4,000 Hz to 20,000 Hz

The following controls are provided on Punch and Crunch:

- **Graphical Display of the Four Frequency Bands.** Each band is represented on this graph and the incoming signal present on each band will modulate the vertical displacement of each band. Threshold for each band can be dragged with the left mouse button to the desired position. When the threshold is dragged all the way to the top, that band will have no dynamic compression or expansion effect. When a band is dragged all the way to the bottom, it will have a maximum compression or expansion effect.

- **Graphical Display of the Expansion or Compression of each band.** This graph is horizontally modulated and located beneath each of the four bands. It is calibrated in dB. It will tell you the amount of compression or expansion being applied to its associated band.

- **Ratio:** This controls the degree of compression or expansion applied by the system. When the system is operating in compression mode, you can choose up to 30:1 compression. When the system is operating in expansion mode, you can choose up to 15:1 expansion. In ALC mode, the ratio ranges up to 30:1.

- **Attack:** This determines the time constant associated with the onset of this effect and is calibrated in mSec.

- **Release:** This determines the time constant associated with the decay time for this effect and is calibrated in Seconds.

- **Output Level:** This allows you to adjust the output level of the system. Use this control in conjunction with the Overload indicator to minimize clipping distortion.

- **Mode:** This allows you to choose either Expansion (Punch) or Compression (Crunch) modes of operation. Additionally, you can choose ALC (Automatic Level Control) mode, which is often useful in Forensics Audio applications. It can be used to improve the intelligibility of speech having wide variations in signal level resultant from multiple persons speaking simultaneously on a recording.

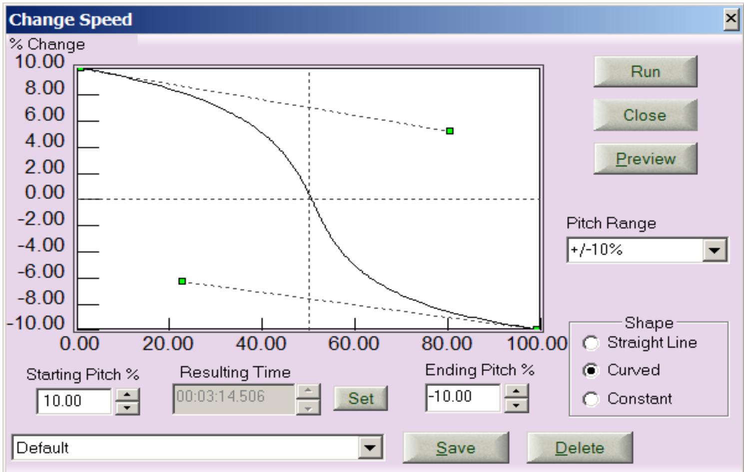
Variable Frequency Mode

This mode, when activated, allows you to shift the location of all of the crossover frequencies anywhere across the audio spectrum. You are not tied into the fixed frequency associated with the legacy mode of operation for the Punch and Crunch effect. Because the frequency ranges are user definable, they can be placed quite close together because the system also employs symmetrical third order slopes at the various crossover frequencies, unlike the asymmetrical nature of the crossovers used in fixed frequency mode of operation. This provides better isolation between bands thus reducing adjacent band crosstalk. To place the system into a variable frequency mode of operation, merely tick off the "Variable Frequency Mode" checkbox. Then, you can drag from side to side the various crossover frequencies (the vertical lines between each of the bar graphs). And, just like in fixed frequency mode, you can vary the threshold at which each band becomes active by dragging the horizontal component of the bar graphs upwards or downwards.

Change Speed

The Change Speed Filter is designed for use in several applications:

- It can be used to correct the speed of a recording.
- It can be used for fractional speed mastering. This feature is important in any of the following three instances:
    1. You have available a 45 RPM turntable but do not own a 78 or an 80 RPM turntable and need to play those formats.
    2. You want to play 80 RPM records, but only have a 78.2 RPM turntable without variable speed.
    3. Your turntable will play all speeds, but the record you are attempting to transfer is so warped that the stylus skips off the record due to the vertical undulations of the tone arm.
- It can make speech transcription easier by slowing it down to the rate at which you can scribe. Also refer to the Stretch & Squish filter for this functionality. Note: The Change Speed tool will alter the pitch of the audio, while the Stretch and Squish Tool will lengthen or shorten an audio file without changing the pitch.
- It can be used to produce interesting special effects.
- It can be used to "tweak" a musical piece to meet a contests time requirement for a choreographed performance.



The Change Speed Effect

The following is a summary of the control parameters and the range of adjustment provided for the Change Speed filter:

- **Starting Pitch Control:** -50% to +100%
- **Ending Pitch Control:** -50% to +100%
- **Display Pitch Range (3 ranges):**
    1. +/-1%
    2. +/-10%
    3. +100 / -50%
- **Shape (Pitch vs. Time):**
  Straight Line (2 Green Cursors)
  Curved Line (4 Green Cursors)
  Constant (Start and End Track each other)
- **Graphical linear or curvilinear Pitch inflection points** (green square cursors on graph)

The Graph shows you how you have programmed the speed to change as a function of the selected .wav file time axis. You can use the mouse to drag the two green cursors to establish the time relationship that you desire. Often, a flat line is appropriate; however, sometimes the speed of the cutting lathe would slow down towards the end of the recording. The reason this occurred is that some of the early recording lathes used wind up mechanical motors with governors rather than electrical hysteretic synchronous motors. To correct this defect, a pitch decrease (negative pitch slope) is necessary towards the end of the recording. When the curve shape is selected, two additional green cursors appear. The new green cursors can be moved both vertically and horizontally allowing you to create numerous curvilinear pitch vs. time relationships. Curvilinear correction is useful for fixing the speed of non-capstan based tape recordings that have been transferred using a capstan-based machine. It can also be used to create interesting special effects when used in conjunction with the looped preview or looped play mode.

Use the following Formulae to calculate the % Pitch Change required when using fractional speed mastering:

$$\Delta \text{ Pitch } (\%) = ((R/T) - 1)(100)$$

Wherein:

T = Actual Turntable Speed (RPM)

&

R = Rated Recommended Record RPM

(see the RPM Chart in the Glossary of Terms section of this manual to determine the correct recommended Record RPM)

One method for determining the pitch change required for a particular recording is to measure the line frequency "Hum" on the signal. This can be accomplished with your Diamond Cut Spectrum Analyzer. Deviations from the ideal values (50 or 60 Hz) can be compensated using the Change Speed feature by applying simple ratio proportions based on the measured hum frequency value compared to the ideal value.

Note: The Change Speed Effect will not appear in the Effects list if no file has been opened.

Automatic Change Speed Compensator

Sometimes, it is hard to determine what the correct pitch correction factor that should be used for a particular recording when the line frequency hum of the original recording can't be discerned from other noise components. But, if you know the length of the piece, you can use the Automatic Change Speed Compensator to correct its pitch. Often, the time duration of a particular track is published in the liner notes associated with the recording. Simply do the following:

A. Make sure that the recording is trimmed carefully with no lead-in or lead out dead-time. You can always add back into the file some silence after the speed compensation process has been completed.

B. Click on "Set" and then enter the published value of time into the "Resulting Time" data entry field.

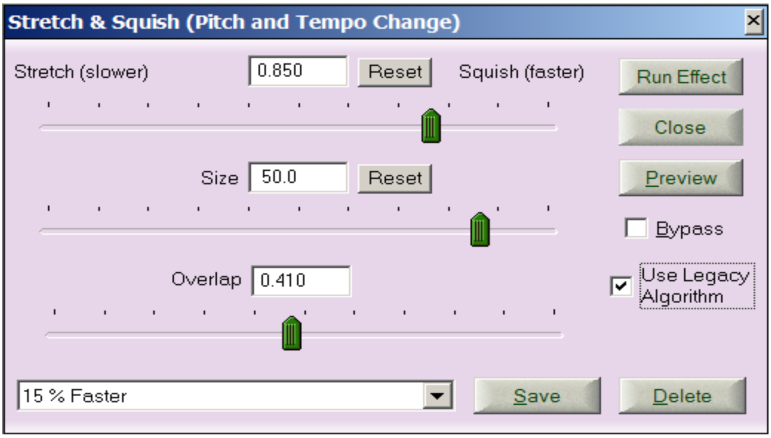
C. Run the filter and thereafter the pitch should have been corrected so long as the published time duration for the track is accurate.

This method can also be used to take a track that may be a little too short or too long for a choreographed performance and correct it per the contest rules (within reason). Another feature to consider for this type of situation would be the Diamond Cut Stretch and Squish time compression algorithm, especially if large time changes are required.

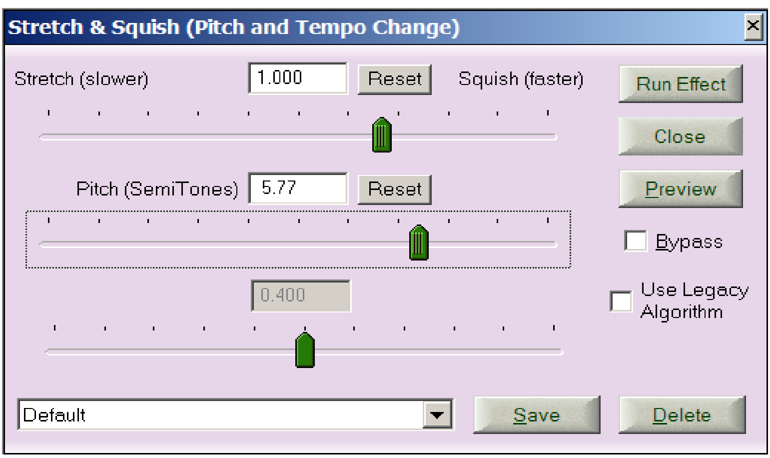
Note: The Automatic Change Speed Compensator should only be used in Straight Line Slope mode.

Time Compression and Expansion (Stretch and Squish)

The Stretch and Squish filter is a variation on the theme of the Change Speed filter. It allows you to change the cadence (tempo or speed) of a piece of audio while maintaining its pitch at a constant value. You can either slow down the beat or cadence of a piece (Stretch) or speed it up (Squish).



The Stretch and Squish Effect shown in Legacy Mode



The Stretch and Squish Effect shown in Pitch Shift Mode

Stretch & Squish is actually a special case of the speed change filter. It has the ability to maintain a relatively constant pitch while modifying the cadence or beat interval of the .wav file that is being processed. Its primary purpose is for Forensics applications in which a spoken word recording needs to be slowed down for transcription to the written word; however, it can also be used to learn a fast guitar riff by slowing it down, or for other special effect situations. It has a range of adjustability that will allow you to half the cadence in “Stretch” mode which means that it is slowed down. Also, it has the ability to double the cadence of a .wav file in “Squish” mode. Stretching or Squishing a .wav file is accomplished with the Stretch control. If you set the Stretch control to 1.000, the file will play or preview at normal speed. However, if you decrease the number to its lower limit of 0.5000, you can double the speed of the recording. If you increase the number to its upper limit of 2.000, it will now play at half speed. Also, a Size control is provided which allows you to vary the update rate of the algorithm. This control affects the overall sound quality of the filter, which is subjective in nature and left up to the user. The following controls are provided on the Stretch & Squish Filter:

- **Stretch or Squish (Ratio):**
  0.500 to 1.000 speeds up the file and represents “Squish” mode
  1.000 to 2.000 slows down the file and represents “Stretch” mode

  - **Size Control** (Legacy Algorithm only)**:** *20-240 (mSec)*
    a. 20-100 (Only useful for special effects)
    b. 100-240 (Useful for Speech and Music without significant distortion)

- **Overlap Control** (Legacy Algorithm only)**:** 0.2 to 0.6 (Determines the normalized degree of frame overlap and must be adjusted qualitatively. The best sound will typically be found between .3 and .5).

- **Reset:** Resets the corresponding slider control to its zero effect position.

Pitch Shift Mode (Semi Tones): (Range - 12 to + 12) In Pitch Mode (operable only when Legacy Mode is unchecked) you can change the pitch of a file while keeping the tempo (rhythm or cadence) constant.

Its range of adjustment is +/- 1 octave wherein +12 represents a one full octave increase in Pitch and – 12 represents a one octave decrease. The calibration is in semitones meaning that the octave above and below the source material is divided into twelfths (12 semitones in an octave). Each unit of measurement associated with the Pitch control represents one note on the chromatic musical scale. You can fine-tune the pitch of the system using the left and right arrow keys in association with the Pitch control. To do so, just click on the pitch control slider and then use the up and/or down arrow keys until you achieve the desired pitch while previewing your file. The exact semitone value being used is displayed in the digital readout located directly above the Pitch control.

The Stretch & Squish Effect includes a set of 6 “Disguised Voice Effect” presets that you can use to shroud human voices behind a sonic veil (making them unrecognizable). You should not “hot switch” between these presets while previewing them, because some of them actually change the algorithm being used to accomplish its effect. Always stop the preview process before selecting another one of these presets to assure proper algorithm initialization.
Note: This effect produces digital artifacts. Artifacts are normal for this type of mathematical manipulation when applied to a .wav file. The greater the amount of “Stretch” or “Squish” applied, the greater will be the degree of artifacts generated by the routine. Quality will decrease as you increase the amount of time compression or expansion.

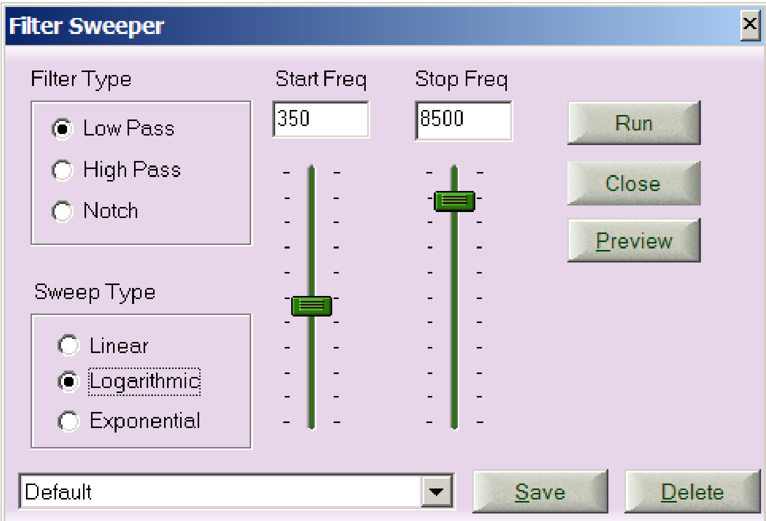
In “Pitch” mode (operable only when Legacy Mode is unchecked) you can change the pitch of a file while keeping the rhythm or cadence constant. Its range of adjustment is +/- 1 Octave wherein +12 represents an octave increase in Pitch and – 12 represents a 1 Octave decrease. The calibration is in semitones meaning that the Octave above and below is divided in twelfths (12 semitones or half-tones are in one Octave).

Note: With the musical scale of *twelve-tone equal temperament* further divides each semitone into 100 parts called cents.

Filter Sweeper

The Filter Sweeper is a novel system providing a variable frequency response versus time capability. The sweeper can be applied to an entire .wav file or any portion by using the Diamond Cut selective filtering capability. Three filter types are available in the Filter Sweeper including a sweep-able Low-Pass, High-Pass, and Notch. Besides producing interesting special effects, the Filter Sweeper has a number of very useful audio restoration applications. Here are two examples.

The hum frequency of a recording sometimes changes as a function of time. This audio defect may be due to the fact that the original analog tape recorder was battery powered and as the battery became weaker during the recording process, its speed decreased. This mechanism will have caused the resultant hum frequency to increase versus time when played back later on a constant speed system. To correct this defect, you can apply the Filter Sweeper set for operation in “Notch” mode. Set the “Start Frequency” to the appropriate lowest hum frequency value of the defective .wav file, and the set the “Stop Frequency” to the highest value of hum frequency. These two values of frequency can be determined by using the Spectrum Analyzer while using the “looped play” mode during the appropriate sections (beginning and end) of the defective .wav file. The “Sweep Type” will need to be determined empirically, because the appropriate compensation curve will depend on the type of capstan servo motor regulator that was used in the offending tape recorder in conjunction with the type of batteries that had been employed.



The Filter Sweeper

Another application of the Filter Sweeper is what we call “masked” Fade-Ins and/or “masked” Fade-Outs. These “masking” techniques not only can be used to produce very interesting effects, but also provide a very useful noise reduction tool. How can a swept filter produce noise reduction? The principle is based on a simple musical concept coupled with the non-linear nature of the human sense of hearing. Some musical material commences with very simple progressions (including only one or two musical instruments), then moving into a series of complex riffs and refrains (maybe including a few crescendos) and then often diminishing in complexity towards its ending (soft curtain). Much of the noise in such a recording is hidden by the complexity of the musical material found it the central body of the piece. The noise on the recording is heard to be most prevalent to the human ear at its beginning and its ending during the compositions simple parts. (In actual fact, the noise level is probably relatively constant throughout the entire recording.) That is why we refer to this phenomenon as “masking. This perception is a trick that the human ear plays on the listener which you can take advantage of as an audio restoration engineer. By first selecting an area at the beginning or end of the piece and then using the various Filter Sweeper presets labeled “Masked Fade-In” or “Masked Fade-Out”, you can smoothly contour the response of the restoration system during these quiet intervals at the beginning and ending of the musical rendition. If there is excessive hiss present during these time periods, use the Low-Pass Sweeping filter. If there is a high degree of low frequency noise (like rumble and or HVAC noise), use the High-Pass Sweeping filter. If there is a combination of both of these types of noises, you will have to apply two passes of the Swept filter to obtain the optimal results, running one operation with the Low-Pass and another with the High-Pass or by using the Multi-filter. Since some musical material is simple at its beginning and end, reduced frequency response will often not be observed by the listener due to the application of this new technique. The "Sweep Type" that you choose will depend on the type of musical progression that you are dealing with, and is also somewhat subjective.

The Filter Sweeper provides the following controls:

- **Filter Type:**
1. Low Pass (First Order Butterworth)
2. High Pass (First Order Butterworth)
3. Notch (Second Order with 0.1 Octave Bandwidth)

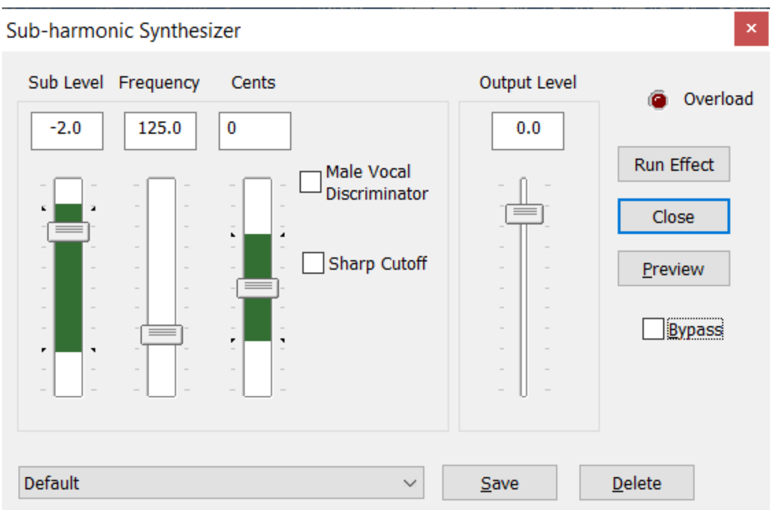
- **Sweep Type:** (Determines the shape of the Filter Sweeper's frequency versus time relationship)
1. Linear
2. Logarithmic
3. Exponential

- **Start Frequency:** (Determines the frequency value in Hz from which the sweep begins.) Range: 10 – 19,999 Hz

- **Stop Frequency:** (Determines the frequency value in Hz at which the filter's sweep terminates over the determined highlighted area of the .wav file.) Range: 10 – 19,999 Hz

Sub-harmonic Synthesizer

Sub-harmonics are fractional multiples of a fundamental frequency. Some audio media have had difficulty recording the lowest octaves of the audio spectrum and thusly are deficient in this sonic area. The Diamond Cut Sub-harmonic Synthesizer is designed to help correct this audio anomaly. It can also be used to add "deep bass" to any recording. The sub-harmonic synthesizer takes audio signals in the bass portion of the spectrum and divides those signal frequencies in half and then adds them back into the system's output mix. When the "Cents" control is set for 0, this division by 2 is exact. But, sometimes music sounds more natural when its subharmonics are not perfect halves. Thus, you can vary the "Cents" control up to +/- 200 cents (each semitone = 100 cents on a 12 tone scale of equal temperament). Higher than 0 Cent value settings will make the subharmonics sound sharper while values less than 0 will make them sound more flat.



The Sub-harmonic Synthesizer

You can choose the upper frequency limit at which this process takes place via the "Frequency" control. A good frequency to begin with is 125 Hz; adjust to your personal taste using the Preview button. You can also adjust the wet/dry mixture of the synthesized sub-harmonics with the original source signal by way of the "Level" control. As an

example, if the sub-harmonic synthesizer frequency control is set to 125 Hz, frequencies are generated in the range of 62.5 Hz on down to the bottom end of the audio spectrum. These signals are then added back into the main signal pathway thereby re-constituting bass (or deep bass) notes which may be missing from a recording. With no checkboxes checked, the transition of the system below the frequency setting is gradual and often works effectively. Bass male vocals can "bleed" into the sub-harmonics sometimes. If that occurs, try checking the Male Vocal Discriminator checkbox which should improve the sound quality. If the male voice still bleeds through, lower the frequency control and/or check the Sharp Cutoff Checkbox instead of using the Male Vocal Discriminator.
The Sub-harmonic Synthesizer includes the following user controls:

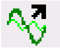
- Frequency: Range is 60 Hz to 300 Hz – This controls those frequencies below which are used to excite sub-harmonics. It is the "corner frequency" setting.
- Sub Level: Range is -60 dB to + 15 dB – This controls the amount of the effect applied to the output of the effect.
- Output Level: Range is – 60 dB to +10 dB – This controls the overall output level of the effect which includes both the wet and dry signals. Keep this control set so that the "Clip" LED does not illuminate.
- Cents: 200 (sharp) to -200 (flat). Creates imperfect subharmonics (often more musically pleasing than perfect divisible values)
- Male Vocal Discriminator Checkbox: This reduces the propensity of the subharmonic synthesizer from creating subharmonics from bass male vocals.
- Sharp Cutoff Checkbox: This feature creates a very steep slope below the frequency setting providing a very high degree of discrimination of lower frequencies being applied to the synthesizer. This feature can often decrease low frequency artifact generation by the synthesizer at the expense of a slightly lower level of smoothness of the frequency crossover transition of the system.

Note 1: For "Ultra Deep" bass effects, you can cascade two sub-harmonic synthesizers together in the Multifilter. Set the second one in the chain to half the frequency value of the first. For example, set the first one to 150 Hz and the second one to 75 Hz.

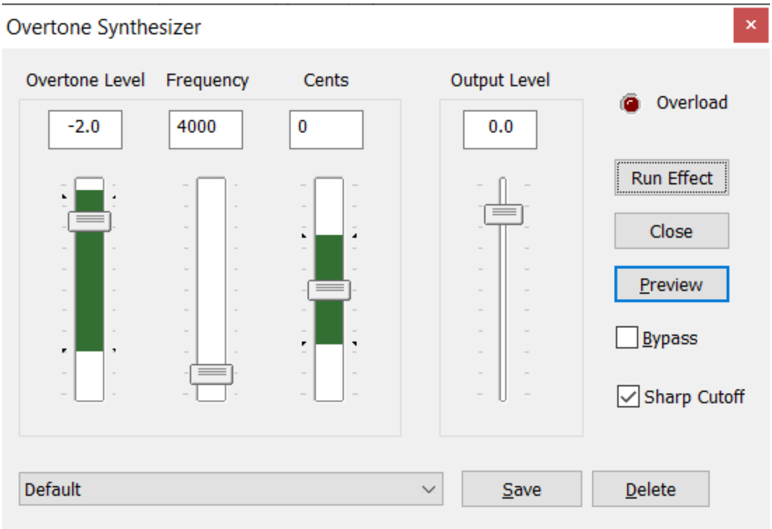
Note 2: Sub-harmonic distortion can be introduced by this effect on male vocals, especially when the frequency control is set to values greater than 100 Hz.

Note 3: It is easy to over-apply this effect thereby creating an un-natural sound. Keep its "Sub-Level" control as low as possible to add in a pleasing effect, but not so much as to create an un-natural effect.

Overtone Synthesizer



Overtones are multiples of a fundamental frequency. Some audio media have difficulty recording the upper octaves of the audio spectrum and thus are deficient in this sonic area. The Overtone Synthesizer effect is designed to help correct this type of audio anomaly. The Diamond Cut Overtone Synthesizer operates in the upper portion of the audio spectrum and is the compliment to the Sub-harmonic synthesizer. It can be used to add treble or "brilliance" to any recording. It can also be used to enhance the hissy – sibilant sounds sometimes missing from highly muffled forensics recordings. The Overtone Synthesizer takes audio signals in the treble portion of the spectrum and multiplies those frequencies times two and then adds them back into the systems output mix. When the "Cents" control is set for 0, this multiplication is exact. But, sometimes music sounds more natural when its overtones are not perfect doubles. Thus, you can vary the "Cents" control up to +/- 200 cents (each semitone = 100 cents on a 12 tone scale of equal temperament). Higher than 0 Cent value settings will make the overtones sound sharper while values less than 0 will make them sound more flat.



The Overtone Synthesizer

You can choose the lowest frequency limit at which this process takes place via the "Frequency" control. A good frequency to start with is 5,000 Hz; adjust to your personal taste. You can also adjust the wet/dry mixture of the synthesized overtones with the original source signal by way of the "Overtone Level" control. As an example, if the Overtone Synthesizer frequency control is set to 4,500 Hz, frequencies are generated in the range of 9,000 Hz on up to the top end of the audio spectrum. These signals are then added back into the main signal pathway thereby re-constituting treble (or brilliance) which may be missing from a recording.

The Overtone Synthesizer includes the following user controls:

- Frequency: Range is 3,500 Hz to 9,000 Hz – This sets those frequencies above which are used to excite overtone generation. It is the "corner frequency" setting.

Forensics Note: The frequency range for this control in the Forensics version is wider and spans from 2,000 Hz to 9,000 Hz.

- Overtone Level: Range is -60 dB to + 10 dB – This sets the amount of the effect applied to the output.
- Output Level: Range is – 60 dB to +10 dB – This sets the overall output level of the effect, including both the basic signal as well as the overtones. Keep this control set such that the “Clip” LED does not illuminate.
- Sharp Cutoff Checkbox: This feature creates a very steep slope above the frequency setting providing a very high degree of discrimination of the higher frequencies being applied to the synthesizer. This feature can often decrease high frequency artifact generation by the synthesizer at the expense of a slightly lower level of smoothness of the frequency crossover transition of the system. In musical terms, it tends to add more “sweetness” to the top end when sharp cutoff is invoked. It is also useful in Forensics application by improving the sibilant sounds of the human voice without adding much distortion to the signal.

Note 1: For “Ultra Brilliant” treble effects, you can cascade two Overtone Synthesizers together in the Multifilter. Set the second one in the chain to half the frequency value of the first. For example, set the first one to 4,000 Hz and the second one to 8,000 Hz.

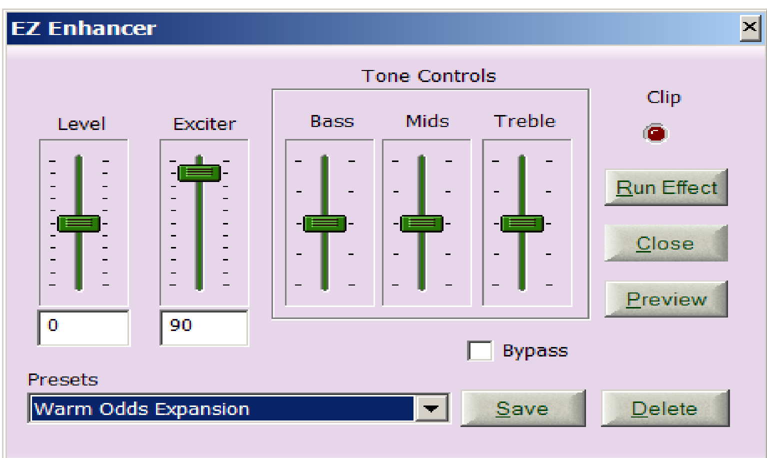
Note 2: Muffled Forensics recordings can be enhanced via the “Forensics Pseudo Sibilant (clarifier)” preset. Adjust the “Overtone Level” for the optimal result. Also, please refer to Note 3.

Note 3: The Overtone Synthesizer is limited for use on files that were sampled at 22.05 kHz and above. This fact is especially important in Forensics Audio applications which use lower sampling rates. If you desire to use it on files that were mastered on files that were recorded below 22.05 kHz, up-sample them to 44.1 kHz first and then apply the Overtone Synthesizer.

Note 4: Use of this effect in moderation is recommended in order to achieve the most natural sonic simulation of the upper end of the audio spectrum.

EZ Enhancer™

The EZ Enhancer provides an easy-to-use method for adding some spice to your recordings. It combines three types of audio enhancement algorithms into one system including harmonic generation, dynamic processing (compression or expansion) and tonal contouring. A large number of factory presets are provided which produce combinations of various harmonic generation types coupled with dynamics signal processing. The names of these various presets are descriptive and so should make choosing the correct one for a given job quite intuitive. The presets work in conjunction with the "Exciter" and the input "Level" controls. Three tone controls are also provided including Bass, Mids and Treble. The system is capable of remembering the state of the EZ Enhancer system controls based on your own preferences via the "Save" feature. Here is a description of the EZ Enhancers controls:



The EZ Enhancer

Level Control: -100 to 0 to +100 wherein 0 represents no gain added on the input side of the dynamics processor sub-system. Since the dynamics processing elements of the various presets are non-linear in nature, you will find that there is an interaction between the Exciter and the Level Controls in terms of sound tonality. Adjust both for the most pleasing result. The level control does not directly affect the overall degree of the system, but only as a higher order effect. 0 is the default setting for this control and in most cases will produce satisfactory results without any "tweaking".

Exciter Control: This control affects the degree to which the harmonics generator and the dynamics processor output signals are applied to the source signal. The range is 0 to 100 wherein 0 represents no enhancement (dry) and 100 represents the maximum enhancement (wet).

Bass Control: This is a shelving type of control having its corner frequency fixed at 175 Hz and having an amplitude adjustability range of +/- 15 dB (15 dB of boost or cut).

Mids Control: This is a Bandpass (peaking) type of filter having its center frequency fixed at 900 Hz and having a bandwidth set for 3 octaves. It can produce up to +/- 15 dB (15 dB of boost or cut).

Treble Control: Like the Bass Control, this is also of the shelving variety having its corner frequency internally set to 3,000 Hz and having an amplitude adjustability range of +/- 15 dB (15 dB of boost or cut).

Clip Indicator: A red clip indicator is provided to show if the dynamic range of the EZ Enhancer system is being exceeded. If this indicator illuminates, then you must determine which control is set too aggressively and lower it until the indicator extinguishes. Otherwise, clipping type distortion will occur, which is not specifically very pleasant sounding.

Note: The best results will be realized by applying the "Normalize Gain Scaling" feature (found under the CD Prep Menu) to – 6 dB before running the EZ Enhancer.

Dynamic Bass Processor (Dyna Bass Processor)

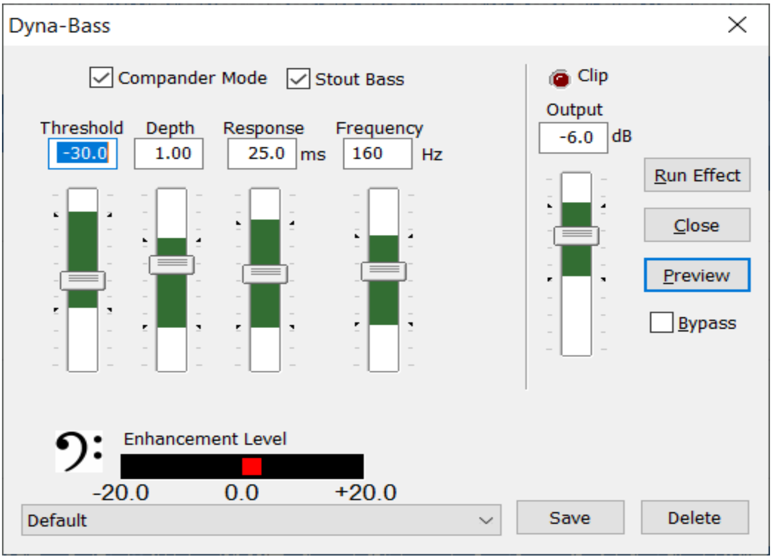
The Diamond Cut Productions Dyna-Bass Processor can be used either as a dynamic low frequency filter or as a bass dynamic enhancement system. Normally, bass frequencies can easily be accentuated or attenuated by using any of the various graphic or paragraphic equalizers. However, there are cases where low frequency random noise signals will also be amplified by those filter types when attempting to boost the bass or attenuate the low frequency noise. One such situation occurs when an audio source may contain a large degree of rumble from a turntable or from the master recording itself. Also, live recordings (especially those made with large diaphragm microphones) can pick up the random rumbling low frequency sounds of the HVAC (Heating, Ventilating and Air Conditioning) system which will be especially pronounced during quiet musical passages. Other low frequency sounds can get onto a recording from road noise, heavy machinery and general recording venue building noise. Oddly, these low HVAC frequency sounds are often not noticed during the recording process itself, but become apparent and dominant during playback after the fact. Using something like a hi-pass filter or equalizer can dramatically reduce these sounds or seismic events from the final recording at the expense of a loss of the legitimate low frequencies of the performance or source recording.

The Dyna-Bass processor can be used to attenuate low frequency noises (1/f noise) while maintaining the desired low frequencies on the recording. In other words, this processor can separate random noises from the actual important bass sounds. In this case, it can be viewed as a low frequency noise reduction filter, but because it is dynamic, it incrementally amplifies the low frequency bass signals when they fall above a certain user settable threshold value. Above the threshold value, both the random1/f noise plus the target good bass signal are passed to the output, with the good bass masking the noise signal. This process is provided by upward expansion of signals that exist above the user set threshold value.

The Dyna-Bass processor can also be used to add “punch” to a bass line. This is accomplished by expanding the dynamic range of the signals below the frequency control setting when their amplitude exceeds the threshold control setting. Generally, this should be performed using the non Compander mode of operation (compander checkbox unchecked). This can create a substantial noise reduced bass sound in some situations providing you with an expanded bass dynamic range and noise reduction at the same time. It should be noted that this effect can easily be over-done, so care should be used when adding “punch” to an audio restoration/enhancement using expansion alone. Alternatively, a less aggressive effect can be obtained with the Compander mode turned on, which can be quite pleasing to the ear and easier to “tame”. This mode upwardly expands the signal below the threshold and compresses it above the threshold setting. However, the greatest noise reduction effect is obtained in simple Expander mode (ALC box unchecked), but you need to be careful with the settings so as to avoid output overload. It is recommended that the “Output” control be set to a fairly low position when starting out and then bring it up slowly after you find the basic settings of the primary controls that you find to be the most effective on your audio material. Additionally, the various control sliders all have “Green Zones” which show you (depending upon the chosen mode) which ranges (zones) will most likely give you reasonable results. This does not mean that you can’t operate outside of those zones, but the highest probability of success will be found when the various controls are set within their green zone ranges. A “Stout” mode is included (invoked via a checkbox) which intentionally adds ripple to the bass signal giving it a more robust (or “colored” sound on some material. It is important to note that the best results will be obtained when previewing using an audio system that includes a sub-woofer when using the dyna bass processor.

The Dyna Bass Dialog Box

A good place to start is to use one of the factory provided descriptive presets. Click on the preset that seems to describe what you want to accomplish and then fine-tune it using the slider controls. After you are satisfied with the result, save that preset for your own future reference.



The Dynamic Bass Processor Dialog Box

The Dyna Bass Processor Controls

Here are the various Dyna Bass Processor controls and what they do.

Threshold (in dB): This control determines what amplitude value at which the system becomes active. Its range is from 0 dB to -60 dB. Use it in conjunction with the bar graph and your hearing to find the active region for the processor.

Depth (relative units): This determines the degree of the effect imparted upon the audio signal. It ranges from 0.1 (minimum effect) to 4.99 (max effect). Use the Green Zones adjacent to the slider as a guide to obtain reasonable results.

Response (mSeconds): This control sets up the time constant for the system to react to signal envelope changes. It ranges from 5.0 mSec to 99.9 mSec. Small values produce fast transient response while high values produce slower responses.

Frequency (in Hertz): The range for the frequency control is 50 to 250 Hz. It determines the frequencies below which activate the bass dynamic processor. Lower frequency values yield a deeper bass sound while high frequencies yield a more nominal bass sound.

Output (relative dB units): The range for this control is +5.9 dB to -30 dB. This controls the overall output level of the dynamic bass processor. Since the processor tends to amplify the bass signal, it is a good idea to start with this control set to its lowest value. Use it in conjunction with the Red Clip Led to assure that no clipping occurs. If clipping is noted, lower the Output control down to a point where it is no longer flickering on.

Compander Mode: This selector box changes the mode of the processor from "Expander Only" to ALC/Expander (**A**utomatic **L**evel **C**ontrol). This mode provides you with a more "tame" behavior of the system, reducing extreme applications of the Dyna Bass effect.

Enhancement Level Bar Graph Indicator: The horizontal bar graph shows excursions when the effect is being invoked based on user settings. The bar graph moves in both directions depending on whether the system is expanding or compressing the audio. Its range runs from -20 dB to +20 dB.

Stout (mode): When checked, this invokes a different signal pathway through the processor which intentionally adds aberrations to the frequency response of the system. The ripple added to the bass signal creates a more robust sound on some material. There are presets that contain this mode for experimentation.

Clip LED Indicator: This red clip LED is located above the Output control. When it lights, it indicates that they system is overloading and distorting. The Output control should be lowered to a point where this light no longer flickers on and off with the audio - - - it should always remain off for the best audio performance of the processor.

Preview Button: This allows you to hear what the processor will be doing to the signal without having to commit to a change in the source file. It operates in real time and allows you to experiment to find the best results for the material you are working with.

Bypass Button: This button allows you to compare the processed signal against the raw signal to help you decide the degree of effect that will provide the proper result when you finally "Run" the effect.

Progress/Cancel Dialog Box: When Previewing (or Running), a dialog box will appear in the lower left of your display. The bar graph shows the progress the processor is having on the file. When you are done previewing, you can cancel the preview via that feature by clicking on "cancel". When you ultimately decide to "Run" the processor, the bar graph will show how the system is progressing through the file.

Run Effect: After you establish the proper settings via the Preview function, click on "Run Effect" and your file will be imparted with the results of the Dynamic Bass Processor.

Close Button: When you are done with the Dynamic Bass Processor, clicking on the "close" button will shut down the processor dialog box so you can proceed to other processes.

Presets Bar: This is found along the bottom of the dialog box and contains numerous descriptive presets to help you get started with an audio problem using the Dynamic Bass Processor.

Save: To the right of the Presets Bar is a "Save" button. When you create your own desirable parameters, you can save them using a descriptive name for future reference.

Delete: To the right of the "Save" button is a "Delete" button. To delete a preset, bring it up and then click this button to eliminate it from the presets menu.

Note: A Dynamic Bass demo file is included so that you can experiment with this Processor. The demo file includes low frequency rumble as well as a simple bass line. A preset is available to clean up this file, but you can experiment with it to your own taste. The file is True Blue Lou, an electrically recorded lateral 78 mastered in 1929 and performed by Annette Hanaahaw.

The Forensics Menu

The Forensics Toolbar

Surveillance, two-way radio, noisy telephone communications and black-box analysis place special burdens on noise reduction software. The DC Forensics10 Audio Laboratory comes complete with a special set of filters and measurement tools aimed at Forensics applications that can be found under the Forensics Menu. Besides being useful in Forensics applications, these filters can be used for a variety of non-Forensics related material. The DC Forensics Audio Laboratory version has a more extensive and higher performance feature set in this regard compared to its more commercially driven DCArt10 counterpart. Certain Forensics filters require more experience to use compared to the others. They are placed under a special category within the Forensics suite called “Advanced Filters” and include the Time Domain Adaptive Filter (TDAF), the Dynamic Spectral Subtraction filter (DSS), the Polynomial Filter and the Spectral Filter. These more advanced filters can all found as a sub-group under the following icon:

Click on the down arrow on the right side of the icon to expand the menu.

In General, noise encountered in Forensics applications can be divided into several basic categories:

- **Out-of-band noise**:
  This type of noise is rejected with the brick-wall band-pass filter.

- **In-band repetitive noise at a level lower than the target signal:**
  This type of noise is rejected with the brick-wall band-stop

filter.

- **In-band random noise at a level lower than the target signal:**

The standard Continuous Noise filter found under the Filter menu rejects this type of noise.

- **In-band repetitive noise at a level equal to the target signal:**

The Adaptive filter set up for the Keep Residue mode of operation can attenuate this type of noise. If this noise is in the form of a "buzz" you should also consider trying the Harmonic Reject filter found under the Filter menu.

- **In-band random noise at a level equal to the target signal:**

This type of noise can be reduced by using the Adaptive filter set up for the Normal (random) mode of operation or the DSS.

- **Subsonic noise which can't be heard, but are important events:**

This type of noise can be visualized using the Subsonic Explorer function and is time correlated to the primary source signal.

Several of the Forensics filters can also be used in those extreme circumstances such as when the noise level is significantly greater than the "good" signal level. In this instance, the ability to hear and understand the underlying speech is typically the goal – not a complete restoration. See the Notes below.

<u>Note 1:</u>
It is important to note that the Forensics filters are not optimized specifically for "high fidelity" applications, but more for improving the intelligibility of speech or the discernment of subtle sounds buried in noise.
<u>Note 2:</u>
The Diamond Cut Forensics Filters perform best on files that are sampled at 44.1 kHz or higher. If your native file is encoded with a lower sampling rate, up-convert it first before using the Diamond Cut

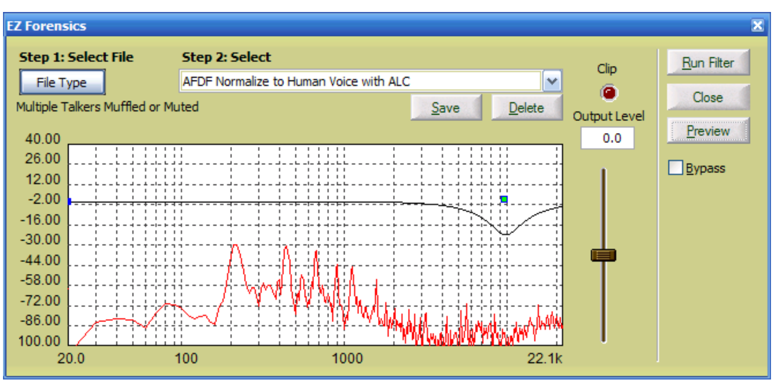
Forensics Filters. While this will not improve the bandwidth of the signal, it will speed up filter processing time.

EZ Forensics Filter

(Forensics Version Only)

The EZ Forensics Filter™ is a very good place to start when working with any Forensics audio file It is very easy to use and often will provide a good solution to your noise problem, even if not 100 percent optimal. It combines a dual question wizard scheme coupled with a high degree of automation, adaptive filtering techniques and 170 pre-programmed filter solutions. In some situations, the EZ Filter process will help you to hone-in on the best solution to a Forensics Audio problem. It will present you with a few filters to choose from after you answer a few questions. By previewing your file and comparing the alternative filters presented, you can then choose the best filter for the job at hand.

A graphical display shows you the amplitude vs. frequency of the input signal to the filter. The black line shows the user modified output response on the output side of the EZ Forensics filter. The adjustable blue inflection point (square dot which is adjustable with your mouse) is the user interface for adjusting the output response of the system after the best filter has been selected.



The EZ Forensics Filter

The basic process for using the EZ Forensics Filter is as follows:

1. Answer the First Question found in "Step 1: Select File Type". Click on the "File Type" button and a "Recording Type" Dialog Box will appear. Choose (by left mouse clicking) between the alternatives based on the nature of your audio source including:

- Single Talker
- Multiple Talkers
- Telephone or 2-Way Radio Communications
- Background Sounds
- Binaural (2 Mic) Decoding

2. Click on "Next" in the "Recording Type" Dialog Box and then answer the second question presented in the "Recording Problems" Dialog Box. You can choose between:

- Something Else Not Listed Here
- Hissy, Sibilant High Frequency Noise
- Low Frequency Rumbling Noise
- Muffled or Muted Sound
- Power Line Hum
- Buzz
- Steady Tones
- Variable or Swept Frequency Tones

3. Under "Step 2:" you will notice that a number of filters are presented to you. Choose the best one by "Previewing" each one and decide on which one does the best job.

4. You can "tweak" the response of the system by left-moussing the blue inflection point on the graphic display to provide emphasis or attenuation anywhere on the frequency spectrum that you desire. Moving the mouse horizontally changes its frequency while moving the mouse vertically amplifies or attenuates the signal at that frequency. You can also adjust the "Output Level" with the slider control having the same name. Be sure that you do not "clip" the output to avoid signal distortion. A clipping condition is indicated by the illumination of the red "Clip" virtual LED indicator.

5. If and when you are satisfied with the results, then highlight the file or a portion thereof and click on the “Run Filter” button.

Now, Wasn’t That Easy?

A description of the filter type being used is annunciated just below “File Type” button. If the EZ Forensics Filter did not provide the optimal result, the DC Forensics10 Audio Laboratory provides you with a wide array of alternative filters that you can customize for a specific Forensics Audio application. They can be used individually (one at a time) or in concert with one another via the Diamond Cut Multifilter.

Advanced EZ Forensics Filters

To “Go Advanced” with the EZ Forensics approach, a good place to start is with the following two Multifilter Presets (found under the Multifilter and not under the EZ Forensics Filter):

EZ Forensics_Proto1 (A Time Domain Adaptive Filter Based Approach)

EZ Forensics_Proto2 (An Adaptive Frequency Domain Filter based Approach)

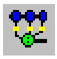
These two Multifilter presets provide you with the basic signal path structure used in the EZ Forensics system. But, in the Multifilter, you have complete control over a myriad of parameters which you can customize and tweak to your specific needs. Consider using this approach in the event that the EZ Forensics Filter is not providing the optimal results that you seek.

Note 1: It is important to convert all forensics .wav files (8 kHz, 11.025 kHz and 22.05 kHz) up to a 44.1 kHz sampling rate before using the EZ Forensics Filter. To accomplish this, use the Change Sample Rate feature found under the Edit menu.

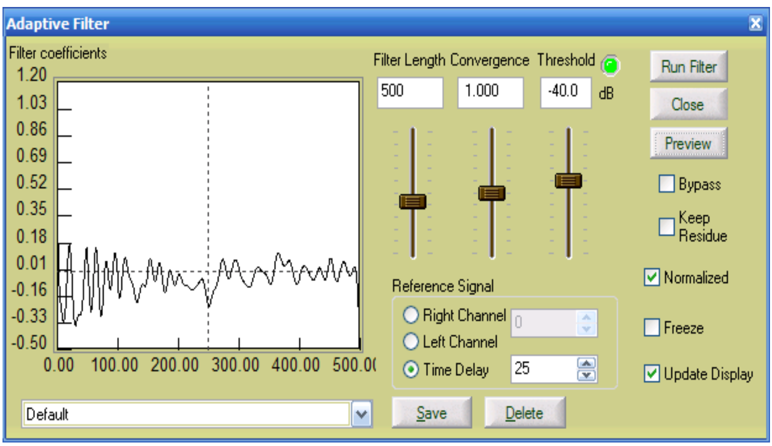
Note 2: The best results will be realized by applying the “Normalize Gain Scaling” feature (found under the CD Prep Menu) to – 6 dB before running the EZ Forensics Filter.

The Time Domain Adaptive Filter (TDAF)

(Forensics Version Only – Advanced Filters)



This **T**ime **D**omain **A**daptive **F**ilter (TDAF) is used largely in audio applications where the ambient noise environment is constantly changing and the filter coefficients must automatically adapt to maintain good intelligibility of an audio signal. This is the Time Domain based "older sister" to the forensics **A**daptive **F**requency **D**omain **F**ilter (Forensics AFDF) found inside the Diamond Cut Continuous Noise Filter (one of the CNF Modes).



The Time Domain Adaptive Filter

The Adaptive filter adjusts itself to remove a modeled signal representing the unwanted time domain waveform while preserving the target signal. It uses an advanced form of an adaptive least mean squared algorithm to provide continuous adaption of the filter coefficients. It works best with a reference signal (in other words, a stereo or binaural source) containing only the noise to be rejected. This second channel or reference signal can be obtained from a second surveillance track with its microphone located near the noise source in the room such as a Jukebox or a television set. However, it can also use its own signal as a reference in conjunction with the time delay function, which is provided for monophonic situations. Additionally, the adaptive filter provides either the main processed signal or a keep-residue mode output signal for rejecting a wide variety of different types of noise sources. Sometimes it will be found that the "keep residue" mode signal is more useful than the main output signal. Trial and error is sometimes the best way to determine the best mode to use. The following controls are included with the adaptive filter:

- **Convergence (Adaptation Speed)** - Slower adaptation speeds produce better noise rejection for stationary noise (not changing), while faster adaptation speeds produce better adaptation response in quickly varying ambient noise conditions.

- **Filter Length (Samples)** - The larger this number the more signal inflection points can be modeled in the time domain signal in order to be rejected or maintained. Generally speaking, a "sweet spot" is often found between 10 and 100 samples. Very long filter samples can be very demanding upon your processor's resources.

- **Reference Signal** - This is the signal to be compared to when a stereo recording is available. (Choose the one that contains the reference signal to be used.)
    1. Right Channel
    2. Left Channel
    3. Time Delay (when there is no reference signal)

- **Time Delay (Samples)** - For use in time delay reference mode only when no reference signal is available. This essentially allows a delayed representation of the signal being adapted to be its own reference.

- **Adapt / Freeze button** - Selects adapt or freeze coefficients mode of operation. Usually this is activated in Adapt mode for the first 5 seconds of noise. In some cases where the sound ambient is constantly changing, one may choose not to "freeze" this filter.

- **Keep Residue button** - Allows the operator to use the error signal rather than the output signal from the

Adaptive filter. This feature is useful for attenuating continuous loud or varying tones (like a siren) that may be masking a Forensics recording.

- **Threshold** - This control sets the level above which the system re-initializes itself based on the applied signal amplitude. Generally, you will find that settings somewhere between –20 dB and –40 dB are useful values. If this control is set to 0 dB, it will not produce any output because it will be continuously re-initializing itself.

- **Threshold LED Indicator -** Just to the right of the threshold control is a Green LED indicator. When this indicator is illuminated, the adaptive filter is active and adapting to the signal that is present. If the LED is not illuminated, move the Threshold Control downwards until the Green LED flashes or illuminates in order for proper filter operation to occur.

- **Graphical Display** - You can choose between two graphical display modes. One mode displays the Amplitude of the Filter Coefficients on the vertical axis (on a normalized basis) vs. the Index of the Filter Coefficients (which is an indicator of how many filter taps or filter length which are being used). The second mode plots the frequency response (relative amplitude vs. frequency) being produced by the adaptive filter on a time-updated basis.

- **Adaptive Normalize** - This button, when checked, places the system into a "Normalized Least Mean Squared" mode, which is useful when attempting to clean up material with widely varying signal amplitude components. This is the preferred default mode of operation.

Important Note:
Because of the wide range of convergence values allowed in the Adaptive Filter, certain audio signals may cause the filter to become unstable and cut out. In these cases, try changing the Convergence

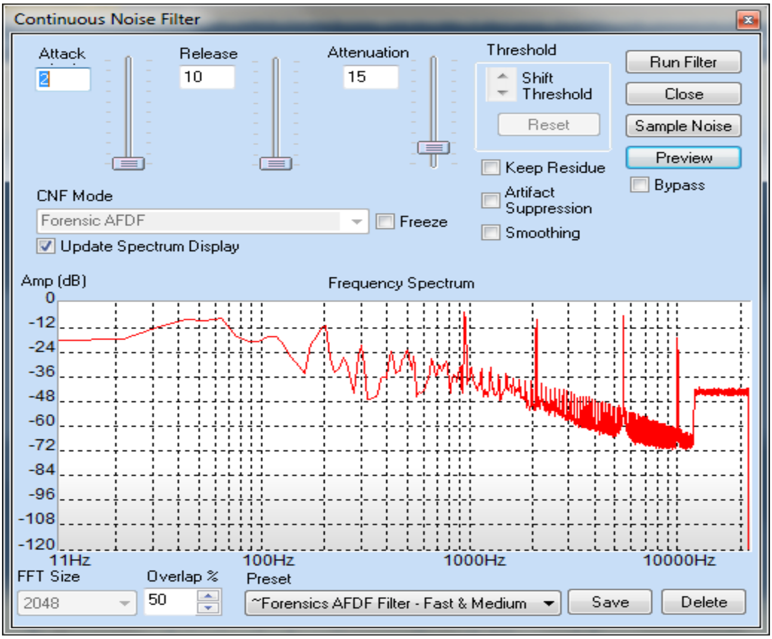
parameter until the audio is restored. Usually lower convergence settings are more stable, but adapt more slowly to changes in the audio signal. Lengthening the filter (Samples) may also increase its stability.

The Adaptive Frequency Domain Filter (AFDF)

(Forensics Version Only)

The **A**daptive **F**requency **D**omain **F**ilter (or AFDF) is the cousin to the Time Domain Adaptive Filter (or TDAF). It uses FFT (Fast Fourier Transform) techniques in conjunction with an adaptive algorithm based on statistical models of noise versus intelligent signals to create a dynamic fingerprint of the noise. When a reference channel is not available, it will often produce better adaptive noise reduction compared to the TDAF. The overall filtering effect can be reduced or increased to varying degrees by the user. This filter is actually a subset CNF Mode of the Continuous Noise Filter, but is directly accessible by the user via the icon. It includes 10 descriptive presets which can be used to get you into the general parametric area when dealing with a noisy forensics audio file. Fine tuning can then be made by the attenuation control in conjunction with the attack and release time constant controls. The frequency resolution of the system can be increased or decreased via the FFT size parameter. Larger FFT sizes result in smaller frequency bins and thus greater frequency discrimination with the tradeoff of greater levels of digital artifact production and poorer time domain transient response.

Note: A good FFT size to use as a starting point is 2048 when working with 44.1 kHz sampled files.



The AFDF in Action

- **Attack Time**

This is the time required for any of the filters to "open up" on the leading edge of a signal that exceeds the threshold line on the spectral graph.

This represents the time constant normalized value at 1 kHz. The time constant for filter frequencies operating above 1 kHz will be shorter than the setting, and the time constant for filter frequencies operating below 1 kHz will be longer.

The Attack time constant value is weighted with a -1 slope across the audio spectrum. Small values of attack provide excellent transient response, while long values provide a minimization of digital artifacts produced by the system. A good value to start with for the Attack Time parameter when using the AFDF is around 5 Milliseconds. The total range of adjustment for Attack time is 1.0 to 300 milliseconds. Smaller settings will improve leading-edge conversation response but allow more digital artifacts to pass through. Larger values will decrease the leading-edge response, but will reduce the production of artifacts during the noise reduction process. The user will need to balance between these two conflicting factors to obtain the best overall result, which will be found to be very source file dependent.

- **Release Time**

This is the time allowed for any of the filters to "close down" or "decay" following a signal that falls below the blue threshold line on the spectral graph. All remaining characteristics of the Release time Constant are the same in nature as the Attack time Constant. A good value to start with for the Release Time parameter for the AFDF is around 25 Milliseconds. If there is too much fast filter "breathing" (a digital artifact that is also sometimes referred to as “pumping”), lengthen this time until you are satisfied with the result. The total range of adjustment for Release time is 1.0 to 1000 milliseconds.

- **Attenuation**

This control sets the degree of attenuation for signals that are present and below the dynamic blue threshold line. The greater that one sets the Attenuation control, the greater will be the degree of noise reduction. However, the greater the degree of noise reduction achieved, the greater will be the loss of the sense of "Ambiance" on the resultant restoration. So you must use careful judgment as to the correct tradeoff between noise reduction and ambiance for the material you are dealing with. A good value for the Attenuation parameter is around 10 to 15 dB as a starting range. If there is too much loss in signal ambiance, decrease this value. If you desire more noise reduction, increase this value. If you increase the attenuation too much, you will begin to introduce some digital aliasing artifacts into the Destination .wav file. The total range of adjustment for Attenuation is 0 to 100 dB.

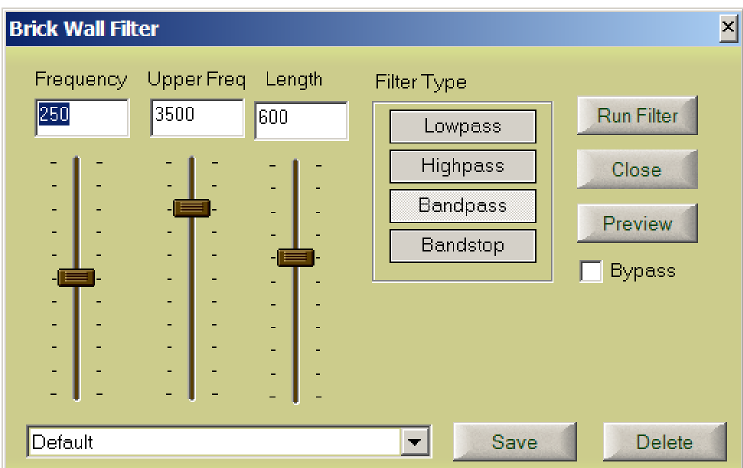
- **Threshold (Blue Graphical Threshold Line)**

This feature graphically and dynamically displays how the filter is adapting in real time to the signal being applied to the AFDF. The horizontal axis represents frequency and the vertical axis represents the varying threshold line in dB.

You should find the AFDF factory presets to be a good way to start off when using this adaptive filter. Their names are descriptive in terms of response and degree of aggressiveness. After you find a preset that is close to producing the desired results, you can go back and tweak the attenuation, attack, and release parameters optimally.

Brick Wall Filter

These filters differ from their "Hi-Fi" counterparts in that they are very steep digital filters used to attenuate signals that are interfering with a poor quality audio signal. These are FIR (Finite Impulse Response) filters that exhibit a very high degree of out of band attenuation. Also, they exhibit a phase-linear response with respect to frequency (all frequencies are delayed by the same constant amount of time). These FIR based filters are predominantly used for forensics audio applications.



The Brick Wall Filter

The Brick Wall Filters include the following choices of filter shapes:

1. **Lowpass:** Only allows signals below the corner frequency to be fed to its output.
2. **Highpass:** Only allows signals above its corner frequency to be passed to its output.

3. **Bandpass**: Allows only the signals between its chosen lower frequency and its upper frequency setting to be passed to its output.
4. **Bandstop:** Rejects all signals between its selected “Frequency” and “Upper Frequency” limits.

Several controls are provided:

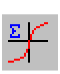
- **Frequency -** (Hertz) (Range is from 3 Hz to 20,000 Hz)
- **Length -** (Samples up to 4,094) - The larger the value of this parameter, the greater the degree of rejection past the filters corner frequency setting. However, the higher the setting for length, the larger will be the computational demands on your computer. So, large values of "length" will take longer to process or may cause your machine "stutter" in preview or live real time modes.
- **Upper Frequency Control** - Sets the upper limit for the filter's bandwidth when operating in either Bandpass or Bandstop mode. (Range is from 3 Hz to 20,000 Hz)*
- **Frequency Control (Lower)** - Sets the lower limit for the filter's bandwidth when operating in either Bandpass or Bandstop mode. (Range is from 3 Hz to 20,000 Hz)*

You can monitor the signal being removed by the brick wall filter by previewing its reciprocal filter. In other words, if you are using the Bandpass filter, switch the system to the Bandstop filter to hear what is being removed. Similarly, if you are using the Lowpass filter at a certain frequency, switch to the Highpass filter at the same frequency to hear what is being removed. The same logic applies to the Highpass filter; just use the Lowpass filter set for the same frequency to hear what is being removed.

*<u>**Note**</u> - Only applies to the Bandpass and the Bandstop filters.

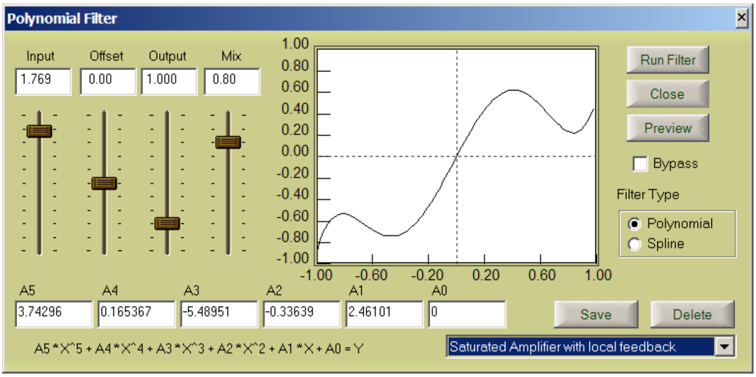
Polynomial Filter

(Forensics Version Only – Advanced Filters)



This allows mathematicians, scientists, and engineers to create their own transfer function using a polynomial expression. For those not so inclined, there is a plentiful assortment of presets to choose from. Also, there are two methods of data entry possible, either numerical or graphical.

This system realizes a transfer function in terms of the input to output signal ratios. It is useful for canceling the non-linearity's that may have been introduced during a recording process. Essentially, this feature can be useful in some circumstances for reducing harmonic distortion that was created by the recording process non-linearity. The method of data entry for the Transfer Function is in the form of the coefficients associated with a 5th order polynomial for which the actual transfer function graph is plotted automatically. You can enter the coefficients numerically for each term or you can use one of the numerous presets that are provided to get you started with a particular function. It is useful to have some mathematical background in order to effectively use this filter using coefficient data entry. However, trial and error is the best method for finding a setting that will reduce the distortion of a particular recording since the recording non-linearity is not generally known ahead of time. Furthermore, sometimes adding non-linearity's can be useful to enhance the intelligibility of extremely muffled conversations. Limited instantaneous dynamic expansion and compression can also be realized by using this feature, for which there are several presets provided for convenience. Interesting distortion creation and frequency multiplication is also possible with this algorithm.



The Polynomial Filter

1. The Transfer Function is given in the general form by the following polynomial expression:

   Y = A5X^5 + A4X^4 + A3X^3 + A2X^2 + A1X + A0

   You have 6 fields in which you can enter the coefficients of the polynomial equation, including:

   A5 =_____ A4 = _____ A3 =_____ A2 =_____
   A1 =_____ A0 =_____

   Note: Values of A can be positive or negative.

- **Input Gain:**

  Range: 0.001 to 1.999 (1.000 represents unity gain.) This controls the level of the signal being applied to the input of the polynomial expression.

- **DC Offset :**

  Range: -1.00 to 0 to +1.00 (0 represents no DC offset value.)
  This control effects the DC offset applied to the input signal to the equation.

- **Output Gain:**

  Range: 0.001 to 5.000 (1.000 represents unity gain.) This control effects the output level of this system after the polynomial equation has been applied to the signal.

- **Mix:**

  Range: 0.000 to 1.000
  This control effects the degree to which the processed signal is added to the input signal for presentation to the systems output. Zero (0) represents no polynomial effect and 1.000 represents complete polynomial effect on the output.

- **Filter Type:**

  You can choose between “Polynomial” or “Spline” modes via the radio buttons. In “Polynomial” mode, you create your transfer function by entering your data as coefficients ranging from **A0** to **A5** in the appropriate numerical fields. If you choose “Spline”, you can create the transfer function you desire graphically by manipulating (with your mouse) the 4 square shaped inflection points presented on the graphical display.

Spectral Filter

(Forensics Version Only – Advanced Filters)

This filter is essentially a very high resolution Graphic Equalizer that uses FFT techniques. It includes four EQ modes of operation from which to choose. In manual mode, it allows you to create a very high-resolution frequency response contour containing up to 32,000 bands of equalization (essentially a 32,000 Band Graphic Equalizer). The user interface system is intuitive and allows you to zoom-in on a particular portion of the audio spectrum that needs accurate and specific frequency response contouring. By using the right mouse button, you can add bands or inflection points, or delete them simply by pointing and clicking on the graph. This is very useful in Forensic audio applications for removing in-band and out-of-band extraneous noises because of its high degree of frequency selectivity and its very steep slope characteristic.

The spectral filter also includes a spectral inverse filter mode which can be used either manually or automatically. This feature measures the signal amplitude per frequency bin in a .wav file and then applies an inverse response curve in order to normalize it to a reference contoured shape. Essentially, it reverse normalizes to constant signal amplitude per unit bin against a user selected (by preset selection) curve, or via a user customized curve. In other words, frequency bins with lower signal levels than the reference curve are amplified until those bands equal the reference level and those frequency bins with greater amplitude than the reference curve are reduced in amplitude until they match the reference curve. One could view this as an automatic equalizer or an Auto EQ. Spectral inverse mode is very effective for creating an automatic equalization of poorly recorded forensics audio files. This can result in dramatically improved intelligibility of distorted or muffled sound files. It can be viewed as a speech clarifier. File sampling for this feature can be performed manually by highlighting the area of interest or automatically by way of the use of the Auto Sample (checkbox) feature. In Auto Sample mode, the spectral inverse filter becomes essentially an adaptive equalizer and continuously re-samples your .wav file (or system input signal in the case of Live Preview mode) on the fly. Live Preview mode is accomplished via the Multifilter feature in your Diamond Cut suite of filters.

The Spectral Copy mode of operation of the Spectral Filter allows a.wav file to be normalized in terms of frequency content per unit bin to another file or to a certain section of a .wav file. To use the Spectral Filter in this manner, simply highlight a portion of a .wav file while in Spectral Copy mode and then click on the Sample Spectrum button. Next, bring up another .wav file and either Preview or Run the filter. The spectral response of one file will then become imposed on the second file. This mode may be found to be useful when an exemplar of a particular sound exists in one file, but is suspected to be buried in the noise of a second file. An examiner could then sample on the exemplar file containing only the sound of interest (its sound-print) and use that response to help amplify signals in that area of the spectrum on a second file in order to makes certain signals more discernable or intelligible.

A fourth Spectral Filter mode of operation is called "Spectral Difference" (sometimes referred to as Spectral Matching or EQ Matching). In this mode, you can impose the frequency response of one file onto another. This is useful if you want two independently recorded sound tracks to sound similar in terms of frequency response. Simply put, you can cause one .wav file to sound sonically similar to another. To use this mode, you must perform the following operations after switching the Spectral Filter EQ mode to "Spectral Difference". You will note when this operation is performed, that a new button appears labeled "Sample Source". Before proceeding, normalize the gain of both .wav files to the same value (this feature is found under the CD Prep Menu).

1. Bring up your reference .wav file (the file which you want your other file to sound like). This first file is your "Source"

file which should be the sonically higher quality of the two files that you will be working with.

2. Highlight a sector (around 5 to 30 seconds) of this source reference file. The sample should be taken somewhere in the middle of the file. It should not be taken at the noisy lead-in sector of the file. Better signal averaging results from longer sampling time values.

3. Click on the "Sample Source" button and the system will then perform some calculations culminating in the creation of a green colored graph which represents the frequency response of the sample.

4. Next, bring up your target .wav file (the file requiring frequency response modification).

5. Click on the Spectral Filter Preview Button and you will notice that the system will draw two more graphs.

6. You will then hear the corrected response of the Target .wav file.

7. Optionally, you can highlight an area of the Target .wav file and press the "Sample Spectrum" button. This will take a sample of the target file and compute a difference spectrum.

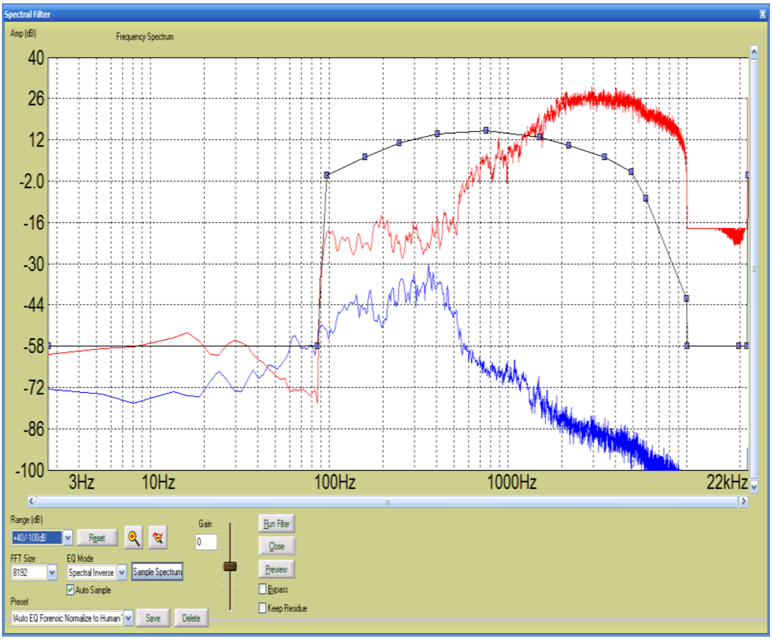
8. The frequency response of the Target .wav file is shown in Blue on the graphic display.

9. The red spectral curve is the difference spectrum and is also the equalization curve that will be applied to the Target .wav file.

The upper and lower crossover frequencies are settable in the "Spectral Difference" mode of operation by way of the "twin green goalposts" which are visible on the graphical display. These vertical green lines can be "dragged" horizontally along the frequency axis using the left mouse button. The left-most "goalpost" controls the lower crossover frequency while the right-most “goalpost” controls the upper crossover frequency. The frequencies in-between the “goalposts” are the values for which the spectral difference is calculated and is reflected by the shape of the red graph. Those signals above and below the “goalpost” crossover frequencies revert to the native target file values. Adjust these “goalposts” until you achieve the most natural sound with the least amount of high frequency hiss and low frequency rumble.

Another mode of operation which may be found to be useful in some applications (having widely varying target frequency responses) is the Auto Sample button option. This will update the calculation of the spectral difference on-the-fly providing you with a variable spectral difference compensation system.

> **Note 1:** Forensics Audio Work may benefit from high values of FFT size while studio audio work (or “Hi-Fi” work) may benefit from substantially lower values of FFT size.
>
> **Note 2:** The Spectral Difference filter can be used in a variety of ways. You can use just one file and sample different portions to generate a difference spectrum or use two different files and apply the spectral difference to a third file.
>
> **Note 3:** When using Spectral Copy and/or Spectral Difference modes of operation, both .wav files must have the same sampling rate and bit depth attributes.
>
> **Note 4:** When using the Spectral Difference mode to match the response of a noisy file to a clean file (Cassette and CD for example), it is important to do all of the restoration work on the noisy file before applying the spectral difference filter otherwise the noise components with be detected as signal and distort the desired response.
>
> **Note 5:** When using Spectral Copy and/or Spectral Difference modes of operation, both .wav files must use the same FFT Size setting.



The Spectral Filter

The following controls are provided in the Spectral Filter:

- **Amplitude vs. Frequency Graph:** Up to three graphical lines are shown on the Spectral Filter graph. In Manual mode, a black line with square inflection points (or touch points) are shown. In Spectral Inverse Filter mode, the blue line represents the frequency response of the .wav file (or a portion thereof). The Red line represents the calculated Inverse Spectrum response required to correct the response to the black line curve.

- **Graphic Representation of the user defined Frequency Response:** Drag the blue inflection points (dots) with your mouse in order to establish the desired frequency response.

- **Range (dB):** This parameter determines the vertical axis range for the Spectral Filter. The following ranges are provided:

    1. + 40 / - 100 dB
    2. + 20 / - 40 dB
    3. + / - 40 dB
    4. + / - 20 dB
    5. + / - 10 dB
    6. + / - 3 dB

- **FFT Size (Number of Frequency Bands):** Higher Values of FFT Size improves frequency resolution / frequency discrimination but with the tradeoff of higher levels of digital artifacting. Choose the best balance between those two requirements based on the needs of the file that you are working on. The following FFT size selections are available in the Spectral Filter:

    1. 256 (128 Bands)
    2. 512 (256 Bands)
    3. 1,024 (512 Bands)
    4. 2,048 (1,024 Bands)
    5. 4,096 (2,048 Bands)
    6. 8,192 (4,096 Bands)
    7. 16,384 (8,192 Bands)
    8. 32,768 (16,384 Bands)
    9. 65,635 (32,768 Bands)

- **Zoom In:** Click on the Magnifying Glass Icon with the "+" sign or drag the mouse over the area of the spectral graph on which you desire to focus.

- **Zoom Out:** Click on the Magnifying Glass Icon with the "-" sign.

- **EQ Mode:** Allows you to choose between the four main modes of the spectral filter. They are Manual Adjustment, Spectral Copy, Spectral Inverse and Spectral Difference. Manual Adjust is a direct manipulation of the frequency response curve similar to a standard Equalizer. Spectral Copy creates a response curve that is the shape of the average response of the sampled file. Spectral Inverse creates an EQ response that seeks to normalize the response of a signal to a desired frequency response curve (or contour). Spectral Difference creates an EQ response that represents the difference is frequency response between to samples.

- **Auto Sample Checkbox:** This feature enables an adaptive mode of operation in which the system automatically samples and applies a portion of the .wav file to the Spectral Inverse or Spectral Difference Filter for normalization / correction on an ongoing basis. This feature is particularly useful in "moving mic" situations when the acoustical environmental is changing throughout the recording. This mode of operation will help optimize the intelligibility of the .wav file in these variable acoustical environment situations. You can freeze the sampling any time desired by un-checking the checkbox while previewing the file. If you freeze the system using the Auto Sample feature, the last sound-print will be held in memory thereafter and will be used as the reference response when the filter is "Run".

- **Gain Control Slider:** Adjusts the Spectral filter overall output level over the range of – 40 dB to + 40 dB.

- **Bypass Check box:** Allows you to hear the source material with the Spectral Filter disconnected from the system.

- **Keep Residue:** Allows you to hear what you are removing when using the Spectral Filter. Mathematically, the keep residue signal is the Input Source Signal minus the Spectral Filtered Signal.

- **Reset:** Returns all off the blue inflection points (bands) on the graphical display to a "flat" (white noise related) response.

- **Sample Spectrum Button:** This control is used in conjunction with the Spectral Inverse Mode of operation. It allows you to take a sound-print (manually) of a highlighted sector of your .wav file for use as the signal to be used to create the desired spectral inverse filter response.

- **Sample Source Button:** This button is used to sample a source reference file's frequency response and is only appears when the system is operating in Spectral Difference Mode.

- **Presets:** (Save or Delete buttons) - The factory presets are especially useful when using the Spectral Inverse Filter Mode in that Normalization curves are provided to White, Pink, Inverse Pink, Brown and Inverse Brown noise contours with ease. Other useful curves are also provided. For definitions of the mentioned noise types, please refer to the Glossary of Terms section of this Users Guide.

- **Right Mouse Button:** To use this feature, select the function you desire with the right mouse button and move the band with the left mouse button. Here are the functions available for the Spectral Filter by using the Right Mouse Button:

    1. Add Point - (This feature adds a frequency band on the graph where you are pointing with the mouse.)*
    2. Delete Point - (This feature deletes a frequency band on the graph where you are pointing with the mouse.)
    3. Reset Point Count - (Use this feature to reset the system to its default value - - - factory default = 0 dB, flat-line response.)

***Note 1:** Adding or deleting frequency bands can also be accomplished by clicking the right mouse button on the graphical display. A dialog box will pop up giving you some optional actions.

Note 2: It is important to convert all forensics .wav files (8 kHz, 11.025 kHz and 22.05 kHz) up to a 44.1 kHz sampling rate before using the Spectral Filter. To accomplish this, use the Change Sample Rate feature found under the Edit menu.

Spectral Filter Application Example #1
You have a Forensics Audio file that is extremely muffled, meaning that its intelligibility is extremely poor because of a lack of the sibilant sounds due to significant high frequency loss. To improve the intelligibility of this file, place the Spectral Filter into Spectral Inverse Mode and check the Auto Sample checkbox. Choose the Auto EQ, Normalize to White Noise or the Auto EQ, Normalize to Human Voice preset. Preview the file. If the sound is improved to your satisfaction, then “Run” the Filter. If the improvement is not adequate, experiment

with some of the other Auto EQ based factory presets until you find the best one for your purposes.

Spectral Filter Application Example #2

You have a recording of a male and a female chatting with one another in an automobile. The recorder microphone is closest to the female and it is very hard to discern the male end of the discussion. The background noise is due to the high speed at which the automobile is traveling. However, there is one small sector of the recording where the car was stopped at a traffic light and both the male and female ends of the discussion are clear and discernable. To improve the clarity of the male voice during the times at which the car is traveling at high speeds, place the Spectral Filter in Spectral Copy mode. Highlight the male voice (only) at the point where the car was stopped at the traffic light and then click on the Sample Spectrum button in order to obtain a sound-print. Then, using this sample, “Preview” or “Run” the Spectral filter in the areas of the recording during which the male voice was not discernable.

Spectral Filter Application Example #3

You have a cockpit voice recording containing a lot of airplane noise. You desire to listen to hear if the flaps-down switch has been flipped at a certain point in time, but at the point in time in question, all you hear is random noise. You desire to enhance the playback of the recording to hear whether or not the sound of that switch being flipped by the pilot of the aircraft is present on the recording. You can go to a flight simulator of the aircraft in question and record the sound of that switch being flipped with a quiet ambient noise environment. Bring up that sound file and highlight the actual sound of the switch being flipped. Next place the Spectral Filter into Spectral Copy mode of operation and then click on the Sample Spectrum button. Bring up the cockpit voice recording .wav file and “Preview” it around the time period in question. This action should provide more gain of the sound-print spectrum occupied by the flipping of the switch potentially making it more audible. Obviously, you can’t prove the negative by this technique, but if the switch is heard, you can demonstrate that something took place at the point in time in question. Of course, that positive result may represent the wrong switch being thrown, but that is beyond the scope of this application example.

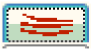
Spectral Filter Application Example #4
You have recordings of an old television variety show from the 1950s (created before the advent of video tape recording). Your job is to assemble a new digital edit of this television series. The dialog and music are both recorded on a cine optical track. However, a much higher quality copy of the musical portion of the show exists on magnetic tape (sans dialog). In the editing process, you need to use the optical track to assemble the edit, but you insert the magnetic track for the musical portion of the job because of its higher sonic quality. However, you observe that there is an unnatural transition between the dialog (optical) track and the musical (magnetic) track. You can create more natural transitions between the dialog and the musical interludes by employing the Spectral Difference mode of the Spectral Filter to the project. Use the Magnetic track as the "Source" Track and then apply that to the Target optical portions of the final edit using the Spectral Difference feature.

Spectral Filter Application Example #5
You have transferred an old acoustical recording of a famous opera singer to your hard drive. You have removed the noise, but the recording sounds honky, hollow, resonant and unnatural. But, you have a more modern recording of this singer created during the electrical recording period of time. You can use the Spectral Difference feature to create a more natural sounding result. Just use the more modern recording as the Source .wav file and then apply it to the acoustical recording as the Target.

Spectrograms

Both the DCArt10 and the DCForensics10 Audio Laboratory versions include spectrogram displays (sometimes referred to as "spectrographs"). The DCArt10 version includes a Standard Definition system while the DC Forensics10 Audio Laboratory version includes a High Definition system in which the user can optimize and tradeoff between optimal frequency or time resolution displays. Since the High Definition Spectrogram is similar in overall operation to the High Definition system, it is recommended that the user should read the section pertaining to the Standard Definition system before proceeding to the High Definition section of this users guide.

View Spectrogram



A Spectrogram (or spectrograph) provides a method for displaying waveform data including Time, Frequency and Amplitude (Loudness) all on the same graph. Time is represented on the horizontal (X) axis, frequency is represented on the vertical (Y) axis, while Amplitude is represented by color intensity or gray scale/brightness. The Spectrogram displays itself in the Destination window when this is checked. The Spectrogram is calculated for whatever waveform is displayed, highlighted and zoomed in on in the Source window and is time aligned with the same. What follows is a description of the Standard Definition Spectrogram found in the DCArt10 version of the product.

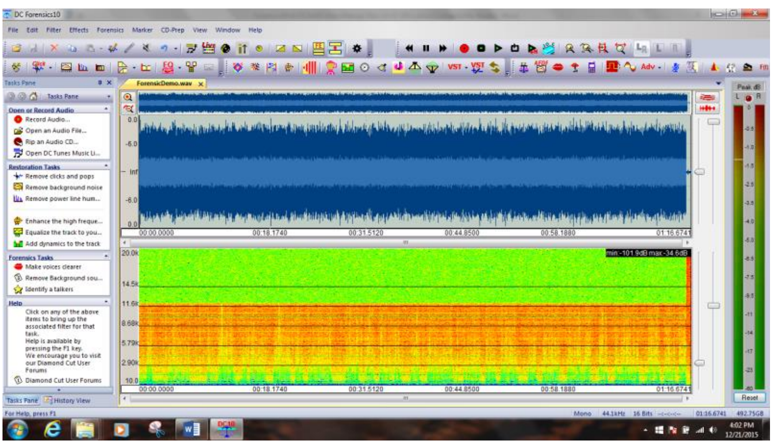


Figure 93 - The DC Forensics Spectrogram View

Display Controls:

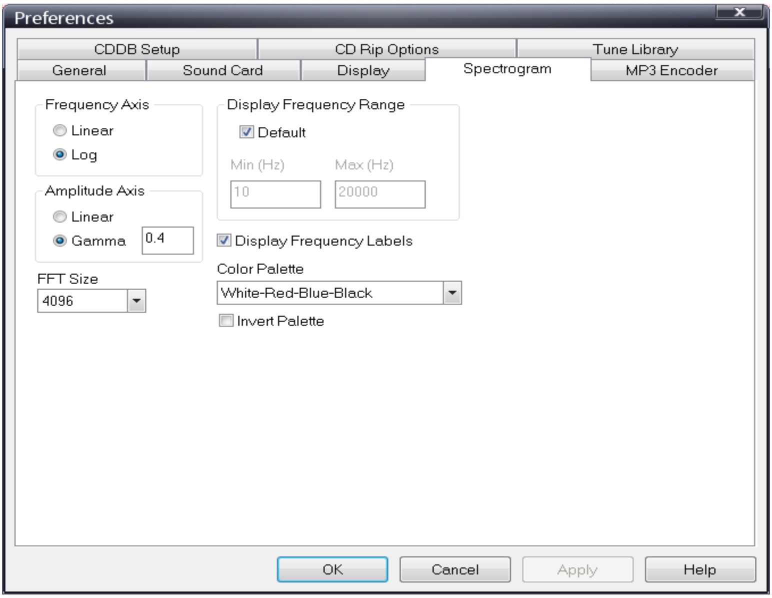
- **Right Vertical Slider**: This slider sets the Maximum Amplitude Level (limit) used for the modulation of the spectrograms Z Axis (chroma level/brightness level). Its range of adjustment is from 0 dB down to as low as – 90 dB. Signals above its setting are suppressed. This applies to the High-Definition version of the Spectrogram found in the

Forensics product. Note that this control operates differently compared to the Standard spectrogram's Right Vertical Slider.

- **Left Vertical Slider**: This slider sets the Minimum Amplitude Level (limit) used for the modulation of the spectrograms Z Axis. Its range of adjustment is from - 25 dB down to as low as - 125 dB. Signals below its setting are suppressed. This applies to the High-Definition version of the Spectrogram found in the Forensics product. Note that this control operates differently compared to the Standard spectrogram's Left Vertical Slider.

- **Zoom In***: Allows you to zoom-in on a portion of the .wav file as displayed in the Source window. After zooming is completed, the system will re-calculate the spectrogram for the zoomed-in segment displayed in the destination window.

- **Zoom Out***: Allows you to zoom-out from a portion of the .wav file as displayed in the Source Window. Similarly, the system will re-calculate the spectrogram for the zoomed-out displayed data in the destination window.



The Spectrogram Display Preferences Menu

Display Preferences:

You can choose between a number of preferences associated with the spectrogram under the Preferences menu found under "Edit" or by left mouse - double clicking on the spectrogram display area. The following preferences are available to you:

- **Frequency Axis Selector:**
  A. Linear
  B. Log

- **Amplitude Axis** (Z Axis or Chroma/Intensity Modulation)
  A. Linear
  B. Gamma Scaling (Co-efficient of Non-linearity ranging from 0.1 to 10 with 1.0 being linear)

- **FFT Size:**
  Choose between 32, 64, 128, 512, 1024, 2048, and 4096. The Forensics High Definition version allows more choices, including 8192, 32768, 65526 and

131072. Small values provide fast FFT update time, while large FFT sizes provide improved frequency resolution. The basic frequency resolution is the FFT size/2.

- **Color Palette:**

  You have the choice of the following color gradients:

  1. Grayscale
  2. White to Blue
  3. White to Red
  4. White to Green
  5. White to Red to Blue
  6. White to Green to Red
  7. White to Red to Blue to Black
  8. White to Yellow to Red to Black
  9. White to Yellow to Green to Aqua to Blue
  10. Black to Blue to White
  11. Black to Green to White

- **Inverse Palette:**

  This inverts the polarity of the video signal providing a different visual perspective of the spectrogram which sometimes is more revealing than the normal polarity. For example, on the grayscale, black become white and white becomes black when the Inverse Palette checkbox is checked.

- **Display Frequency Range**

  1. Enter Value for Minimum Frequency in Hz.
  2. Enter Value for Maximum Frequency in Hz. which is limited to the file Sample Rate / 2.

- **Display Frequency Labels**

  This feature turns the Frequency Labels along the Vertical axis On or Off.

***Note**: For more information on methods for Zooming-In and Zooming-Out, please refer to that section of the User’s Guide.

Important Note: The Sync files feature found under the View menu must be enabled so that the Spectrogram operates properly (stays in sync as you zoom the time display) and should be used in Classic Edit mode.

Spectrograms are useful for applications like spectrographic voice recognition or comparison (sometimes referred to as "voice-printing.") Physiologically, speech is produced by the interaction of two mechanisms consisting of resonance and articulation. Resonance is produced by the nasal, pharyngeal and oral passages while articulators are produced by the jaw muscles, lips, teeth, tongue, and the soft palate. The human voice is acoustically modeled as a 4th order cascaded resonant system with an excitation signal called F0 (produced by the vocal cords). These acoustical signatures are referred to as formants. There are generally 5 formants (acoustical signatures) that are identifiable starting with the fundamental which is usually designated as F0. Resonances produce formants designated as F1 through F4 are generally higher in frequency than the fundamental (F0). All of these formant frequencies lie somewhere below around 3000 Hz. F0 generally falls between around 70 Hz through around 270 Hz. Typically, audio samples that are around 2.5 seconds or less in length with the frequency display range showing information from 100 Hz to somewhere between 3 kHz to 6 kHz are used for vocal comparisons. The so-called English "cue words," often used for comparison are as follows:

{The, To, And, Me, On, Is, You, I, It, A}

Here is a sentence that you can experiment with that incorporates all of the English cue words:

"**It** **is** important that **I** go **to** **the** bank **on** Friday **to** get **a** check for **you** **and** **me**".

There are three pairs of demo files in the DCForensics10 demo wavefiles directory which express the above English sentence. Three pairs of files were made for user testing and experimentation. They include a male voice, a female voice and also that of a child's voice. Each pair of files were recorded simultaneously; in each case one file was recorded through a low quality signal path while the other was recorded via a higher quality signal path. You can use these files in conjunction with the Spectrogram (and the Voice ID System) to study the differences between male, female and child voices expressing the same exact word or the entire sentence. You can also study and compare those same voices as recorded by high and low quality recording systems. The files are as follows:

Female Voice ID Test Sentence - High Quality.wav
Male Voice ID Test Sentence - High Quality.wav
Female Child (12) Voice ID - High Quality.wav
Female Voice ID Test Sentence - Low Quality.wav
Male Voice ID Test Sentence - Low Quality.wav
Female Child (12) - Low Quality.wav

To perform a voice-print comparison, it is necessary to observe the voice same words contained on the two specimens (the display tile feature is helpful for this purpose). Your Diamond Cut Voice ID system is a useful tool for this purpose. You can use it to compare the vowel (generally F1 and F2) and consonant (generally F2, F3 and F4) formants of the human voice. Formant creation is via word highlighting in the spectrogram view. Highlight the area of the spectrogram containing the word of interest and then apply the Voice ID function. The formants shall be plotted on the top of the spectrogram view. It is beneficial if the same recording equipment was used to record both samples of data to be compared, however impractical in most situations. It is also beneficial if the same ambient sound conditions are presented on the two samples. If there is a lot of background noise, consider applying one of the Speech filters before measuring the spectrogram. Speech filters can be found in the Band-pass filter preset menu and the Forensics Brick Wall filter preset menu. See those specific sections of the user's guide for details. Lastly, the emotional state of the person(s) making the expressions should be similar. If one specimen, for example, has the person screaming, and the other has the person sobbing or whispering, it will be difficult to draw any conclusions with a reasonable degree of certainty. Please note that spectrographic "voice-printing" is not admissible evidence in all court systems in the United States. Contact the court system or a legal expert in your area for details.

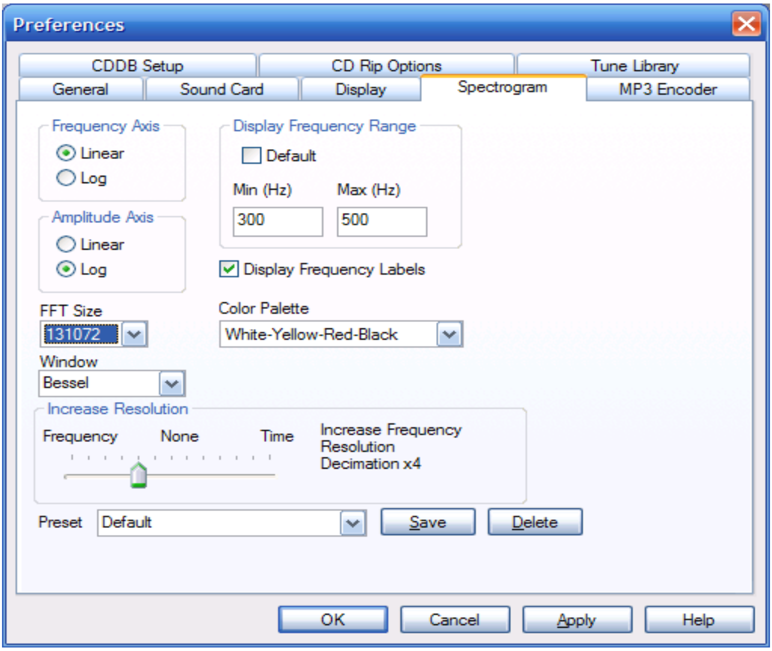
Note 1: Hot key access to the Spectrogram is available via **ALT+"S".** It toggles between the Spectrogram Display and the software's Normal Display mode. Alternatively, you can use the "**Esc**" key to simply turn off the spectrogram.

Note 2: To print a spectrogram, use the Print commands found under the File Menu.

High-Definition Spectrogram

(Forensics Version Only)

The DC Forensics10 Audio Laboratory version includes a very High Definition Spectrogram which allows for FFT Sizes up to 131,072 (65,536 Frequency Bands). It also includes Decimation techniques to further optimize for higher Frequency Resolution as well as Zero-Pad techniques in order to optimize for better Time Interval Resolution. Both the frequency and time resolution parameters have been integrated into one simple and easy to use "Increase Resolution" slider control. Additionally, you can choose between several window techniques depending upon your specific needs. The overall combination of features makes the High Definition Spectrogram well suited to identify tape dubs based on multiple line frequency pickup signals. It is also useful for identifying edit points in forensics audio files. And, it provides an exceptional voice print display for detailed comparison between known and unknown voice sources. The High Resolution Spectrogram dialog box is pictured below:



The High Definition Spectrogram Dialog Box

Most of the functionality of the High Definition Spectrogram is the same as that of the Standard Definition version with the following exceptions:

- **FFT Size:** User Adjustable from 32 to 131,072
- **Window Choices:** Select between Bessel, Blackman, Hamming, Hanning, Kaiser 10, Kaiser 15, Kaiser 20, Rectangular, Triangular and Welch.
- **Increase Resolution Control:** Moving the slider to the left increases the spectrograms Frequency resolution with the reduction of time resolution (by Decimation) while moving the slider to the right increases the spectrograms Time Interval resolution (by Zero Padding) with a loss of frequency resolution. You make the tradeoff that you require using the Increase Resolution Control. The Decimation and Zero Padding values are annotated just to the right hand side of the

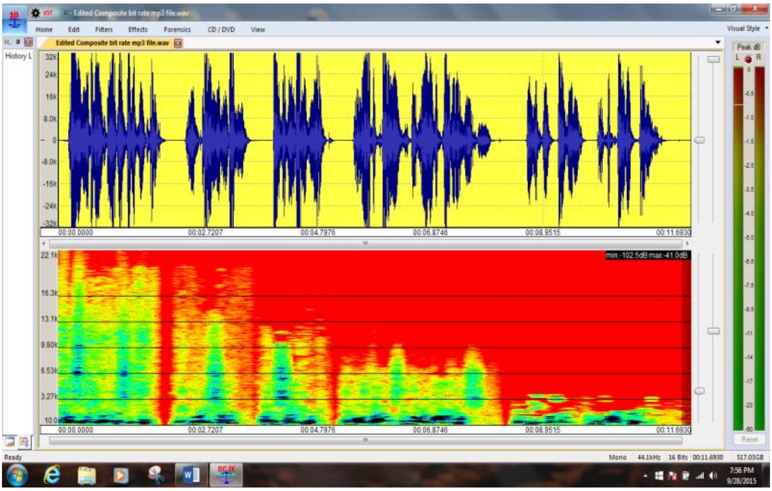
slider control. In the middle slider position, neither Decimation or Zero Padding are applied to the system.

- **Point and Click Measurement:** You can point your mouse to any point on the High Definition Spectrogram and single-left click it and it will display the Frequency in Hz and the Amplitude in dB at that location. The exact measurement location is indicated by a small crosshatch (+) sign on the graphical display adjacent to the numeric display.
- **Presets:** Several factory presets are provided with the software. You can also add your own favorite presets by using the Save command button found in the Spectrogram dialog box. The delete button is used to eliminate unwanted presets from the preset listing.

Note 1: The High Definition Spectrogram responds to the channel selected. When both channels are selected, it responds to the sum of the two.

Note 2: The Spectrogram Dialog Box can be accessed by either double-left mouse clicking on the spectrogram display itself or via the Preferences Menu found under “Edit”.

There is an example of an edited .mp3 file using 5 different bit rates ranging from 224k to 192k to 128k to 48k and finally to 8k. Note the obvious edit points in the file as seen on the high resolution spectrogram (note that this audio file is included with this software as a demo and is titled “Edited Composite bit rate mp3 file.wav”):



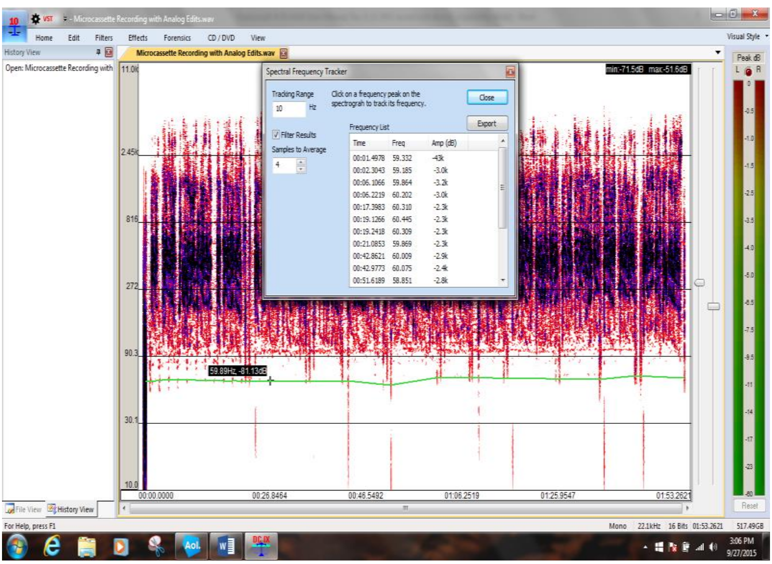
Edited File with varying bit rates shown in sync with hi-res spectrogram

Spectral Frequency Tracker

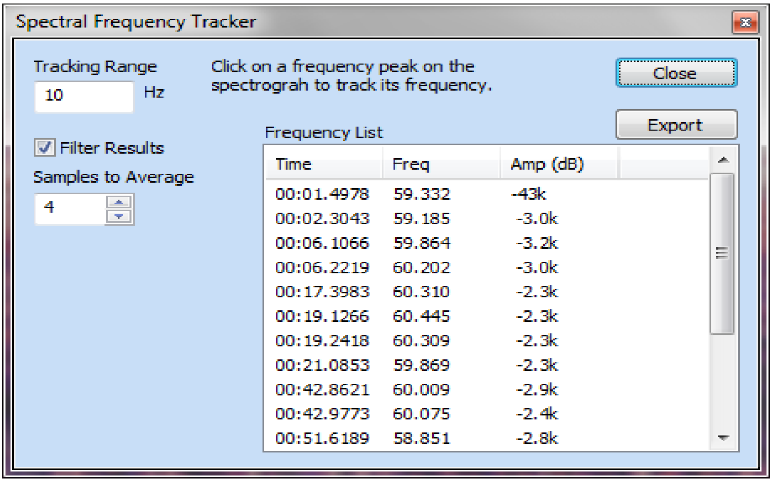
(Forensics Version Only)

There are times when it is desirable to track a specific frequency as it varies over time in an audio file. The Diamond Cut Spectral Frequency Tracker, in conjunction with the Hi Definition Spectrogram provides for frequency tracking and creates an exportable file of the results for further examination (and comparison) via a spreadsheet program. An example where this is useful is involved with the authentication of a recording against the frequency vs time curve of the power grid in a certain locale. Details on this type of analysis can be found in and Application Note called “The Electric Network Frequency Analysis, AN-4” on the Diamond Cut Productions, Inc. website (www.diamondcut.com). Other applications for this function may involve plotting the RPM of an airplane engine based on the 400 Hz generator frequency variations found on a cockpit voice recorder following a forensics event (engines and generator speeds tend to decrease during a stall and speed up during a dive). Additionally, Doppler effects of sirens or bells on vehicles can be studied to help calculate vehicle velocity by exporting this data and analyzing it via a spreadsheet.

The frequency resolution of this system depends on your spectrogram FFT size and resolution settings. The Tracker allows you to choose the frequency window (Tracking Range – 0.01% to 100% of last scanned value) and the Samples (Samples to Average – up to 1000) over which the data is smoothed out. Using your mouse, simply point and click on the spectrogram to track a certain spectral line. When the "Filter Results" checkbox is checked, the data will be logged for direct visual inspection or for export. The data include Time, Frequency and Amplitude (in dB).



The Green line shows a non-constant Line Frequency (located near the bottom of the display)



Spectral Frequency Data Ready for Export

The example above comes from a portable analog microcassette recording. Of note is the variation in the line frequency note or signal as shown by the green line on the Spectral Frequency Graph (which is overlaid upon the Spectrogram). This result is typical of a low cost battery operated analog tape recorder. These data can be dropped into a spreadsheet for further analysis and are in your choice of either .csv or .txt format. Click on the “Export” button to activate this feature.

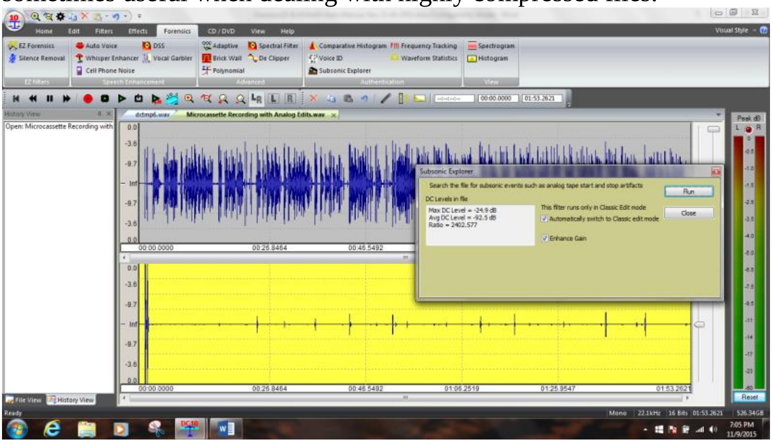
Note 1: The Spectral Frequency Tracker responds to the channel selected. When both channels are selected, it responds to the sum of the two.

Subsonic Explorer

(Forensics Version Only)

Forensics audio recordings often contain very low frequency or DC offset information that can be useful in an investigation. These signals are called subsonic, because they fall below the ability of most people to be able to hear them (and/or for sound systems to be able to reproduce them). The Diamond Cut Subsonic Explorer function allows some analysis of signals that fall below 30 Hz. Some examples of subsonic signals can be distant explosions, arms-fire, earthquakes, slamming doors, vehicle collisions (located far away from the recorder) and stop-start edits produced by analog tape recorders (most commonly – microcassettes). It can sometimes be useful in detecting certain types of digital edits recorded at high signal levels due to DC Offset shifts between two different recorders quiescent operating points. The system works in conjunction with the time domain display in the “classic edit” mode. The top (source) display shows the full-bandwidth audio signal while the bottom (destination) display shows only the subsonic signals derived after processing has been completed by the routine. The horizontal time axis of both displays are in sync for ease of correlation between the two signal portions. When invoking the Subsonic Explorer, the system will automatically revert to “classic edit” mode with the source and destination placed in a “sync files” configuration. Thus, one can observe a subsonic event in the destination display and correlate it to the sonic source file directly above it. Subsonic events generally look like small “blips” or “spikes” on the destination portion of the screen when using this feature. There is usually some form of discontinuity in the Source File that can be correlated with the subsonic event seen in the destination file. In the case of a micro-cassette recording edit point, you may see an abrupt change in the signal level or wave-shape at the edit point. Often, this appears as a “blip” or “spike” in the Subsonic Explorer display. Both files can be played via the playback controls of the Diamond Cut program. Generally, little or nothing will be heard when playing the subsonic destination file (although, something might be “felt” when a high quality sub-woofer is employed). The Subsonic Explorer functionality is nicely augmented when used in conjunction with the Time Display function (found under the View menu) coupled with Markers (the “M” key on your keyboard). Additionally, numerical values are provided for the DC levels of the subsonic signals found by the system. The system is very easy to operate. Simply bring up the file that you want to analyze for subsonic signals. Then go to the Forensics menu and click on the Subsonic Explorer. Click on the run button, and the system will perform some calculations and display the results in the Destination display. Note that the Subsonic Explorer only works in Classic Edit mode. If you had been working in Fast Edit mode, the system will automatically switch over to Classic Edit mode if the “Automatically Switch to Classic Edit Mode” checkbox is checked. An amplitude vs time graph will be created showing the subsonic events, along with numerical data including Max DC Level, Average DC Level, and

Ratio. The ratio measurement is a crest factor calculation of the Max DC Level divided by the Average DC Level and is expressed in linear terms (not dB). A “Enhance Gain” checkbox is provided which imposes a non-linear transfer function to the display which is sometimes useful when dealing with highly compressed files.



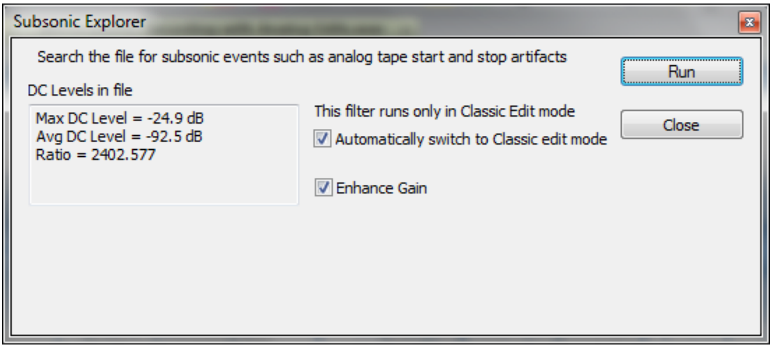
Subsonic Explorer Example: Micro-Cassette Recording with Analog Edits

The screen-shot shown above is that of a Micro-Cassette recording having 9 analog edits. The edits can’t be seen in the regular, full bandwidth signal shown in the upper Source display. However, they become clear in the Destination display after running the Subsonic Explorer. To hear where the potential edits are, just look and listen to the Source display proximal to each blip or spike in the lower display (study the time period 2 seconds before and after each blip or spike looking for anything that may be suspicious).

Note 1: The Subsonic Explorer responds to the channel selected. When both channels are selected, it responds to the sum of the two.

Note 2: An analog edited recording is included with this software so that you can experiment with the Subsonic Explorer. It is called:

Microcassette Recording with Analog Edits.wav.



The Subsonic Explorer Dialog Box

Waveform Statistics

(Forensics Version Only)

There are times when it will become necessary to measure various amplitude related parameters associated with sections (or entire) audio files. The Waveform Statistics system provides that capability. You simply bring up the Waveform Statistics feature and then highlight the area of the file that you want measured. To calculate the data, click the "Build Statistics" button and a display like the one shown below will be created. These data can be copied to your clipboard for use with other software programs such as a spreadsheet or a word processor. The following data are analyzed:

Start Time of the Highlighted area
Stop Time of the Highlighted area
Selected Region (Time Span)
Sample Rate of the File being examined
Bit Depth (resolution)
SHA-1 Hash of File (Unique Cryptographic representation of a file)
RMS (Root Mean Squared)* Value of the highlighted area
Rectified Average Value of the highlighted area
Maximum Positive Peak Value within the highlighted area
Maximum Negative Peak Value within the highlighted area
Crest Factor (ratio of Maximum Peak divided by the RMS Value)
Maximum Low Frequency Value below 10 Hertz
Average Low Frequency Value below 10 Hertz

Number of Clipped Samples within the highlighted area
Number of times the signal passes through zero (zero crossings)
Average Frequency based on zero crossings (this is most meaningful on periodic signals.)

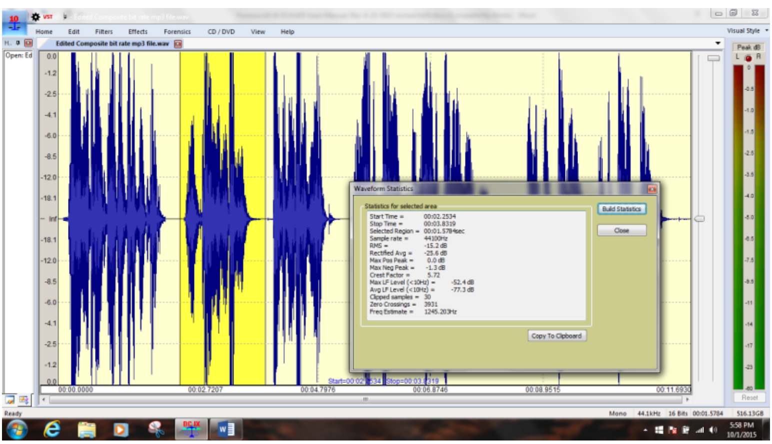
***Note:** RMS is the square Root of the arithmetic Mean of the squared values of the selected set of samples.

The following specific waveform statistics can be used to assist in the estimation of the likelihood that two files (or file segments) were made of the same person(s) at the same venue using the same recording equipment. It is sort of a "litmus test" for audio file equivalence. The following waveform data can be useful in that quest:

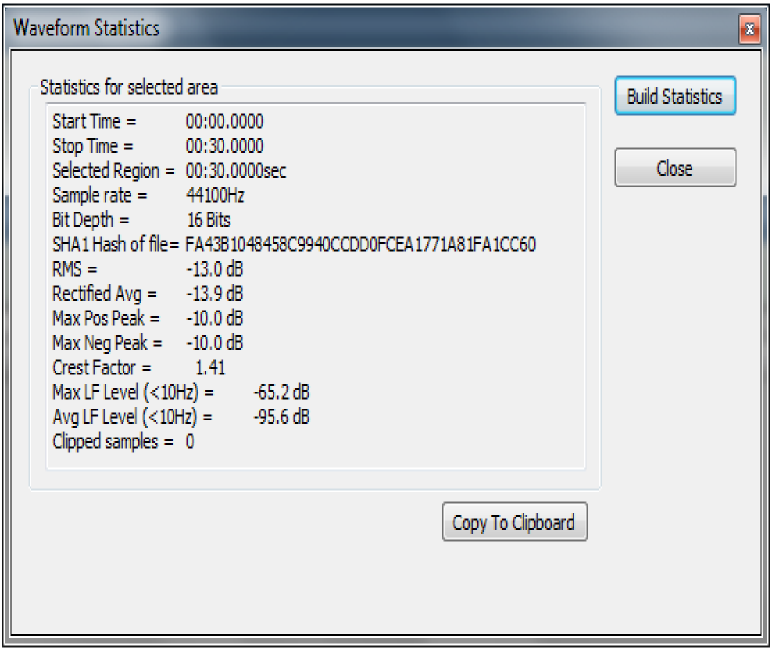
Start Time (of highlighted area)
Stop Time (of highlighted area)
Selected Region
Selected Samples
Sample Rate of the File being examined
Bit Depth (resolution)
SHA-1 Hash of File
RMS (Root Mean Squared)* Value of the highlighted area
Rectified Average Value of the highlighted area
Maximum Positive Peak Value within the highlighted area
Maximum Negative Peak Value within the highlighted area
Crest Factor (ratio of Maximum Peak divided by the RMS Value)
Maximum Low Frequency Value below 10 Hertz
Average Low Frequency Value below 10 Hertz
Number of Clipped Samples within the highlighted area
Number of times the signal passes through zero (zero crossings)
Average Frequency based on zero crossings (this is most meaningful on periodic signals.)

Of special importance is the SHA-1 data, which is unique for any given audio file. It is a cryptographic hash consisting of a 40 digit hexadecimal number which is essentially a unique fingerprint of the entire audio file. This number is a good way to tell if one file is identical to another file. Thus, it can be useful in Forensics audio work to uniquely identify a file or to detect a file that has been tampered with. That hash number is always calculated over the entire file regardless of the selected area or the channels selected. If even just one audio sample of a file is modified, it will create a different SHA-1 hash number. For example, if you bring up the Demo Wave file called "Male Voice Test Sentence BW" and compare it to "Male Voice ID Test Sentence High Quality" they will sound the same and have the same length and other attributes. However, the first one has one sample edited compared to the other and thus they yield completely different SHA-1 values (proving that they are not identical).

Note: The Waveform Statistics function responds to the channel selected except for the SHA-1 hash (which is always calculated for the entire file, including both channels). When both channels are selected, it responds to the sum of the two (except for SHA-1).



Selected File Portion for Waveform Statistics Analysis



Selected File Portion Waveform Statistics Dialog Box

The Waveform Statistics shown in the "Selected File Portion" example above comes from a "Make Waves Signal" having the following characteristics. You should be able to reproduce these results with your software:

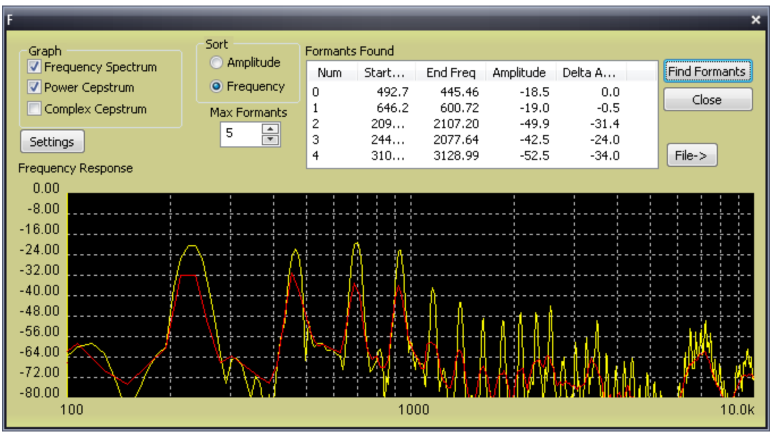
Frequency: 1000 Hz
Amplitude: -10 dB
Length: 30 Seconds
File Type: Stereo
Sample Rate: 44.1 kHz
Wave Shape: Sine Wave
Resolution: 16 bits

Note: Some amplitude related parameters (such as RMS) will be displayed in either Samples or dB or % of full scale depending on your display preferences setting (Edit Menu/Preferences/Display/Y Axis).

Voice ID

(Forensics Version Only)

Your Diamond Cut Forensics10 Audio Laboratory software contains a special feature called "Voice ID" to help you identify and rank the speech formants* of a highlighted cue word or verbal expression. It also includes the ability to display the frequency response, power cepstrum,* (cepstromgram) and complex cepstrum* graphs of the highlighted signal simultaneously.



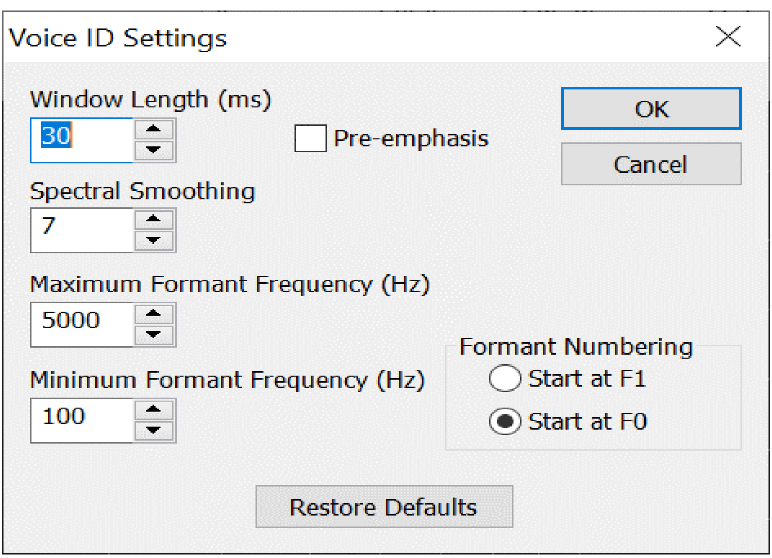
The Voice ID Display Dialog Box

This feature works in conjunction with your high resolution spectrogram and is found under the Forensics menu. The Voice ID feature allows you to highlight a cue word or phrase and then (after activating the Spectrogram) it finds and automatically ranks the various formants contained in that portion of the .wav file. It also provides you with a frequency response and cepstrum graph within the Voice ID dialog box. The system can measure the formants and cepstrum for time intervals up to 5 seconds based on the highlighted sector of the file. Most useful voice vowel formant identifications are performed on a much smaller range of time, usually in the 40-100 mSec range.

You can choose the number of formants that you want to be identified and ranked. Formants are always sorted based on the highest average amplitude of the formant. You can choose between having these results displayed in an Amplitude or Frequency priority order. You can also choose which formant number you want the ranking to commence with by choosing between "Start at F1" or "Start at F0". Maximum and Minimum formant frequencies can also be defined by these features found under the "Settings" dialog box.

The system will generally show the first formant as F0 on the spectrogram (and Num 0 in the table of values). This F0 generally represents the fundamental excitation frequency. The component signals (formants) that comprise a cue word or phrase are displayed in terms of start and stop frequency, amplitude and start time by Fn.

Sometimes, you will encounter noisy files which can confuse the Voice ID System. In these cases, it may be useful to limit the range of frequencies used by the system or to raise the "Spectral Smoothing" value. By default the Voice ID frequency range is set to a lower limit of 100 Hz and an upper limit of 5,500 Hz. You can access and change the various internal parameters of the Voice ID system by using the Settings button on the Voice ID dialog. You can restore the Voice ID system to the factory settings by clicking on the "Restore Defaults" button.



The Voice ID Settings Dialog Box

It is recommended that files should have a sample rate of at least 22 kHz, with 44.1 kHz being a better choice. If necessary, you can convert the file using the file “Change Sample Rate/Resolution” feature found under the Diamond Cut Edit Menu.
You can adjust some of the internal parameters associated with the Voice ID function by clicking on the “settings” button. The Voice ID Settings Dialog Box will appear giving you control over the time window aperture, the spectral smoothing degree, the maximum formant frequency sought after and the application of pre-emphasis. The Window length is basically the signal frequency / time parameter of the Voice ID system. The most common value used is 20 mSec; sometimes it may be useful to try other values depending on the length of the phoneme of interest.

The Pre-Emphasis option is available via the settings dialog. This applies a +1 slope (+6 dB/Octave) from 75 Hz to 5,000 Hz to the signal being analyzed. This is used to compensate the natural roll-off of the vocal tract and flatten the vocal formant spectrum.

After the “Find Formants” has been “clicked” and the calculations have been completed, each formant and its trajectory is displayed on the spectrogram as a number displayed in a rectangular box. They are annotated with both time and frequency coordinates that correspond to the values displayed in the Voice ID dialog box (“Num” column in the table of values). “Start” and “End” frequency values for each formant are annotated in the table of values. Average amplitude values for each formant are given in dB relative to 0 dB which is the maximum value that can be displayed. Also, a column called “Delta Amp” (Delta Amplitude) displays the various formant amplitudes normalized to 0 dB for easier comparison to a reference cue word or phrase.

All of these data can be exported to a text file so that further analysis can be performed with such programs as an Excel or other equivalent data analysis systems. The data that is exported is the frequency and time values for each of the formant tracks.

The software supports two extensions, .txt and .csv, and they both provide you with the same exportable data. Csv (comma-separated values) files are comma delimited while .txt files are tab delimited. To

export the data, just click on the “File” button, then select “Export to a file”. You can set the file path and extension to go to the directory of your choice. You can also choose “copy to clipboard” if you want to bypass writing the information to a file and copy it directly to another program.
Two ceptstrum graphs and a frequency response plot are also provided with the Voice ID feature. The highlighted portion of the file’s Frequency response shows the relative amplitude in dB plotted vs. frequency while the power and/or complex cepstrum can be simultaneously displayed in terms of amplitude vs. quefrency*. Use the “Graph” checkboxes to select the desired graphing mode. The Frequency Spectrum will be drawn in Yellow, the Power Cepstrum in Red and the Complex Spectrum will be drawn in White. You can select any and/or all of these graphical modes depending on your needs. The smoothing control applies to these graphs with higher values producing higher levels of graphic smoothing. The smoothing scaling factor runs from 0 (which produces no smoothing) to 20 (which results in the maximum degree of smoothing). This smoothing control also affects the formants detection functionality.

Six voice ID test files are provided including male (adult), female (adult) and female child in high and low quality versions. All files use the same test phrase for ease of comparison. These files can be found under the File menu/Open Demo Wave Files menu structure. All files include “Voice ID” as part of their file names for ease of location.

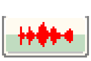
<u>The Voice ID System Operating Procedure(s)</u>

1. Bring up the file of interest in the time domain display.
2. If it is not 44.1 kHz, use the change sample rate feature to change it to 44.1 kHz. This feature is found under the Edit menu.
3. Optionally, go to the “View” menu and bring up the “Time Display” which makes it easier to see your highlighted “Span” time.
4. Go to the “Forensics” Menu and click on the “View Spectrogram” item.
5. Optionally modify the spectrogram properties by using the right mouse button menu to Edit the spectrogram properties.
6. Listen to the file and then highlight the cue word or phrase of interest on the spectrogram.
7. Click on the “Voice ID” function found under the Forensics Menu.

8. Set up the various “Voice ID” parameters to your preference. Generally, the Sort should be set for “Frequency” and the Max Formants would generally be set for 5 (which will allow the system to identify F0 through F4).
9. If you are interested in Frequency Spectrum or Cepstrum graphs, check the appropriate checkboxes in the “Voice ID” dialog box. If not, leave all checkboxes unchecked.
10. Lastly, click on the “Find Formants” button in the “Voice ID” dialog box.
11. The system will then calculate the various Formants and display their various numerical attributes in the table of values within the “Voice ID” dialog box.
12. Please be aware that the voice formant frequencies are very sensitive to the exact location of the selected speech. Small variations in time can cause different formants to be found. Likewise the window length time (under the settings dialog) will affect the type of formants found. Typical analysis is done with a 20 ms window (aperture).
12. The trajectories of the various Formants will appear on the Spectrogram along with their numerical formant labels.
13. If you want to analyze this data statistically or in any other way, you can click on the File->Export button and a file will be created which can be used external to your Diamond Cut software.
*Note 1: Definitions of Power Cepstrum, Complex Cepstrum, Formants and Quefrency can be found in the glossary section of this documentation.
Note 2: The Voice ID dialog box is user sizable; just use your mouse to drag the box margins to create the size that you desire.
Note 3: To print the Frequency Response and/or Cepstrum graphs, use the Alt Print Screen (Alt Prt Scr) command and it will be recorded on your system clipboard to be used as needed.
Note 4: Voices under stress may yield distorted Voice ID results.

View Histogram vs. Time

(Forensics Version Only)

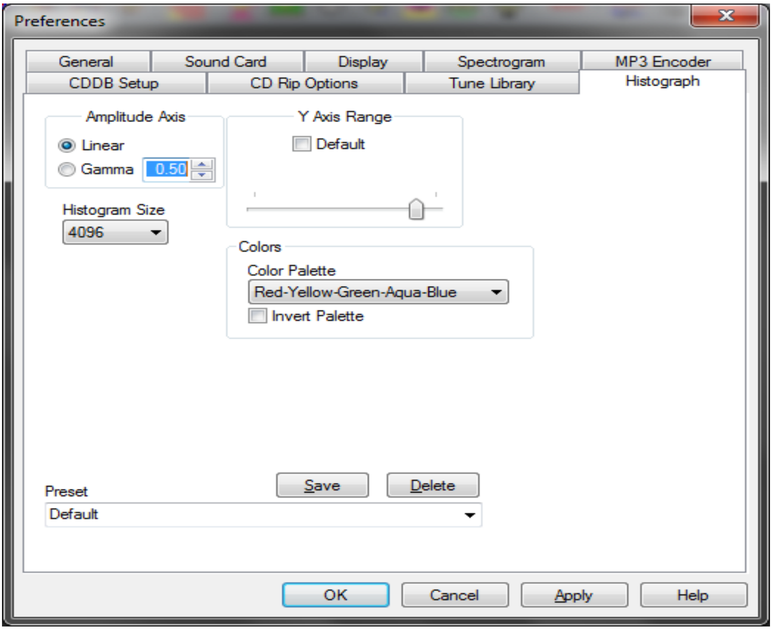


The histogram (sometimes called a histograph) of an audio signal is a probability distribution of the amplitude density of audio signal data found in a sector of an audio file. The Diamond Cut Histogram vs.

Time feature plots this function vs Time. It is normalized so it displays relative frequencies of grouped amplitude occurrences as a function of time. These groups (or bins) have a displayed area which is proportional to the number of certain amplitude events occurring per unit of time. The Histogram vs. Time feature marches through the file and presents a string of histograms to show trends in the file amplitude data distribution throughout the audio file of interest. The Y axis represents counts, while the X-axis represents time. The Z-Axis (color and shade) represents the frequency of occurrence for the scaled (normalized) events. The Histogram vs. Time tool can be useful in the forensics audio authentication process, graphically presenting discontinuities that may be present in a file in question. The routine does not make qualitative determinations, but presents statistical histogram distribution data for interpretation by the examiner. If a bit weight is changed due to an edit in a file, it may show up in "View Histogram vs Time" display, begging further investigation into that region of the sound file.

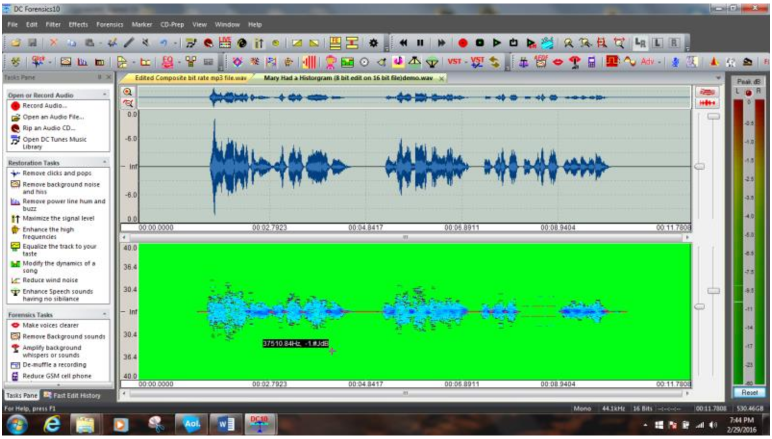
To create the Histogram vs. Time graph, simply click on the item having that name under the Forensics menu and it will calculate the data and create a display for your inspection. Alternatively, simply click on the histogram icon near the upper right side of the time display. The sliders to the right side of the graphical display can be used to enhance the contrast and/or color of the graph; the leftmost slider affects the chrominance (color saturation) level of the signal and the rightmost control affects the background color of the image. To turn the Histogram vs. Time off, just click on that item again in the Forensics Menu and it will be removed from the display screen.

You can adjust the Histogram vs Time graphic display by right-clicking your mouse on the display. You will see "Edit Histogram Properties" at the top of the list and the following dialog box will appear after left clicking on it:



The Histogram Control Panel

Using this control panel, you can adjust the Histogram Size (per bin). It is advisable to use values greater than 256 bins if you care to resolve the difference between 8 bit, 16 bit and 24 bit file edits. A good starting value is 512 bins. The amplitude axis (Z-Axis) can be set for linear or a variable value of Z-axis Gamma. The Y-Axis range can be used to offset the display to hone in on a certain area of the graph.



Histogram vs. Time Example (bottom trace)

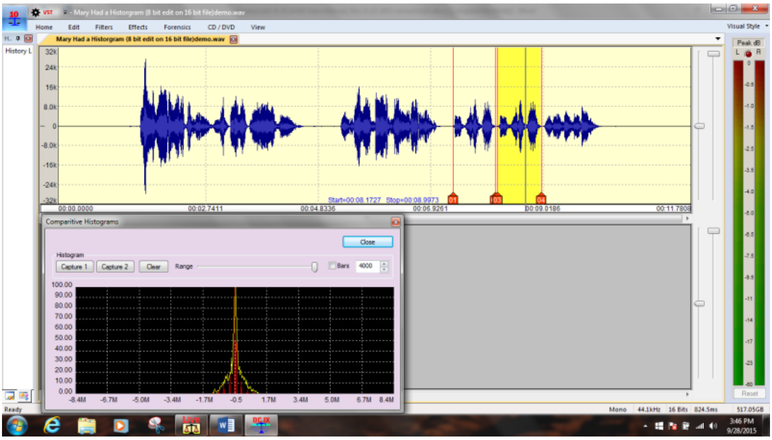
The Histogram vs. Time shown here in the lower trace is taken from the sample file called “Mary had a Histogram” demo file provided with this software. Note the obvious edit shown on the graphical display where an 8 bit edit was performed on a 16 bit file.

Comparative Histogram

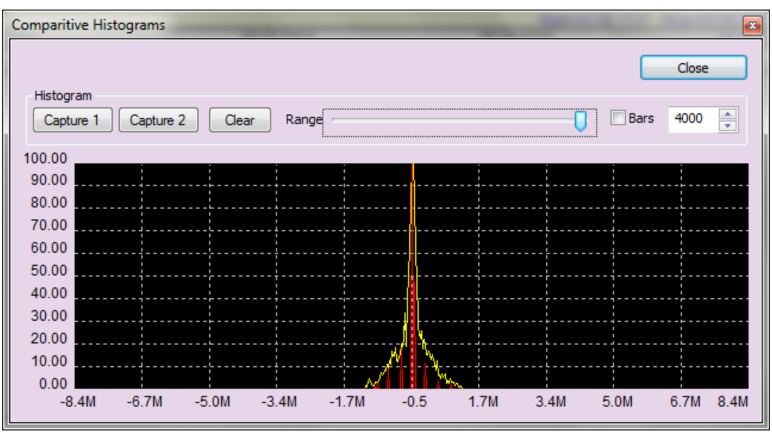
(Forensics Version Only)

A companion tool to the “Histogram vs. Time” feature is the “Comparative Histogram”. It provides you with the ability to compare two different file sectors in terms of their probability distributions of the amplitude density of audio signal samples found on each file segment. This feature can be useful in Forensics audio authentication situations since audio recordings in a given recording environment often display a unique statistical character. The comparative histogram can be used to determine differences in this regard.

The vertical axis represents the frequency of the events scaled to 100% while the horizontal axis represents the amplitude counts within the selected area of the file. This feature can be useful in Forensics Audio authentication situations.   It is sensitive to file portions which may have had varying bit depths when originally recorded. The graphs of sampled portions of a file are graphically over-laid upon one another so that a quick comparison of histogram data is provided.



File Mastered in both 16 and 8 Bit depth and edited together

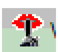


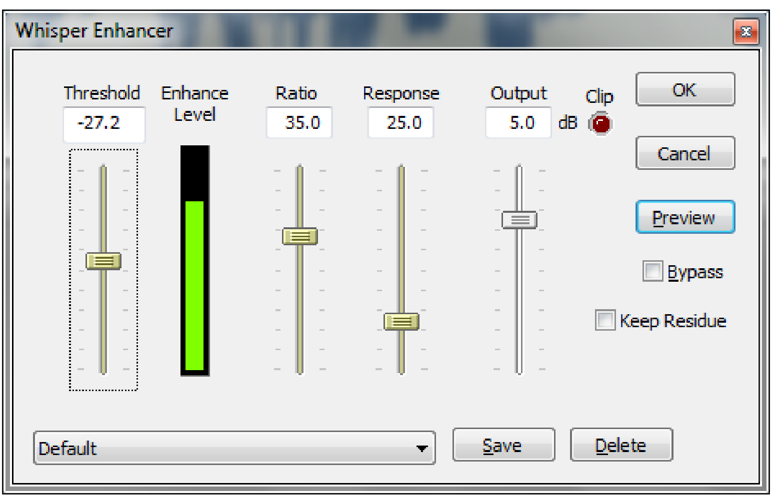
Comparative Histogram of composite 16 and 8 Bit File Edits

The file used to generate the above histograms is called "Mary Had a Histogram (8 bit edit on 16 bit file)demo.wav". The file itself is a 16 bit file, sampled at 44.1 kHz. But, it has an edit at the second yellow highlighted area using an 8 bit file converted to 16 bit and inserted into that time sector. You can't hear or see the edit in the time domain display, however it shows clearly using the comparative histogram function. Note that the first yellow portion of the file is shown as the yellow distribution on the histogram. The second yellow section of the file is shown on the comparative histogram in red. Their signal distributions differ substantially showing that there was an error made by the person who attempted the edit. To use the Comparative Histogram, set the bins number to around 4000 initially. Then, highlight one sector of a file and click on Capture 1 and a yellow graph will be created. Then, highlight another (suspicious) sector of a file and click on Capture 2 and a red graph will be created and laid on top of the first graph for comparison. Note that you can also choose between a line graph (default mode) and a bar graph (checkbox).

Whisper Enhancer

(Forensics Version Only)





The Whisper Enhancer dialog box

Forensics audio situations sometimes arise with multiple parties talking on a recording in which one of them sounds quite far away from the recording microphone compared to the others. This could involve two parties where one of them is at a far distance and the other one being close. Or three parties could be talking clearly to one another, but a third party, adding to the conversation, is at some relatively large distance from the recorder microphone. Another similar situation can occur wherein a completely different conversation may be occurring at another table or location and the investigator is interested in trying to reveal some of that ancillary conversation. Another situation could involve potential gunshots or screaming (or other potentially important events) occurring at a far distance from the recorder and also occurring during breaks in the primary parties conversation. Additionally, the flip of a switch from a flight data recording may be brought to the foreground using the Whisper Enhancer. The Diamond Cut Whisper Enhancer is a special routine designed to bring out soft voices or sounds that are occurring amidst louder voices while softening the louder voices or sounds. It can reduce the loudness of the up-front voices while increasing the loudness of the far distant voices or sounds, so long as they are not occurring concurrently. Sometimes, it will be found that passing the audio through a speech filter (a preset under the Band Pass filter) or the Auto Voice filter first will provide better results with the Whisper Enhancer. It is important to note that this is not intended to be used as a de-noising filter and it has the capability of introducing distortion into the signal when its controls are not properly set for the audio material of interest. Here are the controls that you have to work with in the Diamond Cut Whisper Enhancer:

Threshold: This control sets the signal level at which the system begins to invoke its action.

> Range: -60 dB to 0 dB (0 dB is the least aggressive; – 60 dB will yield the most aggressive setting)

Enhance Level Bar graph: Use this to aid in finding the “sweet-spot” for Whisper Enhancement. The higher that the green bar graph moves, the greater will be the action of the enhancer.

Ratio: This controls the magnitude of loudness between the loudest and softest sound processed. Higher values of ratio produce a greater “Whisper Enhancement” effect. Generally speaking, your best results will occur with ratio values set somewhere between 15 and 50 depending on the nature of your audio material.

Range: 5 to 50 (relative units of measurement)

Response Time: This determines the time delay before the system activates. Small values of response time produce very fast transient response to changes, but produce the greatest amount of distortion. Conversely, high values of response time produce slower transient response to changes, but less signal distortion.

Range: 1.0 mSec to 100 mSec

Output (Level): Adjust the signal level coming out of the Whisper Enhancer algorithm.

Range: -40 dB to +19.9 dB

Clip Led: This red indicator lights up when the output signal hits full scale (maximum = 0 dB). Use the Output (Level) control to reduce the signal level until this indicator extinguishes.

Bypass: Defeats the Whisper Enhancer for signal comparison purposes.

Keep Residue: The difference between the input and processed signal.

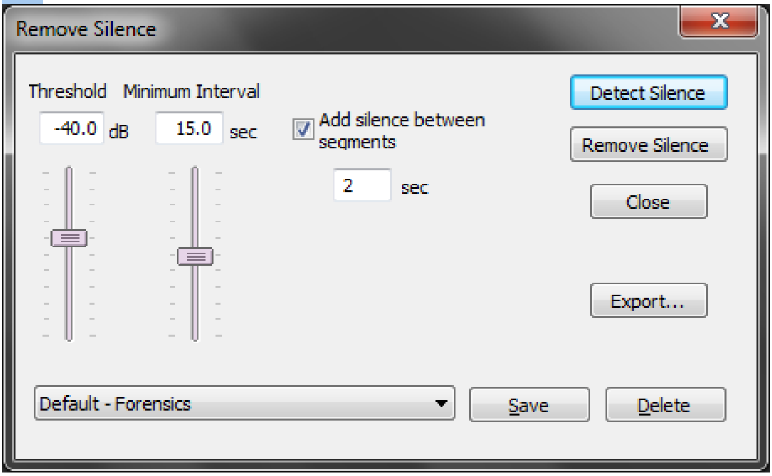
Presets: 15 Descriptive Factory Presets are provided; start with the one that best describes your audio issue and then fine-tune the various parameters for optimal whisper enhancement. After you find the best one, you can "Save" it for future reference using a name of your choosing.

WARNING: THE WHISPER ENHANCER CAN PRODUCE EXTREMELY HIGH SOUND VOLUME LEVELS WHEN NOT PROPERLY ADJUSTED. KEEP YOUR AUDIO SYSTEM VOLUME LEVEL SET TO A LOW VALUE WHEN STARTING OUT WITH THIS ROUTINE. ALWAYS BEGIN A PROJECT WITH THE OUTPUT LEVEL SET TO – 40 dB AND ADJUST TO A REASONABLE LEVEL THEREAFTER.

Note: The Whisper Enhancer works best with files that have been converted to a 44.1 kHz sample rate & having 16 bit depth (resolution).

Remove Silence Tool (Automatic)

(Forensics Version Only)



The Remove Silence Tool Dialog Box

Audio surveillance recordings often include periods of silence. Often, these "dead spots" can last for long time periods. The process of manually marking and/or removing the silent sectors can be quite laborious. The Diamond Cut silence remover automatically deals with this issue. It can automatically mark the silent passages and, if desired, can then remove those "dead spots" from the recording. The system includes pre and post triggering buffers. Thus, leading and trailing edge sounds are preserved to some degree. The removal process is performed in two steps; the first step involves marking the silent sectors and the second step can be used to remove those silent sectors from an audio file. A marker will be placed at the beginning and at the end of each silent sector in the file (2 markers are used per silent sector). Here are the controls that you will need to adjust to use the Diamond Cut Forensics Remove Silence tool:

Threshold: Sets the volume level (in dB) below which the system identifies “silence”.

> Range: 0 dB to – 100 dB (lower settings increase the systems sensitivity to “silent” passages.) If this is adjusted too low, it will trigger on background room noise; it this is adjusted too high, it will not find the silent passages. An initial setting of around -30 dB is a good place to start, but experimentation may be required on a file by file basis.

Minimum Interval (Time): This control sets the minimum time interval that will be defined as “silence” by the systems detector. Time intervals less than this setting will not produce marked sectors of the file. This setting will vary based on the nature of the conversation on the surveillance recording. Again, experimentation is required to find the proper setting.

> Range: 1 Second to 30 Seconds

Add Silence Between segments (Time): This allows you to insert “dead silence” in between audio segments for easier identification of the location of the breaks in conversation.

Detect Silence (button): Clicking on this button activates the Remove Silence tool to search out silent sectors in the file. It will drop a marker at the beginning and ending of each silent sector.

Remove Silence (button): After the silent sectors have been identified and marked by the detector portion of the algorithm, this button can be used to remove those silent sectors, if desired.

Clear All Markers (under the Marker Menu): If you are not satisfied with the placement of the markers, you can re-adjust the “Threshold” and/or the “Minimum Interval” parameters after “Clearing All Markers” produced by an earlier attempt. Clear All Markers is not found within the Remove Silence tool dialog box; it is only found under the Marker Menu.

Export: Allows you to export the time data via CSV formatting for use in other programs (like spreadsheets).

Presets: A number of presets (19) are provided for use as starting points. They are expressed in the following form:

Type XX-YY-ZZ

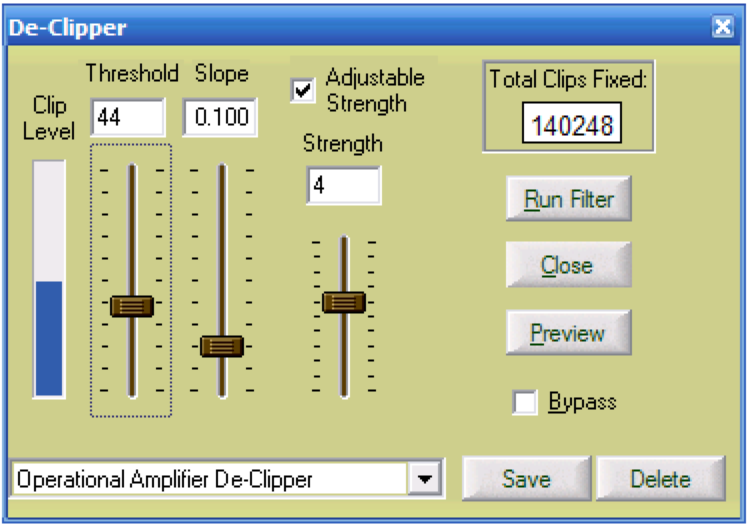
Wherein -

XX is the negative threshold in dB, YY is the Minimum Time interval in Seconds and ZZ is the Silence Added between segments.

Note: Up to 100 markers will be shown on the time domain graph; silent sectors requiring more than 100 markers are logged and the silent areas in between those hidden markers will still be removed with the "Remove Silence" button. Silent sectors greater in 50 in number are still logged and their locations are kept in .csv format and the data can be exported.

De-Clipper

The De-Clipper is provided to reduce the distortion which results from an over-driven or clipped signal. When viewed in the time domain via the Source display window, this problem often looks like the signal has a "crew cut" or is "maxed-out." The De-Clipper is equipped with two modes of interpolation, one having adjustable strength while the other uses a complex frequency domain method of interpolation. The "Adjustable Strength" mode is most useful on signals having a fairly high level of coherence while the "non Adjustable Strength" mode is best suited to signals having a higher level of randomness in their makeup. The "non Adjustable Strength" routine requires a high level of CPU and thus is quite slow, whereas the "Adjustable Strength" mode is much faster. Experimentation is the best way to find the optimal mode to apply to your particular clipped file. We recommend starting with the Adjustable Strength mode first, since it contains the fastest running replacement algorithm.



The De-Clipper

The De-Clipper can be used to repair signals, which were either clipped by digital or analog mechanisms. It performs its magic by detecting signals with very low or zero values of slope (user adjustable from 0 - 0.5) above a settable threshold amplitude value. When this condition is detected, the routine mathematically interpolates a new signal and replaces the zero slope portion of the bad waveform with one containing curvature. This results in decreased distortion. If the material being de-clipped has been directly clipped by the digital recording process (in other words, the signal is clipped at full scale output as indicated on the destination window), then you must first decrease the overall gain of the .wav file by 6 dB (- 6 dB) before applying the De-Clipper. This sort of de-clipping can be accomplished with very low values of slope. If the signal was clipped previous to the transfer to the digital domain by an overloaded analog amplifier, the signal can be de-clipped by raising the slope control until the "total clips fixed" display starts incrementing. The following is a listing of controls available on the De-Clipper:

- **Adjustable Strength Checkbox**
  This checkbox determines which of two different replacement algorithms are used by your De-Clipper. Check this box when

dealing with audio signals which are fairly coherent in nature and un-check this box for signals which are more random (or stochastic) in general nature.

- **Threshold**

The threshold setting determines the amplitude above which signals will be applied to the de-clipper detector. Lower values imply a more aggressive response. The range for this control is from 10 to 100 on a relative and normalized percentage scale.

- **Clip Level**

The clipped level of your signal is measured and displayed on this bar graph and is useful for setting the “Threshold” control. Close alignment of the “Threshold” control with the “Clip Level” while watching the “Total Clips Fixed” display will allow you to find the best “Threshold” setting. The “Total Clips Fixed” indicator will begin to increment when the proper threshold is found. Do not overdrive the system by setting the “Threshold” control to an excessively low setting as this may introduce distortion into the process.

- **Slope**

The Slope control determines the “flatness” of the clipped waveform that will be interpreted as a “clipped” event. Low numbers like zero, imply a perfectly flat line as will be found in digital clipping. Higher numbers represent slight slopes associated with analog clipping. The range for this control is from 0.000 to 0.500 with 0 representing zero slope and 0.500 representing a 45 degree slope.

- **Strength**

The Strength control affects the curvature of the applied interpolated waveform. Adjust this for the best sounding (or looking) replacement waveform. The range for this is from 1 to 5, with 5 being the most aggressive.

- **Total Clips Fixed:**

This is a numerical display indicating how many “clips” were detected and fixed by the De-Clipper. It is useful when using "preview" mode to assure that the routine is detecting the clipping while adjusting the various controls.

- **Bypass**

This bypasses the De-Clipper so that you can preview and hear the before and after results of the de-clipping process in an instant.

De-Clipper Operating Procedure (Tutorial)

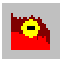
File Preparation: Before de-clipping a .wav file, it is necessary to reduce its amplitude by 6 dB before applying the following procedure. To reduce the gain of the .wav file, use the Gain Change feature found under the Edit menu. The reason for this step is to provide the de-clipper algorithm enough headroom to interpolate the clipped peaks of the .wav file.

1. Set the Threshold control to its maximum value (all the way up).
2. Set the Slope to a value close to zero for digital clipping, but higher for analog clipping.
3. Preview the filter and watch the "Total Clips Fixed" display.
4. Reduce the level of the Threshold until it begins to show increments on the Total Clips Fixed display.
5. Adjust the Strength for a minimization of distortion as heard in preview mode.
6. Run the Filter.
7. Done

Note: You can look at the results produced by the De-Clipper after you have run the Filter. You will notice that the flat-topping has been replaced with smooth rounded waveforms if the controls have been set properly.

DSS Dynamic Spectral Subtraction

(Forensics Version Only – Advanced Filter)



The DSS Dynamic Spectral Subtraction™ feature in the DC Forensics10 software is a unique and powerful tool capable of recovering speech from recordings containing loud music or other coherent noise. Until now, a recording that was covered or masked by loud music was basically a lost cause. DSS decoding is designed to make it possible to attenuate this ambient music and uncover the speech. A demo file is provided to test the DSS called: “DSS Demo”.

Basic Approach of the DSS Filter

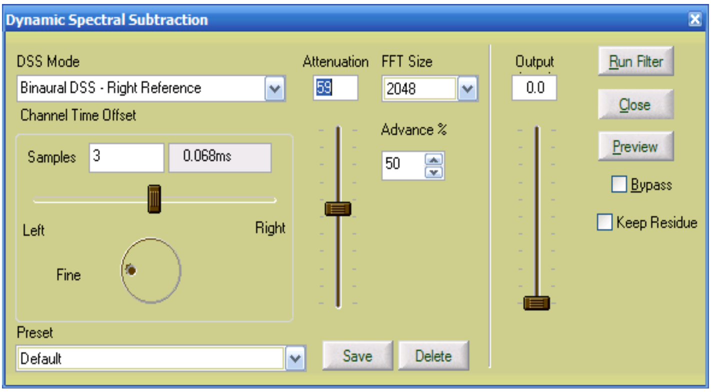
DSS works by performing a continuous and intelligent subtraction of one audio signal from another. Normally, with a Forensic recording containing speech masked by loud audio, you will require a reference recording containing just the audio that needs to be removed. The audio track containing the music or other audio to be removed is called the “Reference Track.”

The DSS Controls

The following four controls are active when operating in DSS mode.

1. Selection Box:
    A. Mono DSS – Delay Reference
    B. Binaural DSS – Left Reference
    C. Binaural DSS – Right Reference
2. Attenuation: Range = 10 to 100 (Tune for a Null in the Noise) Null usually occurs around 50.
3. Channel Time Offset: (Adjust for the best nullification of the unwanted signal.)
    A. Horizontal Slider Control: Course Time Offset Adjustment
    B. Rotary Dial Control: Fine Time Offset Adjustment
    C. Offset Samples Display Window (given in # of samples)
    D. Offset Time Display Window (given in mSec)

4. Advance %: Range = 10% to 50% (Tune for minimum digital artifact production). In most instances a setting of around 40% is effective.
5. FFT Size: 256 to 8192 FFTs (adjust for best unwanted signal rejection)
6. Output: 0 to 20 dB – Used to compensate for gain loss when the "sweet – spot" is found when using the DSS.
7. Red Clip LED: Adjust the Output Control downwards when this indicator lights so that clipping distortion does not occur.



The DSS User Interface

Recording a Reference Track

There are many ways that you can obtain a reference track. These examples should make this clear:

<u>Real Time Methods*</u>

1. Place two microphones in the venue. Place one near the target conversation and place the other near the source of the background audio source such as a TV, stereo system, jukebox, or a live band. Record these two signals with a stereo tape recorder, wireless transmitter(s) and/or computer.
2. Wire the investigator with two microphones. Place one near the investigator's chest and place the other much lower on the investigators body, like down in his or her sock or shoe. Record both signals with a miniature stereo tape recorder.

3. Wire the room with a wireless microphone located near the sound source like the TV, Stereo, jukebox or live band. Wire the investigator with a wireless microphone located near his or her chest. Record both signals with a remote stereo recorder or computer.

<u>Non Real Time Methods</u>* ‡

1. Assume that you have a recording made in a bar or similar venue that was recorded with a monophonic pocket tape recorder. The jukebox or other interfering music source is covering over the targeted speech. You can go back later with the same recorder and record the same exact song that was being played. This will become your reference track for DSS decoding.
2. You have the same situation as stated above, but you have a second tape recorder on site that is recording only the noisy background environment.
3. You have the same situation as stated above, but you record the same music that had been playing at the venue from a commercial audio CD. This process can be performed in non real time back in your audio lab.

*** Note 1:** Digital Recorders produce better results than Analog recorders in DSS decoding applications due to their crystal controlled speed regulation. This may not be as optimal, however if the digital recorder uses "lossy" compression such as .mp3.

‡ Note 2: If the interfering source of audio was a radio or television, many broadcast stations maintain an archive of "air-checks". You may be able to access the required broadcast "air-check" recording through either the use of diplomacy or a court order.

Obtaining a reference recording is an important step in removing loud coherent noise sources such as music. Using the Real Time Methods described above, technique number 1 (there under) will produce the best results. In the Non Real Time methods described above, number 2 (there under) will produce the best results since it will rely on a reference recording that closely resembles the noises that you will be attempting to remove from the target signal.

Modes of DSS Operation

There are 3 DSS modes of operation available in the Forensics software product. To select one, drop down the "DSS Mode" selector box. As you can see, you can select either the right or the left channel as the reference track.

If you have no reference track recording and cannot re-create one, you can try to use the setting called "Mono DSS-Delay Reference". This mode will attempt to attenuate the noise by comparing the audio at an instantaneous point in time and comparing it with a point at some other time before or after the comparison point. This has the effect of allowing the program to create its own reference signal.

Note: This method is inferior in comparison to any method utilizing a true reference track.

Creating a Stereo track from two discrete tracks in non real-time situations: *

The audio file that you will actually clean up using DSS decoding is ideally going to be a stereo file that you recorded in real time at the venue. However, often that is inconvenient and non real-time methods must be used. In these cases, one channel of the file will be the recording with the speech you want to recover (the Forensic recording) and the other channel will contain just the music or other non-random audio. These two recordings will have to be combined into a single stereo (binaural) .wav file recording. The easiest way to accomplish this is to use the File Split and Re-Combine function found under the Edit menu. Here's the procedure:

1. Take the two recordings (the Forensic recording and the reference recording) and convert both of them into monophonic files if necessary by using the File Converter Filter.
2. Use the File Split and Re-combine feature to merge these two mono files into a stereo (binaural) file

3. Time align these two files by either cutting a piece from the beginning of one of them or insert a piece of silence of appropriate length in front of one of them. Note: Using the Markers and the Time Display feature is quite helpful to precisely measure the time displacement between tracks to calculate how much audio must either be cut or inserted to result in the proper time alignment. The two tracks should be time aligned (roughed in) to within +/- 25 milliseconds of each other for optimum results.

***Note:** If the interfering audio came from a live performance, having the live performance re-created by the talent after the fact will not produce a useable reference track for DSS decoding.

DSS Adjustments:
The controls that are active in the DSS filter are Attenuation, FFT Size and Delay. The Attenuation setting will control the amount of noise reduction that is performed by the DSS filter. You can think of this control as being analogous to balancing the weight(s) on a balance scale. Moving it up will reduce the noise more until you pass through a "null" point in the background music or noise. You need to tune the attenuation control for the most music reduction, which generally will occur around an Attenuator setting of 50, as long as both discrete channels are relatively balanced in amplitude with respect to one another.

The FFT size controls the size of the frequency "buckets" that are being used internally by the filter. Smaller numbers allows for more "self-adjustment" of the filter to the mismatched forensic and reference recordings. Larger values produce better frequency discrimination and overall attenuation. We find that settings of 1024 or 2048 generally produce good overall results, but smaller or larger settings should be tried as well.

The Advance % control is generally set to 50%. However, it is worthwhile experimenting with other values of Overlap in order to minimize the introduction of digital artifacts into the final resultant signal.

The Time Offset control can also help with time alignment mismatched audio channels. You can calculate the actual delay time between the reference microphone and the target microphone in milliseconds by applying the following formulae:

TD = (Delay Setting – 1)(FFT Size)(Advance % x 0.01) / Sample Rate

wherein:

Delay Setting is an Integer value from 1 to 10

&

Overlap is a value from 10% to 50%

&

Sampling Rate is a value given in Hertz

&

TD is the resultant delay time given in milliseconds

After adjusting the Channel Time Offset for a minimization of background music or noise, you can ascertain the distance between the reference microphone and the target microphone by looking at the Offset Time Display Window. Each mSec represents about 1.1 feet of distance between the two. As always, simply preview the audio and make your adjustments in the DSS filter window and also with the Time Offset slider in the File Conversion filter.

Primary Compensation Issue with the DSS Filter in Real Time Situations:

The primary problem encountered when using the DSS filter in real time applications arises from the distance between the two microphones used to make the binaural recording. Because a physical distance in the venue separated the two microphones, the propagation delay (sometimes referred to as group delay) of the signal between the two microphones in the room may need to be compensated for. Since sound travels at 1131 feet / second at 70 degrees F (or 1.131 feet /

millisecond), large distances between microphones can cause mis-alignment between the two tracks of the binaural file. In real-time situations, the noise signal on the Target Track will lag the Reference Track at the rate of 0.884 milliseconds per foot. The File Conversion Filter and its Time Offset control can be used to compensate for up to 20 milliseconds of propagation delay representing a distance between the microphones of up to about 23 feet. If more distance had existed between the two microphones, multiple File Conversion Filters can be cascaded in the Multi-Filter to increase the total compensation time.

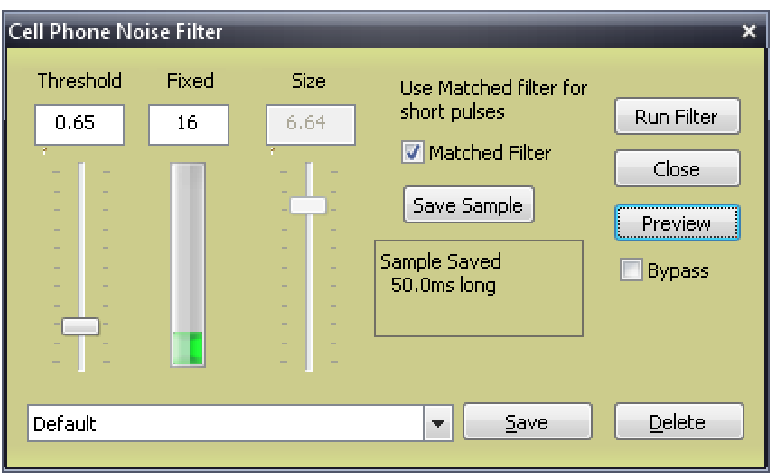
Primary Compensation Issue with the DSS Filter in Non Real-Time Situations:

The primary difficulty encountered when using the DSS filter in non-real time applications arises from the lack of acoustical matching between the Target Forensics recording and the re-created Reference Track. In other words, room resonance, frequency response or natural room reverb may not exactly match your re-created Reference Track. Performing some pre-processing on your Reference Track can compensate for these acoustical mismatches. You can use the 20-band or paragraphic equalizer to match the resonance and frequency response of the room. Also, you can use the Reverb to simulate the acoustical reflection characteristics of the venue. These steps will rely on your own sense of hearing to create the match. When you listen to the music on the Target Forensics recording, focus your listening on its musical content. Then try to create that same sound on your re-created reference track using the above-mentioned tools. Then use this pre-processed track as your final Reference Track to be applied to the DSS filter.

<u>Cell Phone Noise Filter</u>

(Forensic Version Only)

Some cell phone systems can interfere with audio equipment despite their use of signals well outside of the audio spectrum. Their carriers operate in the Giga Hertz range (0.850 – 1.9 GHz). GSM (Global System for Mobile Communications) based cell phones send out their packets of data in short bursts. These RF (Radio Frequency) bursts (pulses) can be radiated into audio system front end amplification devices like bipolar junction transistors or field effect transistors found discretely or inside operational amplifiers. The non-linear nature of these devices coupled with the integrating effect of collector to base (in BJT based circuits) or drain to gate (in FET based circuits) Miller capacitance creates a parasitic AM (Amplitude Modulation) de-modulator circuit. Thus, the envelopes of these RF bursts of energy are de-modulated into the audio range of frequencies. The repetition rate of these RF bursts lie within the audio spectrum. Thus, they can be detected and heard in an audio signal chain that is not sufficiently shielded. Because these noise bursts become audible, they can render an audio signal pathway extremely noisy, contaminated with a “buzz” like sound and thus, often unintelligible. The Diamond Cut Cell Phone Noise Filter is designed to attenuate the noise found in this situation.



The Cell Phone Noise Filter

The filter has two modes of operation. The first and more basic mode is more automatic, but often less effective while the second mode (Matched Filter) is more discriminating. When the Matched Filter is not checked, the system automatically attempts to find the cell phone noise pulses and interpolate them out of the audio signal. When the “Matched Filter” box is checked, you will need to highlight a sample of one of the noise pulses in the time domain display of the software and then click on “Save Sample”. The system will then use a time domain technique of pattern matching to detect further pulses in your audio stream and then replace them with interpolated audio.

Automatic (Non-Matched Filter) Mode

Un-check the “Matched Filter” checkbox. Click on the Preview button and then adjust the ratio control until the pulses are detected and reduced in amplitude. Then, adjust the “Size” control for the best overall signal intelligibility. Size relates to the length of the noise burst and not its amplitude.

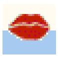
Matched Filter Mode

Check the checkbox labeled “Matched Filter”. Zoom-in on a chain of cell phone pulses in the time domain display. Then, highlight a single pulse event making sure to capture the entire event. Do not highlight a train of pulses, just a single impulsive event. Be sure not to highlight any of the voice signal surrounding the noise pulse or burst. Next, click on the “Save Sample” button of the Cell Phone Noise Filter dialog box. Zoom back out on the time domain display and then hit “Preview”. Adjust the Ratio for the best level of detection (noise reduction). Note that the “Avg Time” control is not active in Matched Filter Mode.

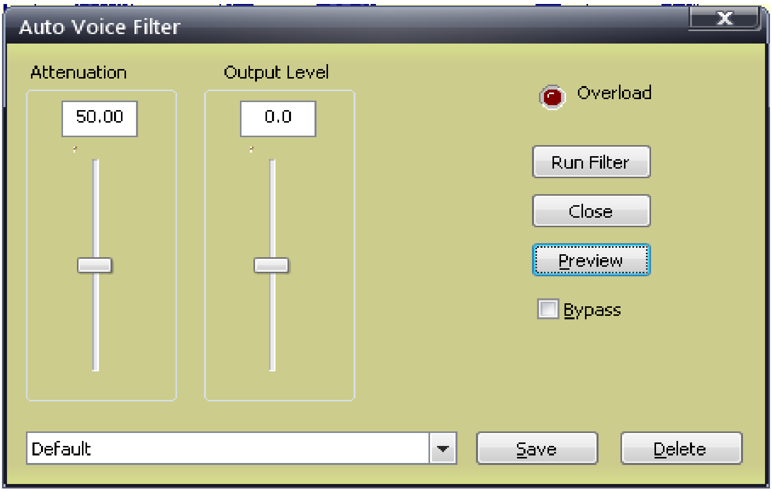
Note: This filter works best after your file has been up-sampled to 44.1 kHz, 16 bits.

Auto Voice Filter

(Forensics Version Only)



The Auto Voice Filter is an adaptive system optimized for separating voice signals from random noise automatically. It uses two independent mathematical processes that work in harmony with one another in order to perform its job. One process identifies statistical noise while the other process identifies the human voice component of a signal. The Auto Voice Filter uses both of those pieces of information to separate the voice from the noise using sort of a "push-pull" methodology. The Auto Voice filter will continually adapt itself to varying signal to noise ratio conditions. It can improve the signal to noise ratio of spoken word recordings. It can also be used to reduce room reverberation (de-verb) sound on speech recording made in large venues. To operate this filter, merely adjust the "Attenuation" slider control for the optimal signal quality while previewing. The optimum quality will be found as the best balance between noise and/or reverberation reduction and speech intelligibility. Moving the attenuation slider upwards raises the aggressiveness of the filter and vice versa. Adjust the "Output Level" control such that the "Overload" LED does not light up on transient signals in order to minimize the creation of distortion. When you have discovered the best settings, highlight the entire sector of the file in need of filtering and then click "Run". As with other Diamond Cut filters, you can save your favorite presets for future recall. This filter is limited to files using sample rates between 22.05 kHz to 48 kHz. If your file uses a different sampling rate, convert it to 44.1 kHz first before attempting to use the Auto Voice Filter. It works best with 16 bit resolution files. The Auto Voice Filter is monophonic only. Stereophonic files are summed to monophonic first and then processed. The resultant processed output file is presented in a dual-channel monophonic format.



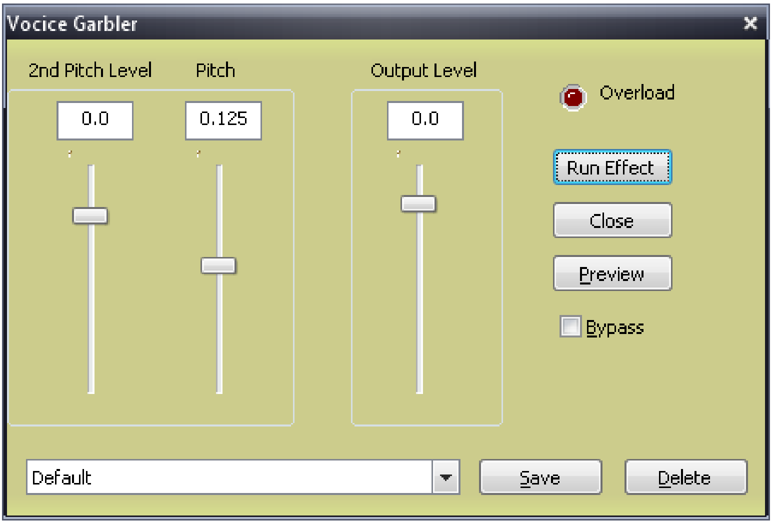
The Auto Voice Filter

Note: This filter is optimized for Forensic Audio only; as such, it is not what one might call a "High Fidelity" system.

Voice Garbler

(Forensic Version Only)

Sometimes it is necessary to disguise a person's voice in order not to break cover or to protect a person's anonymity that may be providing certain types of legal testimony. Your Diamond Cut Forensics10 Audio Laboratory software includes a Voice Garbler (voice disguiser or voice garbler) for that purpose.



The Vocal Garbler

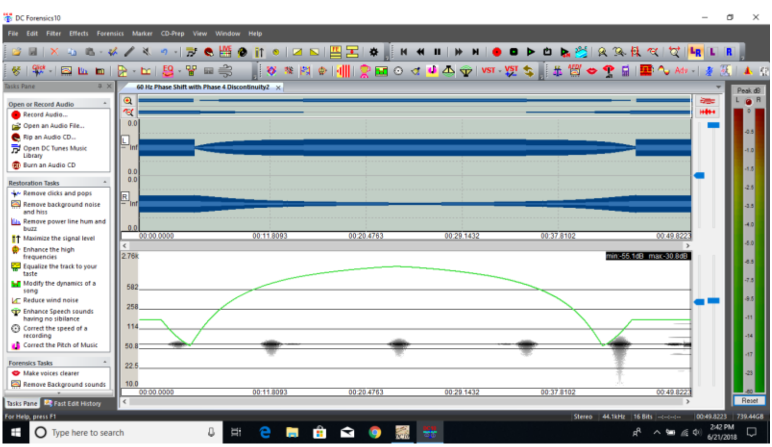
The algorithm used renders the garbled voice almost impossible to reverse when the controls are set properly. This attribute distinguishes this "garbler" from others that you may have worked with. To help maintain its security, we are not going to reveal anything about the method of operation of this algorithm. As a matter of fact, we are not going to use accurately descriptive naming conventions for the primary controls. They are simply "Pitch" and "2nd Pitch Level". There are no units assigned to the numeric values associated with these controls either. To operate the Voice Garbler, simply adjust "Pitch" and "2nd Pitch" until you achieve the disguised voice effect that you desire. Switch the "Bypass" checkbox on and off as you vary the controls until the persons voice can no longer be uniquely distinguishable while maintaining the intelligibility of the content of the speech. The only accurately descriptive control is the "Output Level". Adjust this so that the Clip LED does not illuminate if you are concerned about output signal clipping. As with other Diamond Cut filters and/or effects, you can save your settings by clicking on the Voice Garbler "Save" button.

Note: The Voice Garbler is not a high-fidelity system. Glitches (discontinuities in the signal) are a normal byproduct of its method of operation and are an element (artifact) of its non-reversibility.

<u>View Channel Phase vs Time</u>

(Forensics version only)

A useful "cousin" plot of the X-Y averaging display is the "View Channel Phase vs Time" function found under the Forensics menu. We recommend reading the section on the X-Y plotter to help gain some insight into all channel phase related phenomenon before using this feature. The channel phase vs time routine plots the <u>average</u> phase angle between the two audio channels vs time which can be useful in Forensics authentication applications. The vertical axis (Y) represents the average phase between two audio channels (with a span from 0 to 90 degrees) while the horizontal (X) axis represents time. The middle of the Y axis represents an average phase angle of 45 degrees. The intensity (Z Axis) of the green graph line is intensity modulated as a function of signal magnitude. This system requires a two–channel signal to perform its function and will not function with single channel signals. This plot overlies (on top of) the spectrogram and is shown via a green line. Sometimes, it may be found to be useful to bring up the X-Y plotter (View Menu) while using this tool since it also shows the instantaneous phase of the two audio channels thereby giving you additional information about the file being investigated.



Average Channel Phase vs Time Graph (green line)

Discontinuities in the phase vs time plot may be worthy of further inspection. The graph may relate to the coherence of the two signals or they could indicate a possible edit in the stereo file or people changing position in the venue or even a movement of the position of the recorder. When using this in Classic Edit mode, you can see the time domain signal directly above the phase vs time plot and correlate discontinuities directly to the time domain. You can zoom-in onto the time domain graph or the Channel Phase Graph where these anomalies occur and study those areas of the file for potential authentication related issues. The two views will track each other via the synchronization function, so you can zoom-in or zoom-out via either display.

Procedure:

Note: As with most Diamond Cut Productions Forensics Audio Tools, the best results will be had on files that are 44.1 kHz, 16 bits. If your file is not of that format, change it before proceeding. The file sample rate and bit depth converter are found under the Edit Menu/Change Sample Rate/Resolution.

1. Place the system in Classic Edit* mode (via the FE icon on the system toolbar near the top of the software program). Also, place the system in Source and Destination Sync Files mode (View Menu or Sync Icon).

2. Bring up a stereo (or 2 channel) file of interest in the Source Workspace. Do not bring up a single channel file. This algorithm will only work with two channel files. If only one channel is found, no phase vs time graph will be plotted.

3. You should see two signal lines in the Source Window display and nothing in the Destination display Window.

4. Go to the Edit/Preferences function and then click on the Spectrogram tab.

5. For the <u>best</u> contrast and visibility of the green phase graph, select the Gray-scale Color Palette for the Spectrogram. Other palettes are

possible however with poorer visibility of the Channel Phase vs Time graph.

6. Adjust all of the remaining Spectrogram parameters to your liking.

7. Go to the Forensics Menu and click on the "View Channel Phase vs Time" function (near the bottom of the menu listing)

8. After a few moments, the system will construct both the spectrogram and the phase line (in green) in the Destination display.

9. Look for discontinuities in the phase graph. If you see something suspicious, zoom in on that area of the file in the time domain (source) display. To turn off the phase vs time display, click on the spectrogram icon (button with red lines running through it) in the upper right corner of the Source Display window, or click on the View Channel Phase vs Time function (under the Forensics Menu) again. It is essentially a “toggle-type” selector function.

*Note 1: It is also possible to use this feature in Fast Edit mode if that is preferable by the user.
Note 2: Two demo files are provided to help in understanding this feature. “Phase vs Time speech demo with edits” & “60 Hz Phase Shift with 4 Phase Discontinuities”

Summary of Basic Average Phase vs Time Waveforms

0 degrees – phase vs time line at bottom of the graph (probably one channel only is modulated)

90 degrees – phase vs time line at top of the graph (probably one channel only is modulated)

45 degrees – coherent signals (both channels modulated)

45 degrees with variance thereof – normal stereo signal

0 to 90 degrees – high variance of signal coherence

0 to 90 degrees with discontinuities or step functions – suspicious signals.

The following are descriptions both of the phase vs. time demo files:

1. Phase vs time Speech Demo
   This file was made with a binaural (two-channel) microphone in a 36 foot long venue. The microphone was located on a stand at the half-way point (18 feet). Two people were located in the room, one at each end. The female voice remains physically stationary at the left side of the room, while the male (voice) slowly walks from the right side of the room and ends up on the left side of the room by the end of the recording (both persons being on the left side of the room by the end of the recording). The recording is of Lincoln's Gettysburg address, with sections of the speech alternately being spoken by the male and female starting out with the male voice.   If you bring up this file in classic edit mode and file sync mode and then click on the "View Channel Phase vs Time" feature (found at the bottom of the Forensics Menu), a green line will be displayed atop the spectrogram view. Note how the line has a slight positive slope (up and to the right) indicating the movement of the male voice in the room. Also, note that there are 4 edits points in the file, with two located amidst the speech portion. These are areas of interest of potential file tampering and need to be further explored via the software's zoom function to evaluate what is going on at those positions.

2. 60 Hz Phase Shift Demo
   This file was created using the software. A 60 Hz, two channel signal was created using the Make Waves Generator (Edit Menu). It was then fed through the filter sweeper one channel at a time using the HP and LP functions set an octave away from the 60 Hz signal thus creating a variable phase shift of one 60 Hz signal compared to the other. This is intended to simulate an acoustical environment in which some 60 Hz acoustical noise is present in the venue (from things like HVAC systems, pumps, or large transformers) and is picked

up by the stereo recorder. In practice, a pre-conditioning step would be included to low pass filter everything below 100 Hz so as to only allow the 60 Hz room noise to be recorded. Bring up this demo file along with the “View Channel Phase vs Time” feature. Note that a non-linear line is painted on the screen with discontinuities thereon. These discontinuities are areas of the file that have been modified and the edits are clearly shown in the phase vs time graph.

Additional Forensics Features

There are additional tools that will be of value for forensics work which are not found in the Forensics menu. Some of them are as follows:

1. Automatic Level Control (ALC or AGC) (Dynamics Processor):

Sometimes, these algorithms or systems are referred to as Automatic Gain Controls or AGC's. This feature provides upward expansion of signals below the threshold line and downward compression of signals above the same threshold. This feature is useful in forensics applications where there is a large variation in signal levels between several different parties that may be communicating with one another. It is also useful for the broadcast of live sporting events (if you have the DC Forensics10 version of the product) in which the crowd reaction is of interest when the announcer is not speaking. Simply clicking on the ALC box in the Dynamics Processor activates this feature. The threshold, attack, and release controls are still active when this function is invoked. Many Forensics specialists have also found the automatic level control feature (ALC) in the Dynamics Processor to be very useful in their work. This feature is particularly useful for correcting near-party / far-party loudness discrepancies of recorded phone conversations and other situations where signals vary greatly from time to time.

2. Punch and Crunch Effect operating in Variable Freq Mode:

This is useful for improving the intelligibility of signals having very small values of dynamic range in the speech portion of the spectrum. It is advised that a speech filter is run first before applying this filter for enhancement. For more information, please refer to the section of this documentation that describes the Punch and Crunch effect.

3. Continuous Noise Filter:
This can be very useful for decreasing "in-band" noise found on Forensics recordings. A noise fingerprint will have to be taken from a section of the recording having only noise and no signal in order to use this filter. For more details, refer to the section on the Continuous Noise Filter.

Note: The Continuous Noise Filter in Auto-Spectrum CNF mode can often produce good results on high quality forensics audio signals that need only slight noise reduction in changing noise environments. Compare it with the results produced by the CNF Forensic AFDF mode. The Auto Spectrum CNF mode will maintain a higher level of "fidelity" compared to the AFDF mode, but will not be as effective in extremely noise situations.

4. Harmonic Reject Filter:
This is very useful for attenuating buzzes found on some recorded phone conversations. Several passes through this filter are often required for severe noise situations. Also, it is worth trying both odd and even harmonics of the fundamental buzz frequency. The fundamental buzz frequency can be found by using the Spectrum Analyzer located under the "View" menu.

5. EZ Impulse Filter:
This is very useful for eliminating static from body mike recordings transmitted via an RF carrier. It can also be useful for reducing line frequency related buzz consisting of narrow spikes having a repetition rate of 50 or 60 Hz.

6. Slot (which is a subset of the Notch) Filter:
This can be useful for isolating specific sounds buried in a recording.

7. Median Filter:
This is useful for bringing out the subtle articulations of speech in low-bandwidth or muffled voice recordings, especially when used in conjunction with its "Weighting" function.

8. Spectrum Analyzer in "High Resolution" mode:
This is useful for tape authenticity verification and other forensics analysis functions.

9. Source Time Display Window in conjunction with the Time Display found in the View Menu and the Markers found in the Marker Menu:
These are useful for precisely measuring the time between multiple events such as gunshots.

10. Live Mode is useful for surveillance situations in which signals must be cleaned up "real time."
Please refer to the Multi-filter section of this documentation for details concerning the real-time capability of the Forensics Version of this software.

11. **Live Log to Disk Mode:** This is useful for recording surveillance conversations selectively while in Live, real time mode (Multi-filter).

12. **High Precision Spectrum Analyzer** (Please refer to the section dedicated to this feature for details.)

13. **30 Band IIR Based Graphic EQ** (Please refer to the section dedicated to this feature for details.)

14. **Forensics Adaptive Frequency Domain Filter (Forensics AFDF)** This is the twin sister of the "Adaptive Filter" found in the Forensics menu, but uses Frequency Domain rather than Time Domain techniques to achieve its end. For details, please refer to the section in the CNF description dedicated to this function. The AFDF is found as one of the modes in the CNF within the Forensics version of this software.

15. **Spectral Inverse Filter –** This is useful for improving the intelligibility of muted or muffled recordings and is part of the Spectral Filter.

16. **Cell Phone Noise Interference Attenuator –** This is useful to reduce cell phone interference on electronic recording electronics and is found as a set of presets in the DC Forensics Multifilter.

17. **Overtone Synthesizer –** The Overtone Synthesizer in the Forensics version of this software uses a different set of internal filters and ranges compared to the standard version making this effect more useful for enhancing the intelligibility of garbled speech. Its range of frequency settings overlap that of the standard version (DCArt10) and so it will do things similar to that which the standard version does and more. The lower limit of the frequency setting is reduced from 3,500 Hz down to 2,000 Hz.

18. **Expert Impulse Filter –** This can also be used to eliminate 2-way radio static or very high frequency buzz which is at the power line frequency repetition rate (like the sound of a phase controlled light dimmer interfering with an audio system). The Vinyl LP setting is a good place to start with and the size set to 10 (on 44.1 kHz material), Threshold set to 1 and the Tracking set to around 50. Adjust the Tracking for optimal noise rejection and minimal distortion.

19. **Forensics Demo** - The Forensics Demo Wavefile includes a Multifilter preset using multiple (8) sub-filters cascaded together in order to clean up the audio signal. It is quite instructive to test this multifilter preset on the Forensics Demo Wavefile using the Mutifilter preset titled “Forensics Demo Clean-up Filter”. Note: (This file is an air-check disc recording made on 3-4-1933 of FDRs 1st Inauguration speech which is buried in noise. It was recorded on a relatively low cost home RCA pre-grooved disc recorder located at a residence in Newark, NJ from AM radio station WEAF out of NYC on 660 KHz.)

The Marker Menu

DCArt10/DC Forensics10 provides you with movable red timing markers that can be placed within either the Source or Destination workspace. They are the .wav file equivalents to bookmarks. You will find them to be useful tools when utilizing such editing features as the Copy, Cut, Mute, etc. The Marker menu is also complimentary to the CD Prep menu. It is very useful when you need to take a very long .wav file and Chop it up into smaller files manually for indexing onto a CD-R. Markers are stored as part of the Diamond Cut peak file (.pkf). The following commands are provided under the Marker Menu:

Add Markers:

This feature adds up to 100 markers to the desired Workspace. Clicking here will drop a marker at the beginning of a selected area. If no area is selected, by default, the marker is dropped at the far left of the .wav file. This command can also be accomplished by use of the mouse right click menu.

Clear All Markers:

This feature erases all existing markers from the Diamond Cut peak file (.pkf).

Highlight Marked Area:

This feature highlights in Yellow (or the color of your preference) the area of a .wav file located between two previously defined Red markers.

Drop A Marker:

This command allows you to drop a marker during playback so you can audition your audio while "marking it up" for in-depth study later. Simply hitting the "M" key on your keyboard during playback can also drop markers. When selecting (highlighting) an area, the marker dropped by the "M" key occurs at the end of the swipe. For example, if swiping from left to right, the marker is then dropped by the "M" key at the right side of the selection. If swiping from right to left, the marker will be dropped by the “M” key at the beginning of the selection.

Go To Next Marker:

Exactly as the name implies…allows you to quickly navigate to the next marker on the right. This can also be accomplished by simply pressing the “N” key on your keyboard.

Go To Previous Marker:

Same as above, only this command takes you to the previous marker on the left. This can also be accomplished by simply pressing the Shift + “N” keys on your keyboard.

Re-Number Markers:

If your markers have become out of order, you can have the system automatically re-order them numerically by clicking on this function. This is useful if you are going to use the marker numbers as individual track number indicators.

Label Marker:

By default, DCArt10/DC Forensics10 just places a number on each marker. If you use several markers, it may be necessary to label them more descriptively. Select the marker you want to label by left clicking on it (it will change from Red to Black). The Label Marker command allows you to be more precise with your markers; right clicking with your mouse and activating the Label Marker command can also accomplish this same task.

Lock Markers:

To lock the position of your markers, simply click on this menu item and your markers cannot be accidentally moved while you’re working on your file. In order to move them, you’ll need to press the Ctrl button plus drag the mouse. You can still remove markers if desired; this feature is a positional lock, not a marker erasure protection feature.

Merge Source Markers into Destination:

There are situations where it may be desirable to transmit the marker locations from a Source file into a Destination file when working in Classic Edit Mode. Clicking on this feature will facilitate this operation when markers are present in a Source File and a file of equal length is present in the Destination time display window.

The CD Prep Menu

This menu contains the tools that help you prepare for the final stages of your restoration project. The CD Prep Menu does everything but copy your files to CD.

Quantize for CD Audio

This feature moves a marker to a multiple of 2,352 Bytes to provide compatibility with CD data grouping so that glitch-less indexing can occur. This feature is particularly useful when chopping a large (continuous concert type) .wav file into pieces for transfer to CD-R. If you have a .wav file open, merely click on this menu item and the file will be properly quantized.

Using Quantize on a Single File (Tutorial)

1. Bring up desired .wav file in the Source Workspace
2. Click on the CD prep menu item.
3. Click on Quantize for CD Audio

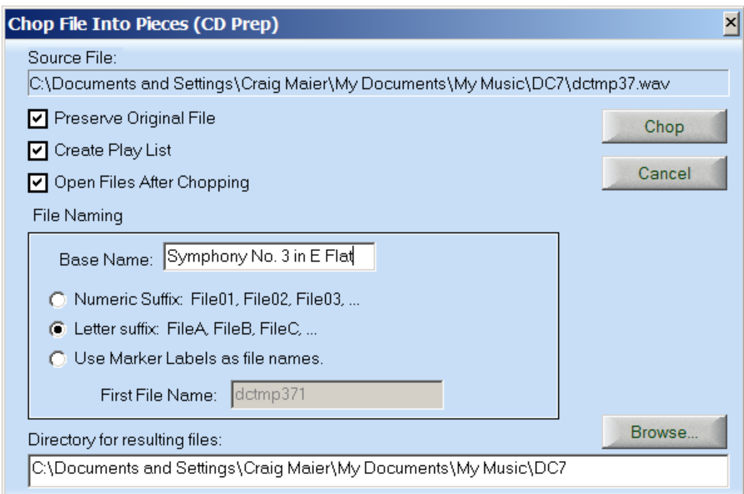
Using Quantize and Chop File on Large File (Tutorial)

1. Bring up desired .wav file in the Source Workspace
2. Place markers at all of the locations at which you desire to break the file into individual .wav files. These are usually placed at the silent portions of a recording, between cuts.
3. Click on the CD prep menu item.

4. Click on "Quantize for CD Audio". All of the markers shall be moved into the proper positions for CD quantization.
5. Chop file into pieces using the command by the same name.

Chop File into Pieces

This command breaks long .wav files into smaller .wav files as defined by the locations of your various markers. This command will remain grayed out until your file has markers placed.



The Chop File Into Pieces Dialog Box

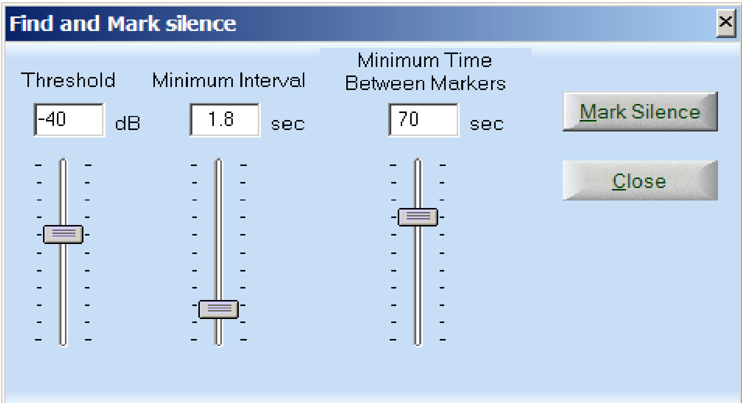
If you've chosen to record a full album length file onto the hard drive and are now ready to break the large file into the smaller "song" files, this feature, used in conjunction with either your markers (manually finding the beginning and ending of your "songs") or with the Find and Mark Silent Passages tool (automatically finding the song tracks by the silence in between two songs…described below) allows you to accomplish this task. It allows you to choose from any or all of the following options:

- **Preserve Original File:** After you've 'broken up" your main .wav file, do you still want to keep it as a master? This option give you that choice.

- **Create Play List:** This launches the Play List editor, which allows you to set up your CD's track playing order. It allows you to add, audition or remove specific titles. The play list also allows you to export your cue sheet for other CD burners in the .cue format. For more info on the playlist, refer to the earlier chapter on the File Menu.
- **Open Files after Chopping:** In most cases, you'll want to audition your "cuts" for accurate beginnings, endings, volume continuity, etc. before actually burning your CD. This checkbox automatically opens your new "cuts" so that review process can take place.
- **File Naming:** This will list the base name of your Master .wav file and then you can choose whether the resulting "chopped" pieces will be named with letter or number suffixes.
- **Directory for Resulting Files:** Allows you to choose a storage location (or facility) for your resulting files.

Find and Mark Silent Passages

DCArt10/DC Forensics10 includes a feature, which will automatically find and mark the silent passages of your .wav file. This is particularly useful when you desire to process an entire vinyl record album (or tape) in one shot through the various Diamond Cut filter and/or effect algorithms, and then break them up into separate .wav files at the end of the process. You have the ability to select the detector system's threshold of silence, and its time duration for the silence. After you have invoked this feature, you will see all of the markers moved to the quiet area between cuts. You can move the markers manually, if you are not satisfied with the separations that were automatically determined by the program. After this has been completed, you can chop the file into pieces, and separate .wav files will be created.
The "Find and Mark Silent Passages" feature will also be found to be useful for identifying long silent sectors of surveillance recordings automatically. This can save a lot of time for the Forensics examiner who must deal with very long recordings, most of which contains no conversation.



The Find and Mark Silence Controls

A more versatile system is available in the Forensics version of this software called the "Remove Silence Tool". For details, refer to that section of this User's Manual.

Gain Normalize

The "Gain Normalize" feature searches an entire .wav file looking for the peak signal level and established that value as its 0 dB reference point. It can be used in fast edit mode or classic edit mode in the source window. Then, it adjusts the overall gain applied to the file so that the overall level is below that value. This will provide the best signal to noise ratio and a reasonable volume balance for each "cut" on your final master. Gain Normalize should be applied before burning a CD-ROM or making your final tape.

Important Note:

In Classic Editing mode, Gain Normalize will search the entire file for the peak and adjust the gain based on that peak. It will ignore any selected area. In Fast Edit mode, Gain Normalize will abide by a selected area, find the peak there and adjust the selected area only with the appropriate gain normalization.

Normalized Gain Scaling

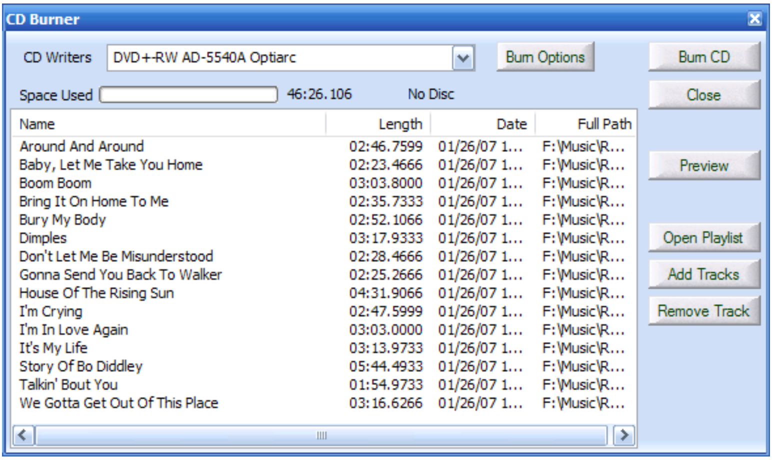
It allows you to scale the gain of a .wav file to values other than 0 dB (full-scale output). The range of adjustment provided is +/- 20 dB, which corresponds to a gain factor range of +/-10X. If you apply gain scaling above 0 dB, some portion(s) of the .wav file will be clipped. This could be useful in a situation wherein a single transient pulse or two are dominant in the .wav files amplitude, and "clipping" it is irrelevant to you. Sometimes, forced clipping (above 0 dB) can provide a useful or interesting effect (sometimes called “overdriving”). If you apply scaling below 0 dB, all resultant signals shall be below full-scale output. Normalize Gain Scaling can be performed separately on each channel depending on whether L/R or L or R channels are chosen. Performing Normalized Gain Scaling on each channel separately can improve the stereo balance of some recordings.

Note 1: Both the “Gain Normalize” and the “Normalize Gain Scaling” functions will query you **“Do you want to save UNDO information for this operation?”** If you say yes to this, the process will take about twice the amount of time that’s required if you choose to answer no.

Note 2: The CD Prep functions can be performed in either the fast edit or the classic edit mode. When using the classic edit mode, some of these functions are only supported for use in the Source Window.

CD Burner

The Diamond Cut CD Burner allows you to create CD’s in conformance with the Red Book Audio CD standard. Its integration into the DCArt10 software package coupled with drag and drop capability from the DC Tune Database simplifies the CD creation process after a restoration has been completed.



The Diamond Cut CD Burner

Song lists can be imported directly into the CD burner from your play lists or from the DC Tune Library, and these can be either in the .wav, .mp3, or .wma format. The software will automatically convert these file formats for compatibility with the Red Book standard. The software will query your CD ROM burner to determine its maximum speed capability and use that value during the burn process.

To import files from a Play list, click on the "Open Playlist" button and then select the files that you wish to burn to your CD –R. The WLS, XML, CUE, M3U and PLS playlist formats are supported by the CD Burner. If you have your DC Tune Library open, just drag and drop the desired sound file(s) into the CD burner. You can add or delete tracks from the list using the "Add Tracks" or "Delete Tracks" button(s). For example, to delete a track, just click on it in the CD burner (to highlight it) and then click on "Delete Track". To hear a particular file in the list before burning the CD, highlight it by clicking on it and then click on "Preview".

When adding tracks, be sure that the "Space Used" graph does not exceed the capacity of the media that you plan to use, or an error will be generated. Place a blank CD-R into your ROM burner and wait a few seconds. A message should appear at the top of the Burner routine indicating “Blank CD-R. If you are satisfied with your song listing, click on the “Burn CD” button to commence the CD burning process. A Progress Bar will appear indicating the progress in terms of “% Done”. The system will report when the process has been successfully completed with the following message:

Success: The CD has been burned successfully. Please remove the disc from the drive. Also, a “Gong” sound will be heard at the completion of a CD burn process.

Burn Options

Clicking on the “Burn Options” reveals a number of user selectable alternatives associated with your CD burner. They are as follows:

Maximum Writing Speed: You can limit the burning speed anywhere from 1X to 24X depending on the characteristics of your optical drive. Sometimes, slower burn speeds improve the accuracy of a CD Burn cycle with the obvious tradeoff (longer burn-time).

Recording Mode: The system is capable of the two primary CD Burning modes including Disc At Once and Track At Once.

Disc At Once Mode (Gapless) Recording: This feature (DAO) is useful for concert or orchestral recordings wherein one does not want to produce any glitches at the point where the track changes during play. A minimum of 2 tracks are required for DAO. After you drop your markers and before you break your file into pieces, you should “Quantize for CD Audio” (found on the CD Prep Menu) so that this feature works seamlessly.

Write CD Text: Incorporates Text into the file header of your CD when checked. The text information comes from the name field in the file list. It could be the track name if it comes from specific files. Note that only certain CD players are compatible with this feature and can take advantage of this mode of operation.

Track At Once: This mode (TAO) is the most common mode in which CDs are burned. It places a gap between each track. The between track time-gap is settable anywhere from 0.013 seconds to

20 seconds. The factory default for this setting is 2 seconds, but it will remember your last data entry value.

Enable Optimal Power Calibration: This optimizes the burn power used to create your CD. This will produce the highest quality and therefore the most robust CD burns. However, it takes a longer period of time to burn a CD when this mode is used.

Note 1: The Diamond Cut CD Burner can import files consisting of the WLS, XML, M3U, and PLS playlist formats.

Note 2: The DCArt10 CD burner defaults to a 2-second gap between each track selection when Track At Once Mode is used.

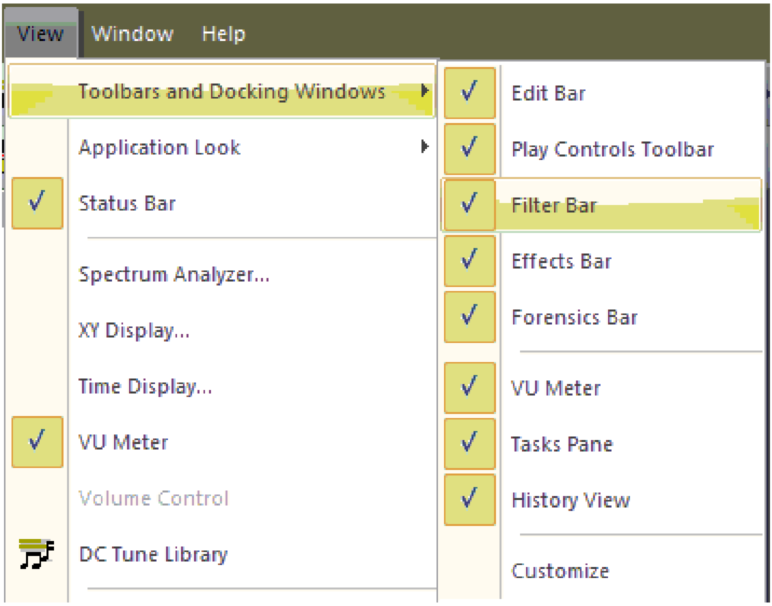
Note 3: Some of the CD burner features are only available in the Forensics version of the product.

The View Menu

The View menu allows you to access the commands that affect the manner in which files, controls, and parametric displays are presented to the user. To activate a particular feature, click on it with the left mouse button. Once it has been activated, a check mark will appear next to the chosen command (feature).

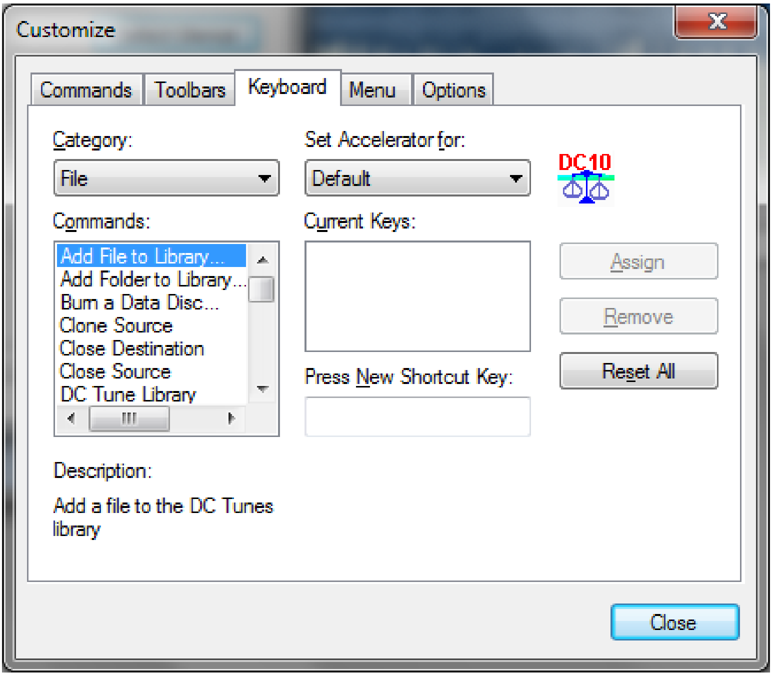
Toolbars and Docking Windows

The Toolbars and Docking Windows provide you with a great deal of flexibility with regard to the look and feel of the user interface. It includes controls over some of the commands, the toolbars, the keyboard assignments, and so forth. Of particular interest are the toolbar selectors. On the most basic level, simply check the box associated with the toolbar that you want to show. Of special interest is your ability to turn on or off the Tasks Pane via it’s checkbox.



Toolbar Selector Dialog Box

Further personalization of the system can be had via the "Customize" checkbox. It brings up numerous options that you can choose between via the Customize dialog box.



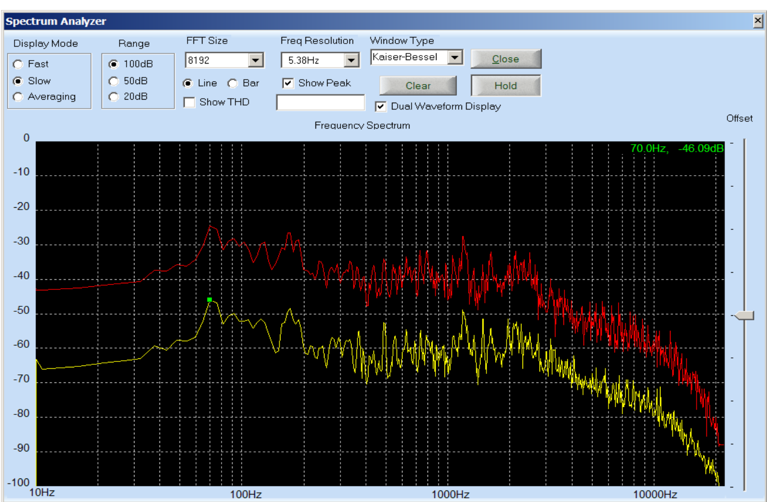
The Customize Dialog Box

The Diamond Cut Spectrum Analyzers

There are two versions of the Diamond Cut Spectrum Analyzer available in the version 10 product family. The DCArt10 version provides a Standard Precision Spectrum Analyzer providing excellent resolution especially at the low frequency end of the audio spectrum. The DC Forensics version provides a High Precision Spectrum Analyzer with higher frequency resolution over the entire audio frequency spectrum coupled with a display which can zoom in on a specific frequency range via a start and stop frequency feature. The High Precision Spectrum Analyzer also offers the user the option of octave based 10 and 30 band displays for use in acoustical measurement situations.

Spectrum Analyzer – Standard Precision (DCArt10)

The Spectrum Analyzer command in DCArt10 brings up the floating and user sizable Standard Precision Spectrum Analyzer. The Spectrum Analyzer is capable of resolving frequency increments as small as 0.02 Hz on 44.1 KHz sample rate files when it is set appropriately. This Analyzer can be used with any of the filters or effects. It is connected to the output of the filter or effect, so that you can see, in the frequency domain, how you have affected the file. If you want to compare the output of the filter or effect to the input, use the Bypass function on the filter or effect window. Since the Diamond Cut Spectrum Analyzer utilizes constant Hertz per frequency band, it will display white noise as a flat, horizontal line. This is unlike octave weighted real time audio analyzers in which white noise produces a diagonal, positive sloped line, and pink noise produces a flat horizontal line. Conversely, pink noise displayed on the Diamond Cut Spectrum Analyzer will be displayed as a negative sloped diagonal line. Normally, the Spectrum Analyzer displays the algebraic sum of the left and right channels. The Dual Waveform Display option will provide discrete displays for each channel. Keep in mind that the use of the Spectrum Analyzer will slow your system down slightly. Therefore, when you are done using it, shut it down.



The Standard Precision Spectrum Analyzer

The following displays are provided on the Spectrum Analyzer:

- **Frequency vs. Amplitude Graph:** The vertical axis indicates 0 dB at the top and ranges down to -100 dB at the bottom by default. The amplitude range can be changed via the Amp Range control. The horizontal axis indicates frequencies from 10 Hz (left) to a little over 20 kHz (right).

- **Two digital readouts indicate the frequency and amplitude of signals feeding the Spectrum Analyzer.** One signal is the peak value and the other is user defined.

The following controls are provided:

- **Display Mode:**

    - **Fast:** Shows the spectrum in almost real time.

    - **Slow:** Shows the spectrum with slower ballistics.

    - **Averaging: On/Off:** This allows the system to provide you with the average signal spectrum rather than in "Fast" real time display. The averaging time interval will be as long as the analyzer is left in operation or until the "Clear" button is activated. The looped play function is sometimes useful in conjunction with the Averaging function.

    - **Dual Waveform Display:** This checkbox allows you to view both the left and right channels simultaneously on the Spectrum Analyzer display provided that you are working with a stereo .wav file. The right channel information will be displayed in red while the left channel is displayed in yellow. If you operate this feature with a monophonic .wav file input, the display will be shown in yellow.

- **FFT Size:** 11 selection alternatives (64, 128, 256, 512, 1,024, 2,048, 4,096, 8,192, 16,384, 32,768, & 65,536). This selection determines the frequency resolution of the spectrum analyzer. The higher the number selected, the better the frequency resolution of the display. The Forensics version extends the FFT Size range up to 131,072.

- **Range:** (Amp Range) The Amplitude Range control scales the vertical axis to 150 dB, 100 dB, 50 dB, or 20 dB for full scale deflection. This feature along with the offset control allows you to hone in on a particular signal.

- **Hold Button:** This is a "toggle" function and will allow you to freeze or un-freeze the spectral display update.

- **Clear Button:** This clears the display. It is especially useful when using averaging mode so that you can clear the display after a long averaging interval, allowing you to move onto a new area of the Waveform to be averaged. Clearing the display will re-initiate the averaging process as well as clearing the display.

- **Show Peak Button:** This feature will automatically find the peak amplitude signal and display its Frequency and Relative Amplitude value in the upper right hand corner of the Spectrum Display screen. The marker and display for this feature are red in color.

- **User Controlled Marker:** You can place a marker anywhere you want on the spectral display by clicking the left mouse button on the signal that you are interested in measuring. To accomplish this, merely point the mouse cursor to the peak of interest and click the left mouse button. A magenta colored marker line will appear at that location and a yellow digital display of the frequency and relative amplitude of the signal that you pointed to will appear in the upper left hand corner of the spectral display. To read another value, merely click the mouse again after pointing to a new spectral line. The marker and the display will then be updated.

- **Frequency Resolution:** (Standard Resolution version – non-Forensics.) The Spectrum Analyzer has the ability to display frequencies with the following values of resolution as indicated on the chart below. Note the interaction between the FFT Size selected and the available Resolution in Hz that are available: “OK” indicates available ranges. “NA” indicates Not Available resolution values by Frequency Band. It is very important to note that small values of FFT Size produce poor values of Resolution, but the response will be very fast. Conversely, large values of FFT Size produce good values of Resolution, but the response will be very slow. This latency time (or display refresh / update) is a function of the Spectrum Analyzer frequency resolution setting. The latency expressed in Seconds, is displayed in the right-hand column for each value of resolution.

Spectrum Analyzer Resolution Chart

FFT Size				
	64	**128**	**256**	**512**
Res. Hz				
689.06	**OK**	NA	NA	NA
344.53	**OK**	**OK**	NA	NA
172.27	**OK**	**OK**	**OK**	NA
86.13	**OK**	**OK**	**OK**	**OK**
43.07	**OK**	**OK**	**OK**	**OK**
21.53	**OK**	**OK**	**OK**	**OK**
10.77	NA	**OK**	**OK**	**OK**
5.38	NA	NA	**OK**	**OK**
2.96	NA	NA	NA	**OK**
1.35	NA	NA	NA	NA
0.68	NA	NA	NA	NA
0.34	NA	NA	NA	NA
0.17	NA	NA	NA	NA
0.08	NA	NA	NA	NA
0.04	NA	NA	NA	NA
0.02	NA	NA	NA	NA

							Latency
1,024	**2,048**	**4,096**	**8.192**	**16,384**	**32,768**	**65,536**	**(In Secs)**
NA	NA	NA	NA	NA	NA	NA	**.002**
NA	NA	NA	NA	NA	NA	NA	**.0039**
NA	NA	NA	NA	NA	NA	NA	**.0078**
NA	NA	NA	NA	NA	NA	NA	**.0156**
OK	NA	NA	NA	NA	NA	NA	**.0312**
OK	**OK**	NA	NA	NA	NA	NA	**.0625**
OK	**OK**	**OK**	NA	NA	NA	NA	**.125**
OK	**OK**	**OK**	**OK**	NA	NA	NA	**.250**
OK	**OK**	**OK**	**OK**	**OK**	NA	NA	**.5**
OK	**OK**	**OK**	**OK**	**OK**	**OK**	NA	**1.0**
NA	**OK**	**OK**	**OK**	**OK**	**OK**	**OK**	**2.0**
NA	NA	**OK**	**OK**	**OK**	**OK**	**OK**	**4.0**
NA	NA	NA	**OK**	**OK**	**OK**	**OK**	**8.0**
NA	NA	NA	NA	**OK**	**OK**	**OK**	**16**
NA	NA	NA	NA	NA	**OK**	**OK**	**32**
NA	NA	NA	NA	NA	NA	**OK**	**64**

Note 1: Because the FFT creates solutions in both the real and imaginary planes, the actual number of frequency bands created is the FFT Size/2.
Note 2: NA = Not Applicable

- **Offset Slider Control:** This allows you to move the centering of the spectral display up or down. It is of particular value when the "Range" control is set to a high sensitivity value such as 50 dB or 20 dB, and the signal appears to be off of the screen. By using the Range control and the "Offset Slider" control, you can zoom-in on a signal of interest.

- **Window Selection:** Seven window selections are provided so that you can establish the appropriate tradeoff between Stop Band attenuation and Lobe Width.

    A. Blackman

B. Hanning
C. Hamming
D. Rectangular
E. Kaiser 10
F. Kaiser 20
G. Kaiser-Bessel
H. Triangular
I. Bessel
J. Welsh

The following chart shows the performance of several of the provided Window functions. Others are provided with the software for completeness.

Window Name	**Side-Lobe Peak Amplitude in dB**	**Main Lobe Band-pass Width**	**Minimum Stop-Band Attenuation in dB**
Blackman	-57	12 pi / N	-74
Hanning	-31	8 pi / N	-44
Hamming	-43	8 pi / N	-53
Rectangular	-13	4 pi / N	-21

Window Name	Characteristics	Application
Blackman	Exhibits the best amplitude resolution and accuracy, but has the poorest frequency resolution	Very Good for measuring single frequency signals in order to look at its higher order products.
Hanning	This exhibits fair frequency resolution but poorer amplitude accuracy compared to the Rectangular Window	Good for measuring Narrow Band random Noise, measuring Periodic Signals and for measuring transient bursts wherein the signal levels prior to and just after the event are much different in amplitude.
Hamming	This window is very similar to Hanning, however it exhibits slightly better frequency resolution and accuracy compared to Hanning	Same Applications as Hanning.
Rectangular	Exhibits the best frequency resolution, but has the poorest amplitude resolution and accuracy. In essence, Rectangular is not a window, per se, at all.	Good for measuring sine waves that have similar amplitudes and whose frequencies are loose, for measuring broad band noise with slowly varying frequency spectrum and for measuring Transient bursts wherein the signal levels before and after the key event are similar in amplitude

Note: Hanning is the recommended default window to use for general applications.

- **Display Type (Option Menu):** Choose between “Line” or “Bar” (Graph) modes. Default is “Line”. When “Bar” is activated, the display reverts to that mode.

- **Display Size (Option Menu):** The overall physical dimensions of the Spectrum Analyzer are user sizable. Simply use your mouse to drag its horizontal and vertical margins to create a display size to your liking.

- **Show Peak (Option Menu):** Automatically finds and displays the largest signal spike on the Spectrum Analyzer display.

- **Dual Waveform Display (Option Menu):** Reverts to stereo display mode showing each channel as separate traces on the graph. The Right Channel is shown in Red while the Left is shown in Yellow.

- **Show Power Spectrum (Option Menu):** This option displays the power x time product (energy) of the presented signal at various frequencies.

Distortion Analyzer (THD Mode in the Option Menu)

This selector box (Show THD) found under the Option Menu converts the Spectrum Analyzer into a high performance Harmonic Distortion Analyzer when enabled. It measures % THD (percent Total Harmonic Distortion). A pure Sine wave obtained from either a very high quality external signal generator or the "Make Waves" generator must be used as the stimulus for an external device under test (DUT) or algorithm under test. The THD meter measures the fundamental signal component separating it from the rest of the total harmonic signals presented to the system. It also "notches" out the fundamental and sums the remaining harmonics. Then it takes the ratio of the fundamental signal component to the square root of the sums of the squares of the measured harmonics and expresses it as a percentage on the digital readout. Even and Odd order distortion components are produced by transfer function non-linearity, both of which are measured by this system. Even-multiple harmonic products are dominantly due to transfer function asymmetry.

Note 1: The Show Peak checkbox must be checked before enabling the THD display.

Note 2: When not using the Distortion Analyzer, turn it off to minimize CPU utilization.

Using the High Resolution Mode:

Set the FFT Size to 4,096 (or greater). If you then click on 0.67 Hz, the system will re-scale the horizontal (X) axis of the spectral graph to indicate 1,380 Hz full scale rather than 20 kHz when you run the Spectrum Analyzer. You can resolve frequency increments as small as 0.02 Hz as indicated by the settings shown on the resolution chart. To augment this capability, use the “Range” control setting set to 20 dB and the “Offset” control to zoom in on the signals of interest. This combination of features are particularly useful when trying to determine if a forensics recording is a "dub" of the original by looking for two discrete "hum" frequencies on the spectrum. For more information on tape authentication, please refer to the “How do I?” section of this user’s manual.

Important Note:
When using the Spectrum Analyzer in High Resolution mode, it will take a long time for the system to integrate. A message will appear at the bottom of the display during this interval stating “Sampling, Please Wait”. Refer to the resolution chart for latency details.

Measure the Spectrum of Either L or R Channel(s) (Tutorial)

The Spectrum Analyzer displays the algebraic sum of both channels in normal operation. If you want to obtain the spectrum of only one channel, use the following procedure:

1. Bring up the Spectrum Analyzer.
2. Set its parameters appropriate for the measurements that you wish to make.
3. Bring up the File Conversions feature found under the Filter menu.
4. Choose Stereo to Left only or Stereo to Right only, depending on the channel of interest.
5. Click on Preview, and only the chosen channel will be presented to the Spectrum Analyzer.

Measuring the %THD of an Electronic component (Tutorial)

The Spectrum Analyzer contains a special feature which allows you to measure the % Total Harmonic Distortion of a piece of electronic equipment, whether it be a CD player or an audio amplifier. It does this with a high degree of accuracy. This measurement will require, however, the use of a professional grade, high performance sound card and/or CD player in order to achieve accurate results. To achieve accurate results, it is necessary that the signal sample rate be at least 44.1 KHz in order to account for all of the Harmonic Distortion products within the audio spectrum. There are two procedures that you can choose from. The first procedure is simple to perform, but less accurate than the second procedure which is more complicated.

Spectrum Analyzer settings for %THD Measurements:

Display Mode: Fast
FFT Size: 16,384
Frequency Resolution: 2.69 Hz
Range: 150 dB
Mode: Bar or Line (your choice)
Window Type: Kaiser-Bessel
Show Peak: Check the checkbox
Show THD: Check the checkbox

Note 1: The Distortion Analyzer feature is accessed via its Options Menu.

Note 2: If you are using the High Precision Spectrum Analyzer found in the Forensics version of the software, set the Start Frequency at 10 Hz and the Stop Frequency at 22,050 Hz.

- **Procedure #1 (Simple Method):**

    1. Create a 10 minute (600 Seconds) Stereo, 16 bit, 44.1 kHz sampled 1000 Hz* Sine waveform .wav file by using the Diamond Cut "Make Waves" feature found under the Edit Menu.

2. Name the file, and make sure you know the path on which it was stored.
3. Use your ROM burner software to make a CD of this .wav file.
4. Connect the output of the CD player to the component under test's input
5. Connect the output of the device under test (DUT) to your sound card's line input.‡
6. Launch DCArt10 or DC Forensics10
7. From the View menu, enable the Spectrum Analyzer.
8. In the Spectrum Analyzer, enable the THD meter feature.
9. While operating in "Live" mode, play the CD.
10. You should see a dominant spectral spike at 1 kHz.
11. The Distortion Analyzer will calculate the %THD and display it in the THD meter window.
12. Done

- **Procedure #2 (High Accuracy Method):**

1. Perform steps 1-3 from the Procedure #1 (Simple Method).
2. Connect the output of the CD player to the line input of your sound card.
6. Launch DCArt10/DC Forensics10.
7. From the View menu, enable the Spectrum Analyzer.
8. In the Spectrum Analyzer, enable the THD meter feature.
9. While operating in "Live" mode, play the CD.
10. You should see a dominant spectral spike at 1 kHz.
11. The Distortion Analyzer will calculate the CD and sound card %THD and display it in the THD meter window. Write this value down and call it %THD(I) (instrumentation).
12. Disconnect the CD player from the line input of the sound card and then plug it into the electronic device under test's (DUT) line input.
13. Connect the output of the Device Under Test (DUT) to your sound cards line input.‡
14. In the Spectrum Analyzer, observe the THD meter feature and write down the %THD indicated. Write this value down and call it %THD(T) (total).
15. Calculate the DUT distortion using the following equation:

%THD(DUT) = (((%THD(T))^2) – ((%THD(I))^2))^1/2

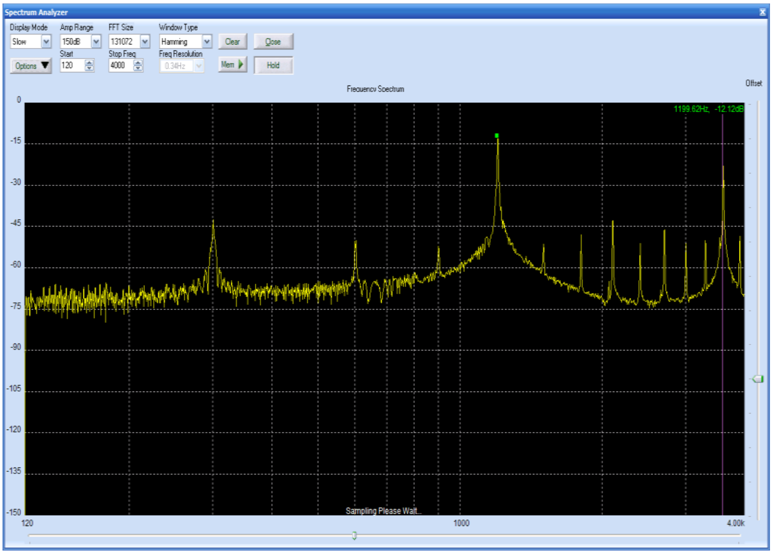
*** Note:** Lower Frequency .wav files can be used.

‡Note: If a power amplifier is being tested, appropriate resistive loading must be connected to its output, along with an appropriate attenuator between the amplifier output and the sound card input. DO NOT connect the output of a high-output audio power amplifier directly to the input of your sound card. Direct connection of the output of an audio power amplifier to a sound card input is a hazard that could result in injury and could result in the destruction of your hardware.

Spectrum Analyzer – High Precision

(DC Forensics Version Only)

A High Precision Spectrum Analyzer is provided in the DC Forensics version of the Diamond Cut version 10 product family (which is sometimes referred to as an FFT analyzer in the Forensics field). It uses techniques including higher FFT sizes, chirp Z-Transforms, decimation methodology, a widened array of window types, waveform storage, and a new graphical display interface in order to achieve its higher level of precision compared to its standard precision counterpart. This high precision spectrum analyzer is useful for making acoustical measurements, authenticating forensics audio recordings by analyzing for potential dubs, verifying that forensics recordings have not been edited, assuring compliance with OSHA Noise Standards (both broadband as well as pure tones), performing mechanical vibration testing, measuring feedback control loop performance, and many other applications.



The High Precision Spectrum Analyzer

The High Precision spectrum analyzer is described here, highlighting the features which are different compared to that of the Standard Precision version found in DCArt10. It is highly recommended that you read the section that precedes this section before learning about the High Precision Analyzer, since many features are shared by both. Here are the salient differences provided by the High Precision Spectrum Analyzer over the Standard Analyzer:

- Added Zoom capability with a Start and Stop setting coupled with Frequency Axis Scrolling Capability
- Added a Memory feature to store and recall spectrums, both Left + Right and individual Left and Right Spectrums
- Added a Vertical Marker Measurement System
- Added Hi-Resolution mode so that all of the FFT bins are applied to the frequency span selected by the user. For example, if you have an FFT size of 4096, and a 1 Octave

span, then all of the FFT bins would be spread over just that single octave presenting the user with a very high frequency resolution display.*

- Added Kaiser 10, Kaiser 20, and Welsh Window types
- Increase FFT Size to 132K maximum
- Added 1.0 Octave and 1/3 Octave Analog Filter based modes for noise and acoustical measurement applications – similar to an analog based Real Time Analyzer (RTA).

***Note:** The selectivity of the High Precision Spectrum Analyzer is still limited by the basic FFT size and the Window choice.

The first difference that you may note between the DCArt10 and the DC Forensics High Precision Spectrum Analyzer is the extensive list of functions provided under the drop-down menu on the left side of the system labeled “Options”. Clicking on this menu provides you with the following options from which to choose.

Option Menu (Forensics Version)

- Show Peak Checkbox – Provides for the automatic measurement and display of the Peak Amplitude Signal level in dB and its Frequency. The two numeric values are shown in Green in the upper right hand corner of the Spectrum Analyzer. Its format is: Frequency in Hz, Level in dB. (Format is Hz, -dB.)
- Dual Waveform Display Checkbox – When this checkbox is not ticked off, the system displays the summed average of both the Left and Right channels. When it is checked, it provides you with the ability to discretely display the Left and Right channels. The Right channel information will be displayed with a Red trace while the Left channel is displayed in Green.

- Bar Graph Mode Checkbox – When this checkbox is unchecked, the system displays each data point with a line drawn in between. When it is checked, the data is represented in a bar graph format.
- Standard FFT Checkbox – This applies the number of bins (FFT Size / 2) to be distributed over the entire audio spectrum.
- Hi Precision FFT Checkbox – This mode allows the number of frequency bins to be applied strictly to the range (or span) of frequencies selected by the user. This mode effectively improves the Frequency resolution of the system when the range of frequencies selected is less than the entire audio spectrum.
- 10 Band Octave Analyzer Checkbox – This changes the mode of operation from an FFT Based system to an IIR based simulation of an Analog, Octave frequency weighted system (like an RTA).
- 30 Band Octave Analyzer Checkbox – This changes the mode of operation to an IIR based simulation of a 1/3rd Octave frequency weighted system like an RTA).
- Smooth Spectrum – This smooths the graphical display based on a set of frequency domain calculations using adjacent sides of a bin within an FFT.

Note 1: The two IIR based Analyzers are useful for measuring relative noise levels against local Environmental laws, especially pure tones. For more information about pure tones, please refer to the Glossary of Terms section of this User's Manual.

Note 2: The 10 Band Analyzer breaks the audio spectrum down into the following center frequency buckets:

31 Hz, 62 Hz, 125 Hz, 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz

8 kHz, 16 kHz

Note 3: The 30 Band Analyzer breaks the audio spectrum down into the following center frequency buckets:

25 Hz, 31 Hz, 40 Hz, 50 Hz, 62 Hz, 80 Hz, 100 Hz, 125 Hz, 160 Hz, 200 Hz, 250 Hz, 320 Hz, 400 Hz, 500 Hz, 640 Hz, 800 Hz, 1 kHz, 1.3 kHz, 1.6 kHz, 2 kHz, 2.5 kHz, 3.1 kHz, 4 kHz, 5 kHz, 6.2 kHz, 8 kHz, 10 kHz, 13 kHz, 16 kHz, 20 kHz

Display Mode

- **Fast:** Shows the spectrum in almost real time with a rapid update rate.

- **Slow:** Shows the spectrum with slower / averaging (and leaky) ballistics.

- **Averaging (On/Off):** This allows the system to provide you with the average signal spectrum rather than in real time. It will integrate for as long as the file is being previewed, played, or loop-played. Its ability to respond to changes becomes slower as the integration interval lengthens. In other words, the longer that the file is played in this mode the slower its response will be and the more averaged will be its resultant display.

Amp Range (Amplitude Range)

This feature controls the Vertical Axis full scale range. You can select between the following:

- 150 dB
- 100 dB
- 50 dB
- 20 dB

Note: Lower values of dB produce greater degrees of amplitude resolution.

FFT Size

The FFT Size determines the basic frequency resolution of the system with the number of frequency bins equal to the FFT size / 2. Higher FFT sizes produce higher selectivity or frequency resolution with the tradeoff of longer integration time. You can choose between the following FFT sizes:

64, 128, 256, 512, 1024, 2048, 4096, 8192, 16384, 37768, 65536, 131072

Note: Frequency resolution in terms of absolute Hertz value at a given fft size will be degraded as file sample rate is elevated. For example, a file having a certain value of frequency resolution at 4096 ffts and 44.1 kHz will have a frequency resolution twice as wide with 88.2 kHz sampled files with all other factors being equal. In other words, bins will become twice as wide at 88.2 kHz compared to 44.1 kHz with all other things being set equally.

Start and Stop Frequency Values

The Start Frequency and Stop Frequency Data Entry fields allow you to establish the horizontal axis frequency range displayed on the Spectrum Analyzer. The Start Frequency sets the left-most graphical setting while the Stop Frequency sets the right-most setting. The frequency set-ability of this system is 1 Hz for each of the two parameters. The maximum range of adjustability for Start and Stop Frequency runs from 1 Hz to half the file sample rate*. So, by way of example, using a 44.1 kHz file, you can set a Start Frequency as low as 1 Hz and a Stop Frequency as high as 22,050 Hz (which covers more than the entire audio spectrum of 20 Hz to 20,000 Hz). On the other hand, you can set a frequency range as small as 1 Hz. For example, you can set a Start Frequency of 59 Hz and a Stop Frequency of 60 Hz. It is important to note that sample theorem comes into play and the highest frequency that you can actually plot with the Spectrum Analyzer shall be the .wav file sample rate / 2.

***Note:** As an example, a 44.1 kHz sampled rate file will allow for a maximum Stop Frequency of 22,050 Hz. Similarly a 96 kHz sampled file will allow for a maximum Stop Frequency of 48,000 Hz.

Frequency Axis Scroll Bar

At the bottom of the Spectrum Analyzer display, you will find a scroll bar. This allows you to move horizontally along the frequency axis in the spectrum display. Use your left mouse button coupled with a drag motion, either in the left or in the right direction with your Scroll Bar.

Window Type

You can choose between the following Window types:

Bessel, Blackman, Hanning, Hamming, Kaiser 10, Kaiser 20, Kaiser-Bessel, Rectangular, Triangular, Welsh

Descriptions of the characteristics of many of these Window types can be found in the section describing the Standard Precision (DCArt10) Spectrum Analyzer.

Mem (Memory) Button

This button activates the trace storage and trace recall features of the High Precision Spectrum Analyzer. You can store a large number of Spectrum Analyzer traces in the directory of your choice. The file extension is .spt for these data. You can only display one set of recalled data at any given time along with the present display data. This feature is very useful for comparing spectral data to a reference set of data and presenting it on the same graphical display. Clicking on it (Mem) brings up the following options:

- Save Trace: Browse to the Directory of your choice.
- Save Left Trace: Same as above, but applies to the Left Trace only.
- Save Right Trace: Same as above, but applies to the right Trace only.
- Load Trace: Browse to the directory containing the trace to be recalled and click on it and it will appear in the Spectrum Analyzer Display.
- Remove Trace: Eliminates a Memory location based displayed signal trace. Clicking on this button clears the

Spectrum Analyzer Display Window of the previously recalled spectral data.

Close Button

Clicking on this button closes down the Spectrum Analyzer Display Window.

Hold / Run Button

Clicking on this button alternately freezes (holds) the display or runs the Analyzer in a toggle manner of operation.

Point, Click and Measure

To measure the Amplitude and Frequency of any given spectral line or signal, just point your mouse at the signal of interest and left – mouse click. The two numeric values are then displayed in Yellow in the upper left hand corner of the Spectrum Analyzer. Its format is: Frequency in Hz, Level in dB (Hz, dB).

Settings for High Accuracy Measurement of %THD with the Precision Spectrum Analyzer

Display Mode: Fast
FFT Size: 65,536
Start Frequency: 20 Hz
Stop Frequency: 20,000 Hz
Frequency Resolution: 0.67 Hz
Resolution Mode: Check “Standard FFT” in the Options Menu
Range: 150 dB
Display Mode: Bar or Line, your choice as found in the Options Menu
Window Type: Kaiser-Bessel
Show Peak: Check this checkbox in the Options Menu
Show THD: Check this checkbox in the Options Menu

Using the Spectrum Analyzer in Real Time Mode

1. Bring up the Precision Spectrum Analyzer and set it up as desired.

2. Bring up the Multifilter and remove all of the filters from its lineup or click on the "Bypass" checkbox.
3. Click on "Live Preview"
4. The Spectrum Analyzer will then display the frequency domain content of your signal in real time as presented to the input of your soundcard after a relatively short latency time period.

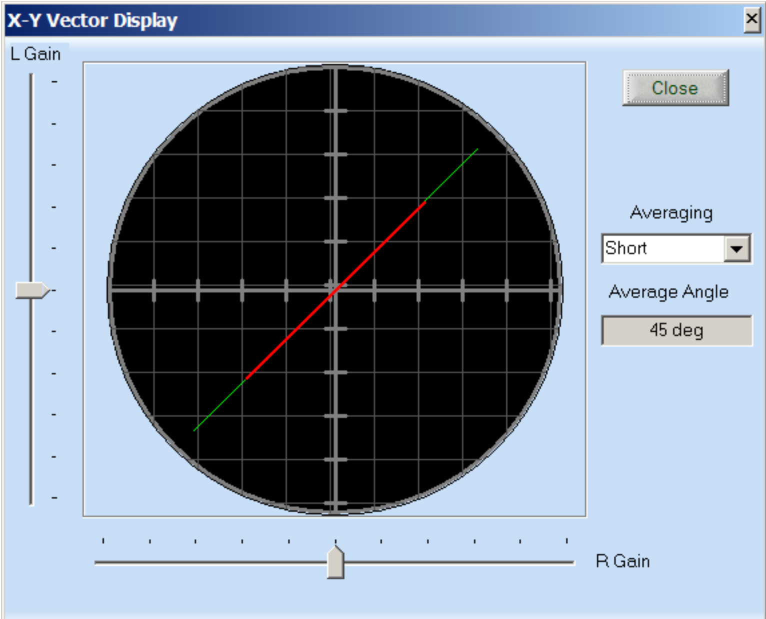
X-Y (Vector) Display

(XY Display)

The X-Y Vector Display plots the phase relationship between two waveforms. It is useful for identifying the nature of two-channel signals. Readily identified are monophonic vs stereo signals using this function. The instantaneous vector is displayed with a green trace while the averaged angle is displayed in red (provided that the averaging mode is enabled). **The Averaging feature is enabled by using the Averaging Selector and choosing between "None", "Short", or "Long". "None" turns off the averaging display leaving only the instantaneous trace active on the X-Y display. "Short" displays the moving averaged vector displacement angle between the two signals over a 4095 samples interval. "Long" displays the average vector displacement angle over the interval commencing at the time when the play button is started until the time when the stop button is depressed or over the interval of the time highlighted in the play source time display. The averaging interval for the "Long" mode is therefore a variable determined by the user depending on the length of the played file or the highlighted area of the waveform selected to be played and therefore measured. The "Averaging" mode is very useful when trying to measure the phase angle of signals having significant amounts of phase jitter. (Low cost tape recorders often display high levels of phase jitter especially at the upper frequencies in the audio band making the instantaneous graph difficult to use while making azimuth adjustments to the units playback head.) This feature is primarily used to align the azimuth of analog tape deck recording and playback heads. To do so, a known pre-recorded azimuth alignment tape is required containing a fixed frequency tone. The X-Y Vector display in conjunction with your azimuth tape and the Time Offset feature can be used to fine-tune a recorders time alignment without the necessity of having to take a screwdriver to your tape deck's head alignment adjustment screws*. The goal here is to correct the azimuth of the .wav file using the File Conversions Filter with the Time Offset slider control. Preview the .wav file with the X-Y Vector Display showing, and adjust the Time Offset slider control until a 45 degree positive sloped (up and to the right) straight line is seen on the display. That will be the optimum value of azimuth correction.

An important “cousin” view of the X-Y display is the “View Channel Phase vs Time” function found under the Forensics Menu (found in the Forensics version of the software only). This routine plots the average phase angle between the two channels vs time which can be useful in Forensics authentication applications. Please refer to the Forensics Menu section of this users guide for more details regarding this feature.

***Note:** The optimum method for adjusting analog tape head azimuth is via the X-Y display in conjunction with an analog azimuth reference tape and by directly adjusting the playback head azimuth screw on the tape player.



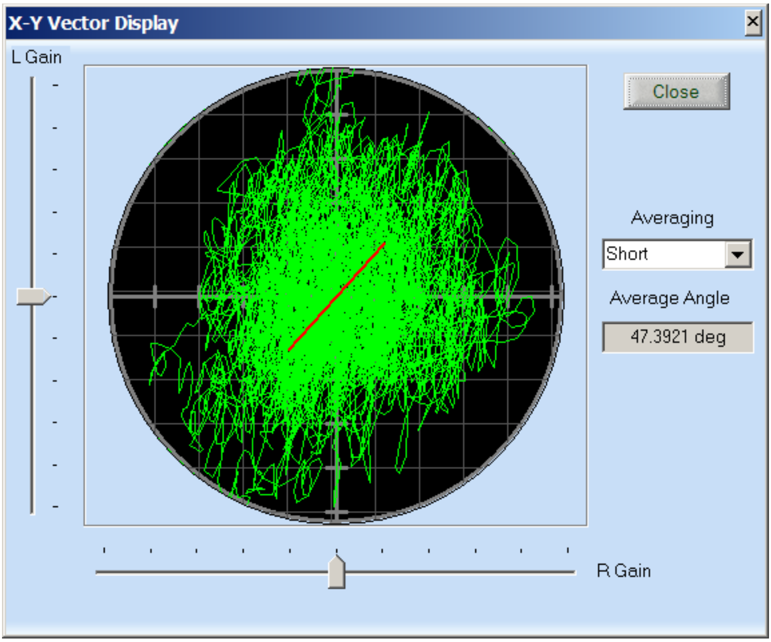
The X-Y Vector Display Showing Proper Azimuth Alignment

The following features will be found on the X-Y Vector display:

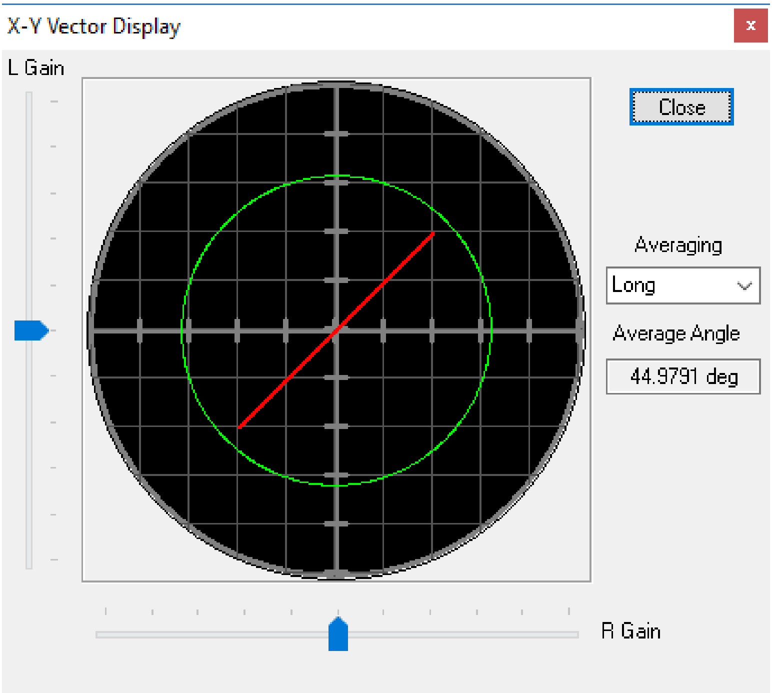
- X-Axis displacement (Horizontal), which corresponds to the Right Channel Input
- Y-Axis displacement (Vertical), which corresponds to the Left Channel Input
- X-Axis gain control slider (Horizontal & bottom in position)
- Y-Axis gain control slider (Vertical & left in position)
- Averaging Selector**
    - None: Averaging function Turned Off
    - Short: Averaging Interval = 4095 Samples
    - Long: Averaging Interval is Variable depending on length of "played" file
- Average Angle Display Window**: Displays the Average Phase Angle value with up to 6 digits of resolution.

Here is a listing of some common vector displacements, which can be observed on the X-Y Vector Display. They have the following meaning and are sometimes referred to as Lissajous figures :

- Straight line at 45 degrees with a positive (up and to the right) slope = Signals are in phase and are Monophonic.
- Straight line at 45 degrees with a negative (down and to the left) slope = Signals are out of phase and are Monophonic. An example of a situation that could cause this would be a miss-wired Stereo Phono cartridge or a miss-wired balanced input to the soundcard.
- Straight horizontal line only = Monophonic, Right channel only signal.
- Straight vertical line only = Monophonic, Left channel only.
- Straight line at 45 degrees with a negative (down and to the left) slope = Signals are 180 degrees out-of- phase.
- Circle = Signals are 90 degrees phase shifted.
- Elliptical tilted up and to the right = Signals are between 0 to 90 degrees phase shifted.
- Elliptical tilted up and to the left = Signals are between 90 and 180 degrees phase shifted.
- Frozen vertical figure "8" = Signals are frequency phased locked to one another but 2:1 in frequency ratio.*
- Moving vertical figure "8" = Signals are not frequency locked, but are about 2:1 in frequency ratio.*
- Random pattern of squiggly lines on the screen and average (red) signal up and to the right = Stereophonic audio signal in proper phase.
- Random pattern of squiggly lines on the screen and average (red) signal up and to the left = Stereophonic audio signal with improper phase.



The X-Y Vector Display Showing a Stereophonic Signal



The X-Y Vector Display Showing 90 degree phase shifted signals (green)

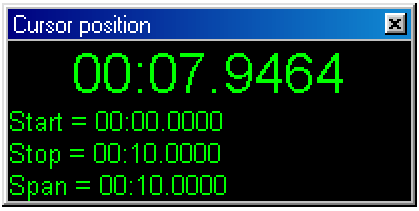
(45 degree average phase shift as shown in red)

***Note 1:** In these examples, the Right channel would be twice the frequency of the Left input. If the figure 8 were lying on its side, then the Left channel would be twice the frequency of the Right input.
****Note 2:** The Vector Angle Averaging feature is only available on the DC Forensics10 version of the software.

Time Display

The Time Display view displays all of the time related parameters associated with a Source or Destination workspace. When activated, a display window will appear containing four sets of timing numbers. This window can be dragged and placed anywhere in your workspace. The following time related parameters are displayed:

1. Cursor Location (largest numerals).
2. Start Location of a highlighted area (small numerals).
3. Stop Location of a highlighted area (small numerals).
4. Span - This represents the total time of a highlighted .wav file area (smaller numerals).

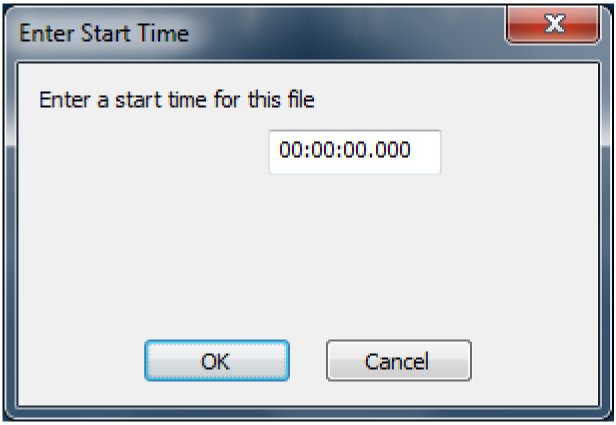


The Time Display Window

You can monitor "Samples" rather than "Time" in this display if desired. Go to the Preferences Menu and choose the "Display" option. Under that category, you will find a "Display Time Format" choice. Click on "Samples".

Set Start Time of File

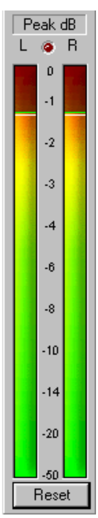
You can set the start time of your wavefile to any point in time that you desire. You simply bring up the file of interest and type in the new start time in the dialog box shown here:



Set Start Time of File Dialog Box

This feature is of value in circumstances where you need to sync your audio file up to other recorded events. An example would be aircraft black box analysis where you need to be in sync with ground communications recordings.

Output VU Meters



Two 100 segment VU meters can be displayed which will indicate the output of any filter or multiple filters that are being used by DC Forensics10 or DCArt10. These meters indicate the level of the left and right channels and have both average and peak reading ballistics. They are calibrated to indicate values from -60 dB to 0 dB, with 0 dB being full-scale output when operated in linear mode. The VU meters are calibrated to indicate -100 dB to 0 dB (with 0 dB also being full-scale output) when operated in log mode. The selection between linear and log modes is made via the "Edit/Preferences/General/VU Meter Scale" preferences tab. Any signal above the 0 dB level will be clipped by the system. Signals that are clipped are indicated by the illumination of a small red "LED" indicator at the top and in the middle of the meter display. The red clip indicator (at the top of the VU meter) trips on at signals above -1.0 dB (slightly lower than 0 dB).

Peak Hold
The VU meters includes two peak indicators. The white horizontal bar is designed to “hold” the indication of an overload for two seconds after it occurs so that it will be more obvious to you that a clipping event has occurred. The green horizontal bar is a peak hold indicator and will hold the value of the audio peak until such time as the reset button, located at the bottom of the VU meters, is activated by the mouse. These VU meters can be activated or de-activated under the View Menu. Also, they can be "dragged and dropped" anywhere on your desktop workspace using your left mouse button or docked and enlarged at the right or left hand margin of the software user interface. The right mouse button can also be used to “hide” or “auto hide” the VU meters.

Volume Control

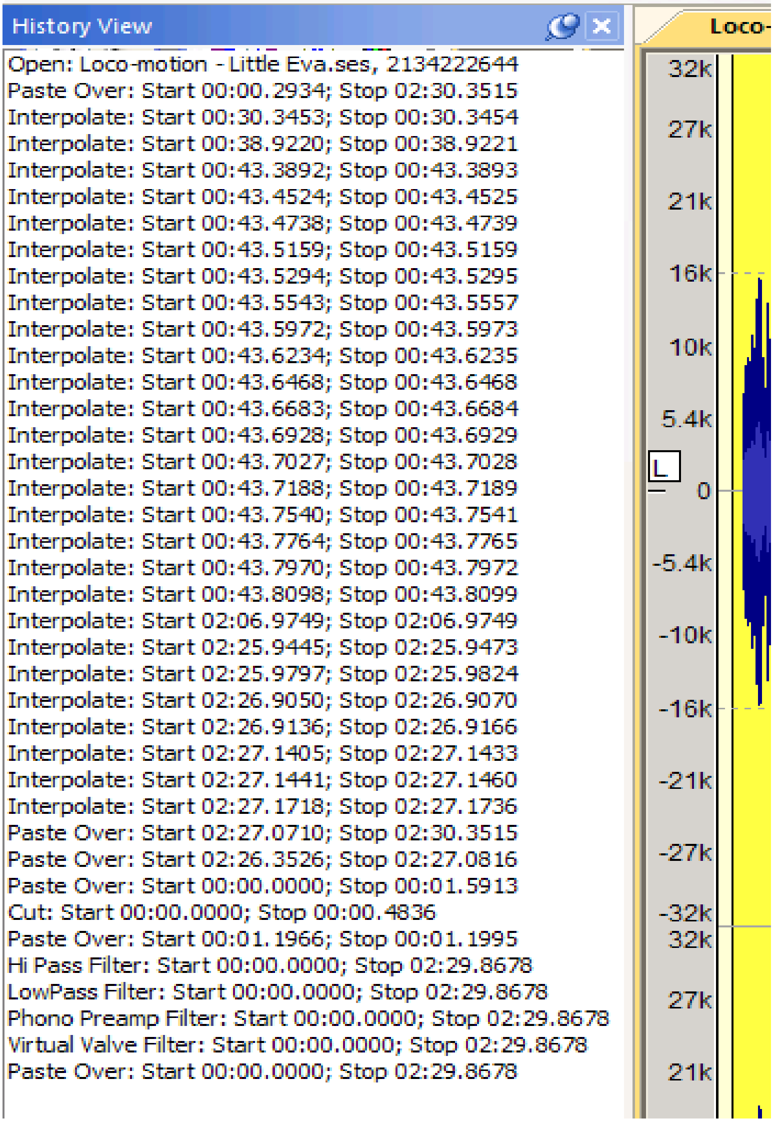
This feature allows you to view and adjust the primary sound card volume controls. Three controls are presented:

1. Main Volume Control
2. Wave Volume Control
3. Balance (between left and right channels)

Important Note:
This volume control is only useful for sound cards that make use of the Windows Mixer. For any sound card that doesn’t use the Windows Mixer, adjustment of volume will take place using the proprietary mixer provided by the manufacturer of that product.

Fast Edit History

When working in Fast Edit mode, this cool little panel on the left side of your screen is like having your own personal secretary. It keeps track of every edit and allows you to navigate quickly from one change to the other. If you decide to start at midpoint in this list and want to eliminate all edits done after this point, just double click on that edit and the system will ask if you’re sure…then eliminate all edits after that point. All of the data pertaining to the fast edit history profile are contained in .ses (session) files. These files are stored with your .pkfs (peak files) and can be deleted once you've finished with the file. Fast Edit temp files include elements of the source file name for ease of identification.

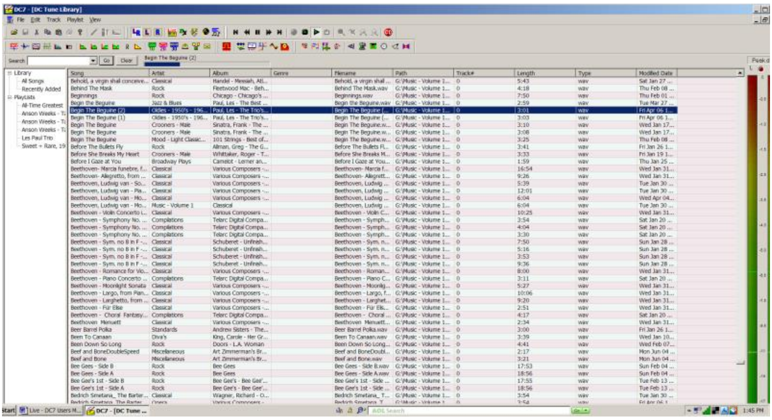


A Typical Fast-Edit History Panel View

DC Tune Library

The DC Tune Library provides you with a full-featured audio file archive capability within the context of your DCArt10 or Forensics10 software. It will search your hard drive and find all audio files and store their path, name, genre and other pertinent data within its file structure. After this database is constructed, it provides lightning speed access to any of your audio files, whether they are in .wav, mp3, or wma format(s). You can search your database, order it in terms of various parameters and create and recall playlists. You can play any file directly through the normal play features of your Diamond Cut software, or you can listen to a file or series of files by using the Preview feature found on the Diamond Cut filters and/or effects. You can even listen to the database by way of a complex series of filters and effects by previewing a file via the DCArt10 or Forensics10 Multifilter.

Details pertaining to the various functions associated with the DC Tune Library can be found in an earlier section of this users guide dedicated to this topic.



The DC Tune Library Spreadsheet Display

Zoom In

The Zoom In feature allows you to magnify any portion of your .wav file, no matter how small it may be. It can be used with either the Source, Destination or Fast Edit workspace. The zoom-in process may be repeated any number of times for a really close and detailed look of your audio waveforms. However, only the last 5 zoom levels are remembered. You can also access this feature by using the **Z** hotkey or the Right Click Menu. Your mouse wheel will also control the Zooming In (or Zooming Out) feature.

Zoom-In X2

Allows you to Zoom in on a waveform by a factor of 2 (binary zooming). The amount of time displayed is decreased by a factor of 2 each time the Zoom-In button is pressed. This zoom differs from the other Zoom function, which zooms in on the selected area. Pressing the + key on the numeric keyboard will also activate this zoom function. It can be used with either the Source or Destination or Fast Edit workspace. The “Zoom In X2” process may be repeated any number of times for a really close and detailed look at your audio waveforms. It’s complimentary function is “Zoom-Out X2”.

Zooming-In/Out on a portion of a .wav file (Tutorial)

To Zoom-In and Zoom-Out on a portion of a .wav file, use the following procedure:

1. With your mouse, left click on the workspace location that you desire to magnify.
2. As you begin to drag the mouse pointer towards the right-hand portion of the screen, you will see two black timing reference markers appear in the workspace.
3. The first marker will remain at the location at which you began to perform the "mouse-drag" operation, indicating the location of the "start" position of your zoom region of the .wav file.
4. When you get to the end of the interval of the .wav file portion you wish to Zoom-In on, release the left mouse button, and the second

line will remain at that location. This line will have indicated the location of the "end" position of your selected zoom-in region of the .wav file.

5. Click the mouse on the Zoom-In icon
6. The selected area will be re-displayed with the time axis expanded to fill the monitor screen.
7. To zoom back out, merely click with the left mouse button on the Zoom-Out button on the toolbar.
8. After you have clicked on the Zoom-In icon, you will notice that the Time Axis Slider control at the bottom of the file will move to the relative position of the .wav file where you began your zoom operation. By using either the slider directly with your mouse, or by using the arrows at each end of the slider, you will be able to move through the original relative position of the entire .wav file, displaying the same level of time magnification that you had just established through the use of your Zoom-In control.

The Zoom In X2 and Zoom Out X2 buttons function in a different manner and give you the option of zooming in or out without changing the selected area of the .wav file. Pressing Zoom-In X2 will take whatever area of the waveform that is currently displayed and magnify it by a factor of 2. Similarly, the Zoom-Out X2 button will take the displayed portion of the .wav file and decrease the time resolution by a factor of 2. Both of these zooming operations can be performed multiple times if a higher degree of focus or de-focus is desired.

Zoom Out

This key performs the inverse function of the Zoom-In key. It allows you to progressively back out of a .wav file which you had previously Zoomed-In on. Hitting this key allows you to *backpedal* through your last 5 zooms. You can also access this feature by using the **X** hotkey or by using the Right Click Menu. Your mouse wheel will also control the zooming In - Out feature. For a brief tutorial, see above.

Zoom-Out X2

This feature allows you to back out of the displayed area by a factor of 2. The amount of time displayed is increased by a factor of 2 each time the Zoom Out button is pressed. Pressing the – key on the numeric keyboard will also activate the Zoom function. It can be used with the Source, Destination or Fast Edit workspace. It's compliment is "Zoom-In X2.

Zoom Out Full

This feature allows you to quickly zoom back to square one. It will display your entire .wav file waveform on the screen. You can also access this feature by using the **Ctrl + Z** hotkey or the Right Click Mouse Menu.

Zoom to Markers

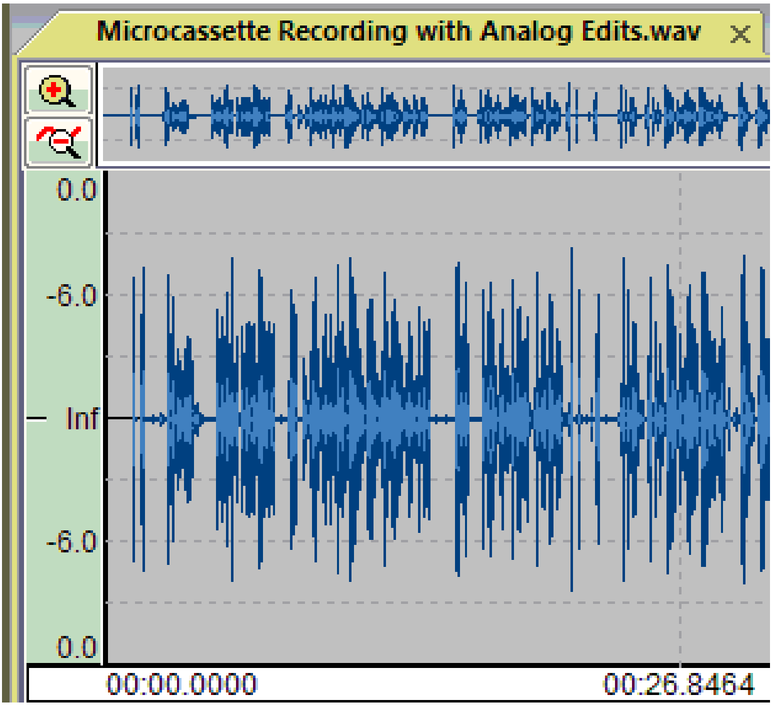
This feature is only activated when markers are dropped. When used, it fills the screen with the material located between two markers.

Box Zooming

To zoom into an area of a waveform without time or amplitude limits, you can use Box Zooming. This is accomplished by holding down the Shift key and then holding down the left mouse button to draw a rectangle around the area you want to zoom in on. When you release the mouse button, the system will zoom into the circumscribed area of the file. To zoom back out, just use one of the zoom out commands.

Waveform Overview Zooming Feature

You can also zoom in and zoom out on a time domain waveform via the waveform overview feature. The feature is located in the upper left corner of the waveform overview display. It includes two zooming buttons, one having a plus sign and the other having a minus sign as shown here:



Waveform Overview Zooming feature

Each time that you click on the plus sign (on left side of the waveform overview display) the system zooms in closer on the main time display, with its relative position shown with respect to the overall file in the overview display. Zooming out is similar, but it requires you to click on the minus zoom button.

Sync Files

This feature, when enabled, mutually synchronizes the Filter and other commands such as Zoom-In and Zoom-Out between the Source and the Destination .wav files. This feature is useful for selective filtering or viewing of a portion of a .wav file. For example, if you run a particular filter, and then find a sector (or sectors) that need further filtering, you can highlight only the portion that needs the additional processing, and apply the filter accordingly. Sync mode will also accommodate different filters being applied to different sectors. Also, when enabled, highlighting and Zooming-In on a particular area of a source file will correspondingly Zoom-In on the same timing co-ordinates in the Destination file. The converse is also true such that Zooming-In on particular set of co-ordinates in a Destination file will also cause the Source file to contain the same Zoom-In co-ordinates. This feature is of particular use when you want to visualize how a particular filter may have modified a waveform from your Source file. Clicking on it with your mouse enables this feature via the Sync Files icon or via the View Menu. When it is active, a check mark will appear to the left of it in the View Menu.

Note: The Sync Files function is only useful in Classic Editing Mode.

To use Sync Mode to apply selective filtering to a .wav file, it is first necessary to create a Destination file from your Source file. This can be done through the normal processing procedure of any of the filters, or it can be accomplished with one of the file conversion options such as Mono-to-Mono or Stereo-to-Stereo. Thereafter, merely highlight the portion of the Source file that needs filtering, click on the appropriate filter, select the appropriate filter values and run the filter. Only the highlighted sector of the two files will be enacted upon by the filter, with the results appearing in the Destination file. If another filter (or the same filter that you had just been working with) is then required to be applied to another portion of the .wav file, just repeat the outlined process.

Non-Sync mode of operation

In non-sync mode, the highlighted section of the source file is read and processed by the filter. The processed section is then written to the destination file starting at the beginning of the file. If a destination file already exists, it will be overwritten (a prompt warns you of this). This mode is useful when only a section of the source file needs to be extracted, or for testing a filter's settings before processing an entire file.

Selective Filtering with Sync Mode (Tutorial)

1. Open the desired Source .wav file.
2. From the Source .wav file, create a Destination .wav file utilizing the Clone Source (File Menu) or one of the File Conversion alternatives or one of the DCArt10/DC Forensics10 Filters.
3. Activate Sync Mode by clicking on it in the View Menu. A check mark will show that it’s active.
4. Highlight the portion of your Source .wav file that needs Selective Filtering.
5. Left click on the "Filter” or “Effect” menu.
6. Left click on the filter or effect that you desire to run on the selected portion of your Source .wav file.
7. Adjust the filter/effect parameters accordingly utilizing "Preview” mode if necessary to get the parameters to sound appropriate to your taste.
8. Click on "Run Filter".
9. The highlighted portion of the Source .wav file will be processed and inserted into the same time-slot in the Destination Workspace. The rest of the Destination file will remain as it had been.
10. If further selective filtering is required on a different portion of the Source .wav file, repeat steps 4 through 9 until you are satisfied.

Display Themes (Application Looks under View Menu)

Application Look - - - have it your way; this feature offers different skins and themes for your Diamond Cut Windows, Filters, Effects and Dialog Boxes.

You can choose between the following depending on your taste:

Windows 2000, Office XP, Windows XP, Office 2003, Visual Studio 2005, Visual Studio 2008, Office 2007, and Windows 2007.

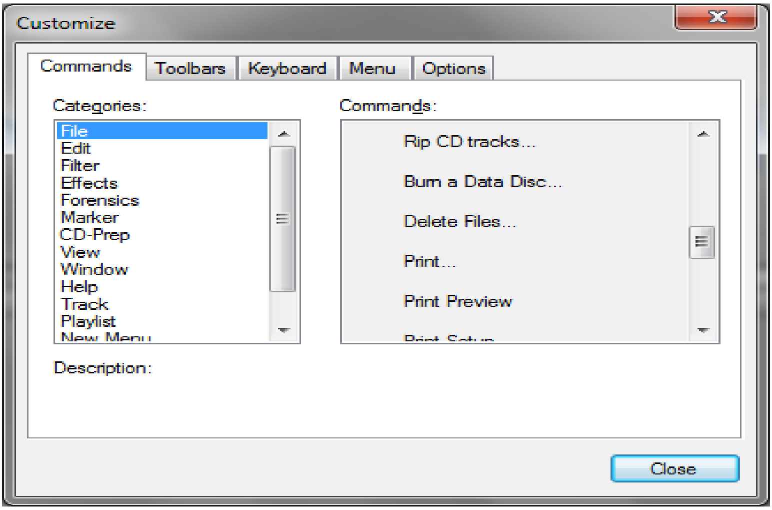
The "Application Look" Dialog Box

You can choose between a wide variety of skins, themes, colors and fonts. Experiment with this until you obtain your favorite look, or if you get bored with a given look, change it on occasion for variety.
If you prefer the classic old way that the software looked with Windows XP, you can get back to that appearance by choosing the Windows XP selection. Also, please note that not all skins and themes support variable color schemes or font sizes. The "Display" preference for showing the dialog color schemes (found under the Edit\Preferences\Display tab) is by default left on when you enable a theme. Users may want to turn these off to see the full effect of the display theme that they have chosen.

Enabling or Disabling Toolbars in DCArt10/DC Forensics10

The following toolbars may be turned on or off by the user by simply checking or un-checking them from the View menu item. The Diamond Cut Toolbars are both movable and dock-able.

Customizing Your Toolbar(s)



One of the Customize Dialog Boxes

You can custom configure your toolbars using the following technique:

1. Click the Toolbars Tab.
2. Click Customize.
4. Click Toolbars.
5. Click New.
6. Add name of new Toolbar.
7. Click “ok”.
8. Click on Commands Tab.
9. Select Tools from menu options.
10. Drag and Drop the Command from the Right window onto the Toolbar.
11. Close customize when you have your toolbar set up as you want it.
12. Reposition or Dock the toolbar depending on your needs.

Toolbars and Customize Windows will bring up this Dialog Box

Channel Toolbar

Just beneath the filter and effects toolbar and towards the left of the display can be found the Channel Selector Toolbar. This allows you to choose which channels are to be operated on by the system. Your choices are as follows:

1. L/R (Both channels)
2. L (Left channel only)
3. R (Right channel only)

The Channel Selector applies to all Filters, Effects and Editing functions accept for tools that change the overall length of the file. Generally, one would tend to leave this feature in the L/R (Both channel) position unless dealing with special editing or sound restoration issues.
Note: You can click on any of the Channel controls while playing to select either the Left, Right or Both channels. When previewing a filter, both channels will always be previewed, but only the channel selected before the preview starts will be affected by the filter.

File Toolbar

The main DCArt10/DC Forensics10 File Toolbar resides along the top of the program window. It contains the most commonly used file functions of the program. Clicking on them with the left mouse button activates them.

Filter and Effects Toolbar

The Filter and Effects Toolbar also resides near the top of the program window. It contains buttons for the most commonly used filters and effects. This toolbar, like the others, "floats" and can be dragged and dropped anywhere within the DCArt10/DC Forensics10 window using the mouse. Clicking on them with the left mouse button activates them.

Forensics Toolbar

The Forensics Toolbar contains buttons for the most commonly used Forensics filters and tools.

Play Controls Toolbar

This toolbar contains the "Transport" controls for recording and playback of the audio.

Status Bar

The Status Bar resides along the bottom of the program window. Six Diamond Cut Software related parameters are displayed therein:

- **Program Mode** is displayed in the lower left-hand field. Initially, it will indicate "Ready". Clicking on the various toolbar buttons (from left to right) will enable the following functions and indicate the following operation on the status bar:
    - A. Open an existing Waveform audio file
    - B. Save the active Document
    - C. Delete a Section of a .wav file
    - D. Copy selection and put it on the clipboard
    - E. Paste Clipboard contents over the selected area
    - F. Context Sensitive Help
    - G. Rewind file position to beginning
    - H. Pause Playback of the selected file
    - I. Fast Forward to the end of the file
    - J. Start Recording an Audio file
    - K. Stop Playback of the Current file
    - L. Play the selected File
    - M. Zoom-In to the selected area of the sound file
    - N. Zoom-Out to the last selected sound file

Note: Clicking on the various menu items will also activate a one-line help text file, which describes each item.

- **Channels Used**: The field to the right of the "Program Mode" field shows the number of channels being used and will indicate either mono or stereo.
- **Sample Rate:** The next field moving towards the right shows the sample rate in kHz.
- **Bit Rate:** The next field moving towards the right shows the bits per sample of the current file.

- **Highlighted Length:** The next field tells you the length of time for a highlighted portion of the Source or Destination workspace.
- **Available Hard Drive Space:** The next field indicates the Available Hard Drive Space in Mbytes. This is useful to determine whether or not there is enough storage space to perform your processing.

File Info

The following information will be displayed regarding the current highlighted .wav file when "File Information" is clicked:

1. File Name
2. File Type
3. File Path
4. File Size (Bytes)
5. Length (Time)
6. Channels (Mono/Stereo, etc.)
7. Sample Rate (kHz)
8. Bits Per Sample
9. Last Modified (Date)

Rebuild Peak File

Peak files (.pkf) provide a graphical representation of the .wav files and its data are stored along with any .wav file created or operated upon by DCArt10/DC Forensics10. It takes 1 sample per 200 .wav file samples and extracts the peak value in that time interval for time display purposes. After many editing processes, it may be desirable to update this file to more accurately represent the present state of the displayed .wav file in the Source or Destination workspace(s). To do this, click on this function, and you will see that the displayed .wav file will become updated after a short period of time, accurately representing your edits. Peak files may be deleted by the user at any time.

The Window Menu

The Window Menu contains all of the commands that affect the manner in which the files are displayed to the user. The following commands are available:

Cascade

The "Cascade" command arranges all of the open Windows so that they appear on the display one behind the other. The most recently opened Window will appear on top, with the remainder arranged in the order in which they were opened.

Tile

The "Tile" command arranges all of the open Windows so that they appear on the display in a matrix configuration (in the same plane) on your display. This feature is used when you need to see all of the workspaces concurrently.

Arrange Icons

The "Arrange Icons" command will arrange all of the "iconized" windows in a row near the bottom of the workspace.

Close All

As the name implies…click here to close all Windows and Files via one command.

Open Files

A listing of all of the open files is provided at the bottom of the Window Menu. The file that is active will have a check mark indicated to the left of the file name.

The Help Menu

With this product, we strive to provide you with the best possible documentation, help and support. To that end, in addition to this in-depth User's Manual, we also include a computerized Help File that is always available at the click of a mouse. Integrated into the Help File are a Table of Contents, a comprehensive Index, a Search Engine and a Favorites selector (Add, Display or Remove items). If you find a topic that you consider to be of great importance, you can print the displayed topic via the Help file print feature.

Tip of the Day

This feature is normally displayed on the splash screen after you launch the software. It can also be invoked under this menu item. It provides useful hints on the operation of the software contained within roughly 330 total tips. It advances through the list one item at a time each time the software is opened. When it gets to the last tip, it returns to the first item in the listing and starts the sequence over again. The tip file is common to two Diamond Cut Software programs, so some tips may not be relevant to your application(s). If you are not interested in using the tip of the day feature, uncheck the box that indicates "Show Tips on Start-Up" found in it's dialog box.

Restoring a Recording

This feature links you to a section of the Help File that describes step-by-step details regarding the restoration of an old recording.

Restoring the demo1.wav

This feature links you to a section of the Help File that describes in step-by-step detail the process for restoring the demo1.wav file that is included with the software package. This is an excellent exercise to go through to get an initial idea of the operation of the system.

Contents

The DCArt10 & DC Forensics10 program contains an extensive Help file. These topics are printable from within the help file for your convenience. The following types of information are available:

- DCArt10/DC Forensics10 Program Overview
- Tutorial Information on each filter
- Step-by-Step procedures for utilizing each filter and editing feature
- Audio Restoration process procedures
- Practical Hints on recording procedures
- Extensive Glossary with definitions, tables of values, and Appendix information.
- Licensing Agreement, Warranty, and Software Program History information
- FAQ's
- Acknowledgements

Context Sensitive Help

This button will provide you with on-line context sensitive Help from the DCArt10/DC Forensics10 Help file.

Using Context Sensitive Help (Tutorial)

Method #1

1. Point the mouse pointer to the portion of the DCArt10/DC Forensics10 program display about which you would like some Help file information.
2. Depress the special function key F1.
3. Help file information will appear in a window. This information is usually only summary procedural information on the particular topic. For more detailed information, refer to the actual Help file, and look for tutorial information on the topic of interest.

4. When done, click on the - button at the top left-hand side of the Help window.

Method #2

1. Using the left mouse button, click on the button with the "?" contained within its perimeter on the toolbar.
2. A black, mouse controlled "?" will appear next to the mouse pointer.
3. Move the pointer to the area of the program display about which you would like some Help file information.
4. Click on the left mouse button.
5. Help file information will appear in a window. This information is usually only summary procedural information on the particular topic. For more detailed information, refer to the actual Help file, and look for tutorial information on the topic of interest.
6. When done, click on the - button located at the top left-hand side of the Help window.

Note:
Under some circumstances, Method #2 will not activate the context sensitive Help file. When this occurs, use method #1.

The Help file search engine includes some special search filter checkboxes located in the lower corner of the search box including:

1. Search Previous Results
2. Match Similar Words
3. Search Titles Only

Important Note:
Features which have not been implemented in your particular version of DCArt10/DC Forensics10 but which may be provided in different releases of the program will be described in italics.

Online Knowledge Base

Clicking on this will link you to a body of knowledge and information that you may find helpful with your audio work. It includes numerous application notes that are always being added to over time. It is worthwhile to check out this knowledge base now and then to stay up to date.

User's Manual .pdf

Clicking on this item will take you to the documentation package in .pdf (User's Guide) for the software products so that you can download it on various computers if desired.

Web Homepage

Clicking on "Web Homepage" will link you up to the Diamond Cut Productions, Inc. website where you can view our latest product offerings.

Check for Updates

This like will compare the version of software that you have against the latest release and let you know if you are up-to-date.

About DC-Art

The following information is obtainable under the "About DC-Art" feature:

- Program Name and Version #
- Copyright Information including Copyright Dates
- Upgrade Contact Information
- Program Registration Information
- System RAM Size Free
- Hard Drive Disk Space Available for use on the temp drive
- Program Registration Name (email address of user)
- Serial Number (25 digit alpha-numeric code)
- Registration Code (12 digit hex code)
- Manufacturer Contact Information
- Check for Updates (this will ping the Diamond Cut Server and let you know if your installed version is the latest)

Tutorials

Do you need help with a specific problem? This could be the place for you. Here, we provide you with procedures that may require more than one tool or one step to accomplish.

Where are the Demo .wav Files?

You will find a number of demo .wav files in the following directories or accessed directly under the File menu: The path to navigate to these files is as follows:

For DCArt10 version 10.x

Using the Windows XP operating system:

<user>\MyDocuments\ DCArt10\Wavefiles

Using the Vista or Win7, Win8 or Win10 operating systems:

<user>\Documents\DCArt10\Wavefiles

For DC Forensics version 10.x

Using the Windows XP Operating system:

<user>\MyDocuments\ DCForensics10\Wavefiles

Using the Vista or Win7, Win8 or Win10 Operating systems:

<user>\Documents\DCForensics10\Wavefiles

Note: For the Win 7 Operating System, the "Documents" folder appears in the "Libraries" folder.

Shortcuts to the demo .wav files can be found in the "Diamond Cut Audio" program group. From the "Start" menu, choose "All Programs", then "Diamond Cut Audio" then "Demo Wave Files" and select the desired demo file shortcut.

These files are intended to help you understand through experimentation the use of some of the features found in your software. Here is a listing and description of the demo .wav files and some of their possible tutorial uses:

What Are the Demo .wav Files?

(Test Audio Files)

Getting started demo1.wav This file is a raw transfer (with proper turnover EQ applied) of an Edison Electrical Lateral Cut Disc (thin-cut) test pressing recorded circa the late 1920s. It is just a snippet of a song called "My Sin" and can be found in its entire form and completely cleaned up on the Diamond Cut CD entitled "The California Ramblers - - - Edison Laterals 2". This demo .wav file contains a large number of clicks and ticks (and some hiss too) and thus is primarily designed to be used to demonstrate the operation of the various impulse filters in your software. In particular, you will find a preset titled "***demo1.wav de-clicker***" within the EZ Impulse filter. This preset will de-click that file, but you can also use this demo file to experiment with the various parameters within that filter. It can also be used in conjunction with the EZ Clean and the Expert Impulse filters for experimentation purposes too.

RadioDemo.wav This file is a recording made on the East Coast of the USA of a German Broadcast as received on a short wave radio receiver at around noon time, EST. It contains a lot of hiss and the audio level varies a bit throughout. This .wav file is useful to show the multiple filtering/effect capability of the Diamond Cut Multifilter. To hear what it can do, just bring up the Multifilter and then bring up the preset called "***Short Wave Radio Anti-fade & Cleanup***". You will see a lineup consisting of 5 filters and effects connected in cascade. Use the

Preview button in the Multifilter to hear the result and use the bypass button to hear the unfiltered sound. Experimenting with the various filters and parameter settings contained within this preset in conjunction with this .wav file should prove instructive.

BigClickCracked78Demo.wav This .wav file is taken from an electrically recorded 78 RPM record which has a crack running from the outside rim all the way through to the center hole. It is a snippet of "The Blue Danube" and was recorded in the early 1950s. This demo file is intended to instruct you in the operation of the "Big Click" filter. Note that ratio settings of around 1.4 to 1.7 are effective and that the "thump" produced once per revolution of the record can be squelched by turning on the "De-Thump" feature. After the Big Clicks are repaired, use the EZ Impulse Filter to remove the remaining smaller clicks and ticks. The Preset in the Big Click Filter to repair this demo file is called "***Big Click Cracked 78 Demo Repair***". There is a more thorough filter called "***Big Click Cracked 78 Demo Repair (Comprehensive)***" which not only eliminates the big clicks, but also the smaller crackle heard on this badly damaged record.

Aircheck 1933 Demo.wav This .wav file was recorded in 1933 on an RCA pre-grooved radio/phono/recorder combination located in Waukesha Wisconsin, 370 miles from the 'clear-air' 50 kW AM radio station KMOX from which it was live-broadcast from St. Louis, Missouri. It comes from a one of a kind record and is the first recording including a popular musician/inventor. We'll let you guess who it is. There is a Multifilter preset that is all set up to clean this crude recording called "***Aircheck Radio Recording 1933***". This is a good file to use to help you to become familiar with some of the tools found in the Diamond Cut Productions software, since the record suffers from a large number of audio defects.

Wind Noise Demo.mp3 This file is taken from a field recording of a jam session and contains wind noise mixed with musical content. Use this file to help you learn how to use the Wind Noise Filter. A preset has been set up for you to experiment with called "***Wind Noise Filter Demo***". Although this file is musical in nature, the filter is actually optimized for spoken word recordings in very windy environments. The other presets found indicate the level of wind that this filter can be used on.

HissBurstDemo.wav This .wav file has a hiss burst roughly 8 seconds into it. The hiss burst was likely caused by the momentary loss of the RF carrier during an FM broadcast. This hiss burst is used to demonstrate the various functions associated with the Bi-Modal Interpolation (the "I" key) feature. You can highlight the hiss burst and experiment click on the I key, and it will be interpolated out of the piece. It's as simple as that.

Vertical Cut Demo.wav This is a Pathe' vertical cut acoustically mastered disc which permits you to experiment with the Virtual Phono Preamp (VPP) and it mono – vertical cut feature. The recording is called "Skies of Normandy". The VPP will resolve this very noisy recording's vertical component of displacement and only leave behind the true signal which is in the vertical undulation direction and rejecting most lateral direction influences. A preset is provided called "***Vertical Cut Demo Preset***" as a demonstration of what this filter can do to the vertical cut demo file.

Dyna-Bass Demo This is a song titled "True Blue Lou" performed by Catherine Annette Hanshaw, circa 1929 on the Diva record label. It is a lateral cut 78 having considerable amounts of rumble, but some deep bass buried with the musical performance. Use the "***Dyna-Bass Demo"*** preset as a starting point to enhance the bass on this file while rejecting much of the rumble. Note that there are two other demo presets that you can experiment with in the Dyna-Bass preset listing.

dssdemo.wav This .wav file is a simulation of a loud bar room situation in which a male person is discussing a situation on the phone with loud rock music playing on the jukebox in the background. One side of this phone conversation is picked up by another nearby party in the bar via a "wire" that he or she is wearing. The recording is binaural in nature with one microphone located close to the jukebox and the second microphone hidden on a person and located in proximity to the person who is on the telephone. It is intended to demonstrate the usefulness of the DSS (Dynamic Spectral Subtraction) Forensics filter in terms of rejecting loud background music (or other types of coherent noises) from a surveillance recording. To hear this demo in action, bring up this file and then the DSS filter found in the Forensics menu. Then, bring up the DSS preset titled "***dssdemo***" or the preset titled "***dssdemo-with gain***". To compare the results, switch the filter between normal "Preview" operation and "Bypass" mode. You may want to increase the "Output" control on the right side of the DSS to hear the conversation more clearly. An alternative preset to explore with the DSS filter is found in the Multifilter and is called "***DSS Demo – Enhanced Performance***".

Note: The dssdemo.wav is only available in the DC Forensics version of the software.

Male Voice ID Test.wav This file uses a Broadcast Wave (BW) format for testing that software feature and can also be used to experiment with the spectrogram.

ForensicDemo.wav This file is a snippet of an air-check recording of the famous inaugural speech made by F.D.R. in 1933 in which he stated that "We have nothing to fear but fear itself". This recording was captured in real – time via an AM radio on a Victor Pre-Grooved Record recorder at that time. It was recorded from a radio transmission out of the New York metropolitan area via radio station WEAF and was received in Newark NJ via a combo Victor AM radio / disc record recorder machine. This file includes a combination of random and coherent noise signals (hiss and heterodyning) which are damaging to its intelligibility. Much of the noise that you hear is native to the original air-check recording. Other noises have been added in to demonstrate the de-noising capability of 8 filters/effects cascaded in the Multifilter. To evaluate this functionality after bringing up this file, go to the Multifilter and click on the preset titled "***Forensic Demo Clean-up Filter***", and then Preview it. You will note that the voice is barely intelligible until you enable the preset. You can easily switch back and forth between the raw file and the enhanced version by using the "Bypass" checkbox in the Multifilter.

Note: This demo .wav file is only available in the DC Forensics version of the software.

Composite Bit rate mp3.wav This file is useful when experimenting with the Histogram vs. Time and the Comparative Histogram features. Suspicious edit points can be seen with this file.

Whisper Enhancer Demo.wav This file shows the power of the Whisper Enhancer feature found in the Forensics version of this software. The two demo presets are titled "***Whisper Enhancer Demo Preset***" and "***Whisper Enhancer Demo Preset 2***". Their parameters are set differently so that you can observe those effects. Experiment around with the slider controls since you can always get back to the original factory demo settings. This demo file and filter/preset is only available in the Forensics version of the Diamond Cut Productions, Inc. software.

Voice ID / Voiceprint Demo Files: *(Forensics versions only)* Six sample .wav files are provided with the software to help you learn more about the voiceprint patterns of a female's, male's and child's voice as recorded through high and low quality signal paths. These files consist of three persons saying the same "cue word" sentence. The low quality versions were recorded simultaneously with the high quality versions for ease of comparison using the Diamond Cut Spectrogram and Voice ID feature. Here are the files that are included:

Female Voice ID Test Sentence - High Quality.wav
Male Voice ID Test Sentence - High Quality.wav
Female Child (12) Voice ID - High Quality.wav
Female Voice ID Test Sentence - Low Quality.wav
Male Voice ID Test Sentence - Low Quality.wav
Female Child (12) - Low Quality.wav

The "Low Quality" files were recorded using a very low cost dynamic microphone manufactured circa 1970 and a commercial grade mixing board manufactured in the mid-1970s. The "High Quality" files were recorded with a studio grade large diaphragm condenser microphone in conjunction with a "dedicated" studio grade microphone pre-amplifier. All 4 of these files were digitized via a high-end soundcard. All of the files are sampled at 44.1 kHz with 16 bit depths.

Note 1: These demo .wav files are only available in the DC Forensics Audio Laboratory version of the software.
Note 2: The persons whose voices you hear on these demos were born, raised and live in Northern NJ, USA, in case you are trying to identify their accents.
Note 3: The Female Child's voice is that of a 12 year old.

Forensics Files with Edits:

Microcassette Recording with Analog Edits.wav.
This file has analog edits made on a microcassette machine which can be detected using the Subsonic Explorer feature. The Subsonic Explorer is found via this path:

Forensics Menu/Authentication/Subsonic Explorer. The edit points will be seen in the destination display as impulse “spikes” where the edits were made. Just look above the Destination display at the Source file since the impulse spikes point to the edit points. There are no filter presets associated with this file.

Mary Had a Histogram (8 bit edit on 16 bit file).wav
Use this file to experiment with the Forensics Comparative Histogram and/or the Histogram (vs. time) feature. You will discover the edit point.

60 Hz Phase Shift with Phase 4 Discontinuity Final.mp3
This file will help you to experiment with the View Phase vs Time Forensics Feature found under the Forensics Menu. It includes varying phase as well as several edits. The phase discontinuities are graphically shown and clearly identified.

Analog tape recording Transfer Tips

Analog tape recording was developed in Germany sometime in the early 1940's. This technology was brought to the United States sometime after the end of WW-II. Ultimately, tape recording replaced other methods of audio recording, such as direct to disc mastering, acetate and wire recording. Early recorders held the recording media (ferrous oxide) on paper tape backing. The grainy texture of paper produced discontinuities with the contact with the recording and playback heads, producing numerous analog artifacts. Later, more sophisticated backings were employed to get around that problem, but new problems were introduced that only showed up in subsequent years. Some of the problems that were introduced are of concern to the audio restoration engineer or archivist, since the effects of age on the tape backing may have degraded the oxide material. Care in handling and pre-processing the tape should be considered when transferring old tape recordings to your hard drive for restoration. The following pre-processing and handling precautions should be taken depending on the tape medium.

- **Paper Backed Tape:** This type of tape was recorded monophonically. The signal is located in the center of the tape. Therefore, a full track playback head is optimal for the best reproduction. If this is not available, use the two inner tracks of a 4-track playback machine transferred binaurally. Later, you can decide which track is the cleanest or make a decision whether or not the two tracks should be summed to mono. If a 4-track machine is not available, re-adjust the height (if you have the mechanical inclination to do so) of the playback-head on whatever reel-to-reel machine that you have access to until the tape players output signal is maximized. Also, adjust the azimuth for the best reproduction of the upper registers of the audio spectrum.

- **Acetate Backed Tape:** Acetate tape was used through the 1950's and halfway through the 1960's. If it is uncertain whether or not you are dealing with acetate-backed tape, hold the reel up to a light bulb looking through the layers of tape. If the view appears to be translucent, the tape is very likely to be acetate. This tape backing deteriorates over time, often giving off the odor of acetic acid (vinegar). This deterioration can cause playback problems, because the tape changes dimension, and often bows, not allowing it to pass over the playback heads of the tape deck evenly. This produces dropouts if not properly dealt with. The only way to deal with this problem is to increase the tape take up reel tension. Some sophisticated machines (3 motor tape decks) have electrical controls or internal adjustments to facilitate this. Three motor tape decks use the reel motors to establish the tape tension by applying low levels of DC to the motor field winding(s). Simpler machines will require more imagination on your own part. Another alternative is to install felt pressure pads on your machine's playback head. In either case, realize that the increased tension required to force the tape to pass evenly and flatly across the playback head will increase head-wear. But that may be a worthwhile price to pay if you are transferring a priceless tape for restoration and digital archiving.

- **Mylar and Polyester backed Tape:** Mylar and polyester backing provides a superior backing for magnetic audiotape. However, there were some manufacturing problems that were widespread in the US tape manufacturing industry using this technology in the mid to late 1970's over the 9-year period that followed. The problem involved a chemical breakdown of the binder, which affixes the oxide layer to the tape. This chemical breakdown process produces what has become known in the industry as "sticky shed syndrome". It leaves a residue on the tape path components of the tape player, and distorts the location of the oxide layer as it passes through the head assembly mechanism. This produces a permanent degradation of the frequency response of the tape recording and can also damage your playback machine. Therefore, a pre-processing step is necessary if you believe that you have a tape from this era having a Mylar or Polyester backing. The standard process is to bake the tape is a pre-heated industrial grade electric oven (between 120 to 125 degrees F) for around 4 to 7 hours (4 hours for ¼ inch and 7 hours for 2 inch tape). (The "H" fields produced by an electric oven are not high enough in level to modify the position of the tapes magnetic domains.) An industrial convection oven is ideal for this purpose, but a conventional electric oven will work also. If you are very careful, you can use a home electric oven, provided that you do not trust its thermostat to establish the baking temperature. Do not use microwave ovens; microwaves will destroy the tape. Do not use gas ovens because they actually increase the moisture level in the tape baking environment, potentially making the problem worse. Make sure that the thermostat is calibrated on your oven before proceeding. To do this, use a high performance reference thermometer located inside the oven in order to establish the temperature setting calibration. You must be sure that the temperature of the oven is stable before placing the tape therein. Allow the oven a good hour to stabilize in temperature before inserting the tapes. If multiple tapes are placed in the oven, they should be spaced away from each other by an inch or so. After the tapes have been baked, remove them from the oven and let them cool for about a day (24 hours). The tape(s) should be transferred to an archival media shortly after this process has been completed, since the tape(s) will begin to degrade quickly after performing this procedure.*

*__**Note:**__ Some plastic tape reels may deform using this process depending on their chemical make-up. To assure that this does not occur, it would be best to transfer the tape to a metal reel before proceeding with this process.

- **Cassette Tape:** Cassette tape transfers present special problems. Since the head gaps for cassette machines are very small, and the speed is only 1 - 7/8 inches per second, a slight miss-alignment of the playback head will produce a pronounced distortion of the stereo image and the fidelity of the upper registers of the musical scale. It may be necessary to use the "Time Offset" feature to correct tape azimuth mis-alignment problems or to directly re-calibrate the playback head azimuth alignment. The X-Y Vector plotter is very useful in this regard. Also, refer to the File Conversions section of this manual for details pertaining to the use of the “Time Offset” feature.

- **Analog Tape Hiss:** Analog tape hiss can be removed using the Continuous noise filter. A "fingerprint" sample of the noise will have to be taken before the noise can be removed. Refer to the Filters section of the manual for details.

Tape Dropout Repair (reel-to-reel)

One problem often plaguing old reel-to-reel magnetic tapes is tape warpage, which is one cause of tape dropout. Tape dropout is observed as a periodic loss of signal, especially in the high frequency portion of the audio spectrum. You can easily note if warpage is the source of the problem by observing the tape as it glides over the playback deck tape guides and heads. If the tape moves evenly over the guides and heads, then the source of the dropout may be lost magnetic coating. However, if the tape moves unevenly (wobbles) across the tape guides, the probable cause of the audio dropouts is tape warpage. The reason for the dropout is that the magnetic tape surface does not maintain good physical contact across the playback head thereby lengthening the magnetic coupling pathway (and increasing the magnetic reluctance) of the system. This will impact the top-end of the spectrum to the greatest degree (random fading in and fading out of the high-frequency signals). There are two solutions for curing this problem at the transfer stage of your project. If you have an old cheap tape recorder that used pressure pads, it will do a better job of transferring such a tape compared to a pro-machine which does not employ pressure pads. But, if all you have is a professional / high-grade machine, you can create your own homegrown pressure pad for the transfer. Follow these steps:

1. Remove the Tape Head Cover / Shield.
2. Thread the tape through the tape path as you normally would do.
3. Identify the play head (it will be the last head (right-most) in the tape pathway).
4. Obtain a small cotton ball and create a piece about ¾ inch square.
5. Cut a piece of Duct Tape to a dimension of about 2 x ½ inches.
6. Affix the cotton ball to the center of a piece of duct tape.
7. Wrap the tape with the cotton over the top of the head and down to the chassis of the machine, locating the cotton ball directly over the tape and the tape head.
8. You may have to experiment a little to obtain the proper tension for a clean playback, but this will dramatically reduce the dropouts of your old tape.
9. Replace the Tape Head Cover / Shield to minimize hum pickup during the transfer.
10. You will have to remove the tape and cotton ball and repeat this process after each tape side that is transferred in this manner.

Enhancing Reel-to-Reel Tapes Recorded at Slow Speeds

Sometimes, old reel-to-reel tapes were recorded at low speeds such as 3 ¾ ips (inches per **s**econd). These tapes lost much of the high end because the magnetic recording head used had relatively large gaps. These tapes can be brought up to a much higher level of performance by applying the following procedure in the prescribed order:

1. Check to be sure that the tape oxide side of the tape is facing towards the recording heads of your tape player. Sometimes a reel-to-reel tape has been accidentally re-wound with a 180-degree twist, causing the backing side to face the playback heads. The tape will play in some instances, but with very poor frequency response and output level (the recorded information may be backwards).
2. Digitize the tape and store it onto your Hard Drive.
3. Run the Virtual Valve Amplifier using the "Red Hot Jazz" preset.
4. Make the Destination the Source.
5. Run the Virtual Valve Amplifier again, this time using the "Bright & Brashy Brass" preset.
6. Make the destination the source.
7. The noise floor of the tape will now be substantially elevated, but so will the high end of the audio spectrum, so let not your heart be troubled. The next phase of the process will reduce the noise leaving behind a cleaner "top end."
8. Highlight a few seconds of the noise signature found at the beginning of the Destination .wav file.
9. Run the Continuous Noise filter to reduce the residual hiss. Go lightly with this filter to avoid digital artifacts.
10. Run the Dynamic Noise filter using the "3 ¾ tape hiss attenuator" preset to further reduce hiss.

Archival Recording Philosophy & Methodology

The archival recording process should be approached differently than audio restoration recording. The goals are quite different between these two concepts. When producing an archival recording, one should strive to capture all of the information possible on the source recording and then store its media under controlled environmental conditions. When performing audio restoration, one is involved in the modification of that source material by eliminating noises and modifying the spectral distribution of acoustical energy so as to be more pleasing to the ear. The philosophy behind archival recording is that the resulting product should contain as much information as possible for future use, and this includes all of the noises, clicks, pops, thuds, hum, buzz and any other extraneous noises from the audio source recording. These noises can always be removed at a later point in time without disturbing the master archive.

Applying any type of filter prior to producing the archival recording is a very bad practice. Transfer the recording either "flat" or via its known inverse equalization curve. If the inverse equalization curve is not known, the best fallback position is to transfer it flat. Also, if you are archiving a collection of monophonic recordings from a disk or cylinder source, it should be transferred by using a stereophonic magnetic phono cartridge. This way, both the left and right groove walls are recorded and maintained as individual entities, and since one may be in better shape than the other in certain areas of the transfer, you want to capture both. You should use the best equipment that you can afford for this task. Keep in mind that today's technology will always be poorer than the technology of the future. And in some cases, the only material that someone in the future will have to work with will be derived from what you have done today. Even if this statement is not exactly true, when you take an analog recording and digitally archive it, you are stopping the chemical degradation process dead in its track at that particular point in time. Perfect clones of your digital master can be reproduced every five years or so in order to insure that no further degradation occurs.

If you are working with old full track tape recordings, and do not have a high quality full track recorder available, use a half track machine and record both channels independently. You may find that one track contains some dropouts while the other does not. And even if both tracks have drop-outs, sometimes you will find that the drop-outs do not occur simultaneously, which will allow someone in the future to fix the problem using a tool like DCArt10/DC Forensics10.

<u>Note:</u> The digital archival recording standard established by the US Library of Congress is, at present, Linear PCM @ 96 kHz Sampling Rate in conjunction with 24 Bit Resolution (.wav meets the requirement when set to these parameters during digitization).

CDR Prep from a Commercial Cassette Tape

1. Record side one of your cassette tape, creating one continuous .wav file at a 16 or 24 bit 44.1 kHz sample rate.
2. Record side two of your cassette tape, creating a second continuous .wav file at a 16 or 24 bit 44.1 kHz sample rate.
3. Using the Mute and/or Cut function found under the Edit Menu, eliminate excess lead in from the two .wav files. Do the same thing at the end of the .wav files to eliminate excess "dead-time."
4. Run the Continuous Noise filter to attenuate wideband noise such as hiss and rumble.
5. Enhance the recording by running any of the audio enhancement tools such as any of the equalizers, dynamics processor, reverb, or Virtual Valve amplifier.
6. Click on the CD Prep Menu.
7. Click on Find and Mark Silent Passages.
8. Check to be sure that the markers have located themselves correctly between "cuts." Adjust them with your mouse if necessary.
9. Click on "Quantize for CD Audio."
10. Click on "Chop File into Pieces."

CDR Prep from a Vinyl Record

1. Record side one of your record, creating one continuous .wav file at a 16 or 24 bit 44.1 kHz sample rate.
2. Record side two of your record, creating a second continuous .wav file at a 16 or 24 bit 44.1 kHz or 88.2 kHz sample rate.
3. Using the Mute and/or Cut function found under the Edit menu, eliminate excess lead in time and the "needle-drop" from the two .wav files. Do the same thing at the end of the files in order to eliminate the "needle-lift" and excess "dead-time."
4. Run the Impulse noise filter using vinyl mode on each file in order to remove ticks and clicks.
5. Convert the file to 44.1 kHz sample rate at a 16 bit depth (if it is not already in that format).
6. Run the Continuous noise filter to remove wideband noise such as tape hiss.

7. Enhance the recording by running any of the audio enhancement tools such as either of the equalizers, dynamics processor, reverb, or Virtual Valve amplifier.
8. Click on the CD Prep Menu.
9. Click on Find and Mark Silent Passages.
10. Check to be sure that the markers have located themselves correctly between "cuts." Adjust them if necessary.
11. Click on "Quantize for CD Audio."
12. Click on "Chop File into Pieces."

Characterize Frequency Response / Equalize Audio Response
(Forensics only)

The following describes how you can use your Diamond Cut software to characterize the frequency response of an audio system including the acoustical environment in which it operates. This method involves the use of random (white) noise in conjunction with the Spectrum Analyzer used in "Live" mode. The general theory is to stimulate the sound system and its acoustical environment with a known distribution of noise (White Noise - constant energy per unit Hertz, in this case). Then, a signal representing a portion of the acoustical output of the system will be fed back into the software program via a microphone and a full duplex soundcard in order to measure the resultant response. The Hi-Resolution Spectrum Analyzer will display the real time result of this measurement. If the sound system has an equalizer in the signal chain, it can be used to "flatten out" the measured response while viewing the Spectrum Analyzer. The following equipment will be required to perform this measurement and/or system equalization:

Equipment:

1. High quality full-duplex sound card.
2. Flat responding Electret microphone ($20.00 at most electronics dealers) and stand.
3. Diamond Cut Forensics10 Software.
4. Intel Pentium 4 Computer or better.
5. Sound system (preferably with a graphic equalizer in the signal chain) to be evaluated.

6. Optional: In some cases you will need a small mixer with a microphone input if your sound card is not designed to handle microphone level inputs (in other words, the sound card only has a line level input).
7. Optional: 1/3rd octave 30 band or 5 band parametric equalizer inserted in the signal path of your sound system. (Please note that an octave-weighted 10-band equalizer is not particularly well suited for this application.) One of these devices is needed only if your goal is to equalize your sound system. If you are merely interested in characterizing your system, an equalizer is not required.

Setup:

1. Set the microphone on a stand a few feet in front of the listening spot most often used centered between the two speakers.
2. Connect the output of the microphone to the mic input of your sound card. If you do not have a microphone input, you will need a mixer having a microphone input to take the signal up to line level in order to be compatible with the line input of your sound card. In that case, the microphone would be connected to one of the microphone inputs of the mixer, and the line output of the mixer should then be connected to the line level input of the sound card.
3. Connect the output of the sound card to an auxiliary set of line level inputs on your sound system.
4. Defeat all tone controls on the sound system. If there is an equalizer in your system, make sure that it set for "flat".

Procedure:

1. Create a 10-minute Stereo .wav file containing white noise. This is accomplished by using the "Make Waves" feature found under the "Edit" menu. Set the parameters as follows:
   a) Wave shape: White Noise
   b) Length: 600 Seconds
   c) Amplitude: -10 dB
   d) File Type: Stereo
   e) Sample Rate: 44.1 kHz

2. Run the generator by clicking on "OK". After the computer performs this process, you should see a signal in the Source display. This process may take a minute or so.
3. Make sure that the "Output Device I/O" is set to the high performance sound card in your system and that your audio system input is connected to the sound card output.

   **See the optional System Stimulus method outlined below.*
4. Open up another copy of DC Forensics10.
5. Set the "Output Device I/O" of the second version of the program to a sound card other than the one that you are using for the test. This is found at the bottom of the "Edit" menu. Perhaps, choose the sound card that came with the computer for this setting since it does not have to be high performance.
6. Set the "Input Device I/O" of the second version of the program to the high performance sound card in your system.
7. Click on the "Live" button on the second version of the program.
8. Click on the View Menu and bring up the Spectrum Analyzer.
9. Set the spectrum analyzer as follows:
   a) Display Mode: Slow
   b) Range: 50 dB
   c) Frequency Bands: 4096
   d) Frequency Resolution: 2.69 Hz
10. Activate the system by clicking on "Live Preview" in the Multi-Filter box.
11. Using the "Offset Control" on the Spectrum Analyzer, center the signal on the screen.
12. Play and loop the .wav file using the first version of the program that was used to create the .wav file. Adjust the volume on your sound system until the "hissing" sound is at a moderate listening level.
13. Re-adjust the "Offset Control" on the Spectrum Analyzer for the best view of the signal.

The Spectrum Analyzer should now be displaying the frequency response of your system. The horizontal (x) axis indicates frequency and the vertical (y) axis indicates relative amplitude in dB. If the displayed signal is not flat and you desire it to be, you can engage your systems' graphic equalizer and adjust it until the spectrum display is as "flat" as possible.

***Optional System stimulus method:**

Rather than using two versions of the software program operating at the same time on your computer, you can create a CD ROM of the .wav file containing the white noise signal. Play this CD on your system CD player while using the spectrum analyzer operating in "Live" mode. This method has the added potential benefit of including your sound systems' CD player in the measurement results.

Converting White Noise into Pink Noise

1. Under the Edit menu, click on "Make Waves".
2. Select "Random" noise, with the amplitude set to -10 dB, and the length set to 5 or 10 seconds.
3. Click "OK", and a random white noise file will be created in the Source window.
4. Click on the "Filter" Menu and then the Paragraphic Equalizer.
5. In the preset menu at the bottom of the Equalizer, find and click on the "White to Pink Noise Converter" and then run that filter.
6. Pink noise will now be found in the Destination window.

De-clicking a Vinyl LP Record with the EZ Impulse Noise Filter

This procedure makes use of the EZ Impulse Noise Filter. In extremely rare and difficult cases, consider trying the Expert Impulse Noise Filter.

1. Download the Vinyl LP .wav file into DCArt10/DC Forensics10 using the "Open" command.
2. Highlight the portion of your .wav file on which you desire to apply the EZ Impulse Noise Filter. (You may also choose to highlight the entire file.)
3. Click on the Filter Menu and then "EZ Impulse Noise".
4. Set all three controls to their mid-position (50) as a starting point.
5. Adjust the "Scratch" control to reduce the large scratch impulses, but not so aggressively as to produce distortion. If extremely large gouges exist on the recording, check the "Big Clicks" checkbox.
6. Set the "Crackle" control to reduce high density medium sized impulsive noise, but not so aggressively as to produce distortion.

7. Set the “Narrow Crackle” control to a value where small ticks are eliminated, but not to the extent wherein high-frequency loss or distortion occurs.
8. Run the filter or use Preview mode, and determine if the .wav file is being adequately de-clicked. If it is not, raise the appropriate setting(s). If it is de-clicking, but producing distortion on the sibilant sounds, then lower the various control(s). Continue this process until a good balance is established of minimum sibilant distortion and minimum click feed-through. If it is de-clicking, but leaving a bit of an artifact behind, try turning on the “Solo/Brass” feature.
9. When you determine the best setting(s) of the control(s) for your particular .wav file, click on Run filter. When the filter has completed its operation, the results will appear in the "Destination" workspace.

Important Note: The EZ Impulse Noise filter works best on .wav files that have been sampled at 44.1 kHz or higher. The setting examples given below are based on .wav files that have been recorded at a 44.1 kHz sampling rate only. Different sampling rates will require different settings.

De-clicking a Vinyl LP Record with the Expert Impulse Filter

This procedure makes use of the Expert Impulse Noise Filter. You may also want to try the powerful EZ Impulse Noise Filter for quick and pain-free de-clicking.

1. Download the Vinyl LP .wav file into DCArt10/DC Forensics10 using the "Open" command.
2. Highlight the portion of your .wav file on which you desire to apply the Impulse Noise Filter. (You may also choose to highlight the entire file.)
3. Click on the Filter Menu and then “Expert Impulse Noise”*.
4. Click on the "Vinyl LP" checkbox.
5. Set the Threshold Control to its minimum setting of 1.
6. Set the "Tracking" Control somewhere in the 25 to 30 range to start.
7. Set the "Size" control to a value somewhere in the range of 10 to 15 samples.

8. Run the filter or use Preview mode, and determine if the .wav file is being adequately de-clicked. If it is not, lower the tracking control. If it is de-clicking, but producing distortion on the sibilant sounds, then raise the control. Continue this process until a good balance is established of minimum sibilant distortion and minimum click feed-through. If it is de-clicking, but leaving a bit of an artifact behind, increase the value of the "Size" control.
9. When you determine the best setting of the control for your particular .wav file, click on Run filter. When the filter has completed its operation, the results will appear in the "Destination" workspace.

Important Note 1: Vinyl LP mode works best on .wav files that have been sampled at 44.1 kHz or higher. The setting examples given below are based on .wav files that have been recorded at a 44.1 kHz sampling rate only. Different sampling rates will require different settings.

Important Note 2: Unlike all of the other DCArt10/DC Forensics10 controls, Vinyl LP mode cannot be turned on or off "live" in Preview Mode.

***Important Note 3:** Alternatively, you can use the EZ Impulse Filter which is very effective on Vinyl LPs, 45s, or 78s. The EZ Impulse Filter is probably the best place to start except in extreme impulse noise or forensics audio (static) circumstances.

Manually de-clipping an over modulated Wave file*

Sometimes in the process of digitizing an analog signal into a .wav file, clipping can occur if proper attention had not been paid to the signal level during the recording process. Clipping is the process wherein the analog signal's amplitude is occasionally requiring more than the full-scale capability of the A/D converter in the recording system. This produces "flat-topping" of signals at their peaks that can yield a harsh sounding distortion upon playback if it happens too frequently. ***An automatic De-Clipping filter is now included with this product, but some prefer to perform the de-clipping function manually.*** The following is a procedure for manually de-clipping the sections of an over-modulated digital signal:

1. Open the clipped .wav file in the Source workspace.

2. Using the Gain Change algorithm (Edit Menu) reduce the amplitude of the signal by changing the gain of the entire .wav file by -6 dB.
3. The resulting areas of the signal that have been clipped will be obvious after you perform this operation by simple inspection of the Source display. The signals which had been clipped will appear as if they have had a "crew-cut"
4. Zoom-In on the areas that appear to have been clipped until you see the actual "flat-topping" of the problem section of the .wav file.
5. Highlight the area of the waveform that is "flat-topped." This process must be performed one clipped event at a time. You should not highlight a section of the waveform that has multiple "clipping" events for this process to work. Be very careful to highlight only the area that is clipped (flat-topped), making sure not to highlight too much of the signal on each side of the clipped section of the waveform.
6. Depress the "I" key (to interpolate the peak of the signal).
7. If you are de-clipping a stereo .wav file and prefer to apply the correction to only one channel, use "Ctrl-I" to interpolate the right channel only or the "Shift-I" to interpolate the left channel only.
8. You will see that the replaced signal has a rounded top rather than a flat top. The rounding of the flat top substantially decreases the distortion produced due to the clipping at that point in the signal.
9. Continue looking through the .wav file for all signals that are "clipped," repeating the procedure outlined above.

*Note: Alternatively, consider using the automatic de-clipper (found under the Forensics Menu) which often will do an effective de-clipping job on most material.

Decode Touch Tone Signals into Alpha-Numeric Values

1. Highlight the section in the recorded conversation containing the DTMF touch tone signals.
2. Play looped this entire section of the .wav file with the VU meter turned on (View Menu).
3. Using the Adjust Gain feature (Edit Menu), set the signals to indicate approximately –8 dB on the VU meter at the right of the screen while playing.
4. Bring up the Paragraphic Equalizer (Filter Menu).
5. Find the preset labeled DTMF Comb Filter (Normal)*.

6. Bring up the Spectrum Analyzer (View Menu).
7. Set the Spectrum Analyzer up in the following manner:

    A. Slow Mode
    B. Show Peak Enabled
    C. Resolution = 0.67 Hz
    D. Frequency Bands = 4096
    E. Range = 100 dB

8. Highlight and then Preview the first (or following) tone burst using the preview button in the Paragraphic Equalizer‡.
9. Note that there are two major spikes on the Spectrum Analyzer. The peak detector will have automatically detected one. Click on the other spike so that a square box marker is following it.
10. Note that these two frequencies are displayed in the right and left corner of the Spectrum Analyzer display. Write down their values.
11. Continue this process (back to step 8) with the next tone burst, writing down each frequency pair.
12. When done, go to the DTMF chart in the glossary/appendix portion of this manual. From this chart, you will be able to decode the frequency pairs into a telephone number.

***Note 1:** Alternatively, you can select DTMF Comb Filter (Narrow) in very noisy situations. Also, you can choose DTMF Comb Filter (Ultra-Wide) in situations where the tone frequencies are off tolerance.

‡Note 2: The Forensics Brick Wall filter also has a special DTMF Band-pass Filter which can be used instead of the Paragraphic Equalizer when you encounter situations where there is an extreme amount of out-of-band noise which needs to be rejected.

Forensic Tape Authentication

(Forensics Version Only)

Audiotape authentication is necessary if it is to be used in a legal proceeding. The process of making this determination is called tape authentication, meaning to determine that the tape specimen is the original.
Two separate "hum" signals existent on the media usually suggests the presence of a "dub." The reason for this is related to the imperfect

nature of analog tape recorders that are commonly used in the forensics world due to their small size. These recorders pick up small stray electromagnetic waves from the power distribution grid in proximity to the machine. An "original" recording will have just one noise signature at either 50 Hz or 60 Hz or harmonics thereof depending on the line frequency of the power grid of the environment in which the recording was made. Since analog recorders produce a certain amount of tape slippage, this frequency will be slightly offset from the actual line frequency. If a "dub" is then made of the master, a new noise frequency will appear on the specimen being analyzed with the spectrum analyzer (or the high definition Spectrogram). For example, if you see two spectral lines at, let's say, 60.1 and 60.4 Hz, this usually indicates that the test specimen is a dub. On the other hand, if you only see one spectral line, the tape is usually an original. To perform tape authentication utilize the following steps:

1. Make a Hard Drive copy of the tape on DAT &/or on your Hard Drive. CDR backup is also recommended.
2. Bring up the .wav file of the tape.
3. Bring up the Spectrum Analyzer (View Menu).
4. Set the FFT Size to 131,072
5. Set the Start Frequency for 40 Hz
6. Set the Stop Frequency for 70 Hz
7. Highlight a relatively quiet section of the .wav file, but one that is within the context of the recorded conversation and use Loop Play so that the analyzer can build its data for presentation.
8. Use the Range Control setting at 20 dB and the Offset control to zoom in on the signals of interest at around 50 or 60 Hz.
9. If there is only one signal around that particular frequency, then the tape is usually an original.
10. If there are multiple frequencies around the line frequency, then the tape is usually a dub.

Important Note:
The power grid in the United States & Europe can have an instantaneous frequency variation of up to +/- 0.5 Hertz for up to a 10 second interval. However, when measured and averaged over a 24-hour time interval, the frequency tolerance is much tighter. Analysis of a recordings hum frequency vs. the power grid frequency is possible

based on historical frequency records of a power grid. The technique known as Electric Network Frequency (ENF) analysis, is helpful when forensic scientists need to separate genuine, unedited recordings from those that have been altered. For details, please refer to Diamond Cut Productions Applications Note titled – **"AN4 - Using DC Live/Forensics: The Electric Network Frequency Analysis".** It can be downloaded from the Diamond Cut website which is located at www.diamondcut.com.

Automatic Micro-cassette Tape Start-Stop Sequence Detector

(Forensics Version Only)

Forensics examiners are often confronted with the problem of authenticating a recording. Often, these recordings are of the micro-cassette variety. At issue is the question of whether or not the tape has been edited or modified in some way. Often, tapes are edited on a different machine rather than on their original recording device. If a different machine was used to edit the tape, and since each machine leaves its own distinctive stop / start pattern when viewed in the time domain (display), it can be determined if the start / stop breaks were all made on one machine or two or more.

But, often the examiner is confronted with hours of tape and needs to focus in on the start / stop sequences rather than sitting through the entire tape(s) looking for these interruptions.

The best utility to use for this is the Subsonic Explorer (found under the Forensics Menu/Authentication). Please read that section of the Users Guide for more information.

Alternatively, the "Microcassette Tape Start-Stop Detector" Multifilter preset will provide a way to Mark the file where these breaks occur so that a close examination of the file can be made in the time domain display without listening to the entire file(s) to look for recorder erase / record head signatures on the recording.

Here are two methods that one can use to operate this Multifilter Preset.

Method 1:

This method is the easiest to use, but is not self-documenting. In Classic Edit mode, run the attached preset on the Source file with the system (View Menu) set for Sync mode. Each start / stop sequence will be identified by a very large rail to rail pulse which shows up in the destination display. You can then zoom-in on the various start / stop events in the Source Display for further study.

Method 2:

This method is a bit more complicated to use, but it is self-documenting (meaning that the audio material and the markers are integrated into the same .wav file).

1. Use the File Converter Filter to make a Stereo File out of the Mono one that you have from the Microcassette recording.
2. Make the Destination the Source.
3. Clone the File.
4. Run the above mentioned Multifilter preset on either the Left or Right Channel only.

Results: You will see a large pulse appearing on one channel wherever there is a Microcassette start / stop sequence. The pulse should be clear and obvious and "rail-to-rail" in amplitude.

Attenuating GSM Cell Phone Noise from Forensics Recordings

(Forensics Version Only)

There are two common cell phone standard systems in wide use. One of the two, GSM (Global System for Mobil Communications) can cause interference to electronic equipment proximal to a cell phone (by a process involving Amplitude De-Modulation from non-linear devices interacting with Miller capacitance(s) in your recording device or other electronic system(s)). The high crest factor pulsing sound produced by these cell phones can overwhelm a Forensics audio recording making it indiscernible in its native format. Your software has a dedicated Cell Phone Noise Filter found under the Forensic Menu designed to attenuate this noise type. Refer to that section of this documentation for details. If that filter is not effective for your purposes, consider using the following alternative.

There are some Multifilter presets that may be useful for cell phone noise attenuation. Here is a procedure to try on recordings plagued with this type of cell phone noise that were not able to be reduced by the Cell Phone Noise Filter:

1. If the recording is not sampled at 44.1 kHz, 16 bit, then convert it to the stated format using the Change Sample Rate/Resolution feature found under the Edit menu.
2. Normalize the Gain of the file to -4 dB using the Normalize Gain Scaling feature found under the CD Prep Menu.
3. Bring up the Multifilter.
4. Find the Multifilter Presets having “Cell Phone Noise Interference Attenuator - x” as a prefix. You have 6 to choose from, with each one in the numerical sequence being progressively more aggressive.
5. Preview each one and choose the one that performs the most effective noise reduction of the GSM cell phone noise without damaging the target audio signal.
6. If needed, adjust filters for the optimal results on the target audio signal.
7. Run the best filter.
8. Perform the usual post-processing steps to clarify the speakers voices using whatever your favorite Forensics filters may be.

Note 1: These Multifilters require a great deal of processing power. As a result, your CPU may not be able to keep pace with them in real time in “Preview” mode. Thus, you may hear some stuttering when previewing. This stuttering will not be heard after the filter has been “Run”.

Note 2: CDMA (Code Division Multiple Access) phones do not cause this type of interference to other electronic equipment.

Security Tips for Forensics Audio Laboratories

Here are a few security tips one should consider when setting up a Forensics Audio Laboratory:

1. The laboratory should be secured in a manner in which only authorized personnel are permitted entry into the Forensics audio laboratory proper.

2. When unauthorized persons need to gain access to the laboratory area, written protocols should be established and strictly followed to allow clients and visitors entry into the laboratory including sign-in/sign-out sheets. But, these persons should never be provided with unfettered access to the lab; these persons should be made to leave the lab area when there are no authorized persons therein.

3. Unauthorized persons should never be left alone in a Forensics Audio Lab. This is very important and that is why we repeat it.

4. A security system should be installed in the laboratory portion of your Forensics facility. This should include intrusion detectors on windows, passive infrared detectors, and a keypad or biometric entry system coupled to an electrically activated locking system on the labs entrance.

5. An independent computer system (not associated with your Forensics Audio Workstation) should be used to monitor the security system. This system should be hard-wire monitored by a central security service.

6. Video cameras located in the lab area are recommended and should be enabled 24/7.

7. Your Audio Forensics Workstation computer(s) should NOT be connected to the internet in any way. Wireless connections to these workstations should be disabled. You need to be able to assure clients and the legal process that files on these computers were not tampered with in any way. One way to reduce that possibility of tampering is to eliminate outside communications connections with your Forensics Audio Workstations.

8. Another major security risk is portable USB drives. Extreme care needs to be taken when using a portable USB drive. These drives can

carry virus files to your system and be used to extract critical information. The use of USB (thumb-drives) should be controlled.

9. Your Forensics Audio workstation(s) should not be networked with computers outside of the audio forensics laboratory proper. The network within the lab should be hard-wired via a hub and not wireless and should have no internet connectivity.

10. Keep only software programs that are absolutely necessary for your Forensics Audio work on your Digital Audio Workstation (DAW). Billing and admin computers should be located elsewhere in the facility and not networked with the Forensics audio workstation(s).

11. Software registration should be performed manually. Use an administrative computer outside the lab to obtain the necessary registration codes for your software and "sneaker-net" that information to your Forensics Audio Workstation(s).

12. Master (originals) audio materials should be stored in a fireproof safe. Backup copies of these audio materials should be kept in a secure area off-site. Audio materials should always be kept under lock when not being transferred to your Forensics Audio Workstation Computers.

13. Client Audio materials should never be left unattended outside of the laboratory proper.

14. Master hard-copies of your clients source materials, audio restoration software and work-product files should be kept in the laboratories fireproof safe and this safe should be locked when the hard copy materials are not needed.

15. Cell phones should not be allowed to be used within a Forensics Audio Laboratory. They should be turned off (not just silenced) or left outside the laboratory proper. This rule should apply to authorized personnel and visitors to the lab.

Forensics Audio Handling & Chain of Custody

If you are dealing with Forensics Audio materials or evidence, certain protocols need to be followed. There are several reasons for this.

1. The need to protect the evidence from potential tampering by anyone, especially third parties.
2. The need to prevent accidental damage from occurring to the materials while in transition or while they are in your possession.

3. The need to survive legal scrutiny under examination in a legal venue pertaining to the proper handling of the materials.

In terms of maintaining a traceable "Chain of Custody", the following procedures should be followed:

1. Shipment to or from a client should be traceable via Registered or Certified US Mail.

2. A packing slip or letter should be included with the shipment describing the material enclosed.

Certain precautions should be followed when shipping Forensics Audio materials to assure its integrity throughout the process. The following recommendations should be followed:

1. Do not ship the audio materials in the same container with any other non-related items.

2. Six inches of packing material should be provided around the extreme dimensions of the media. This is to minimize the effects of shock and magnetic fields on tape based materials such as analog tapes, diskettes, or hard drives.

3. Use only non-shedding packing materials such as bubble wrap. Do not use shredded newspaper or anything similar since fibers can negatively affect tape mechanisms.

4. Use anti-static packing materials whenever possible (like pink bubble wrap).

5. Keep a written and photographic record of the packing process used and maintain that in a document folder for your client's project.

When you receive Forensics Audio materials from a client, it is advisable to follow this procedure:

1. Immediately inspect the outside of the package for damage. If there is any damage to the package, photograph it and document your findings in writing and place that documentation in the clients project folder. Immediately notify the carrier of the damage noted on the shipping material and inform your client of the damage noted.

2. Open the container and inventory the contents found therein. Compare the contents against the packing list or letter. If there are any discrepancies, note these in writing, place those notes in the client's project folder and inform your client of your findings.

3. Save all shipping labels & documents attached to the shipping materials/package.

4. Carefully inspect all packing material to be sure nothing of evidentiary value is discarded.

5. Photograph the recording(s) media as received.

6. Date and mark the recording(s) for identification.

7. Conduct a detailed physical examination of the recordings. Points of interest should be photographed. Keep detailed written notes as you examine the materials and maintain those notes in the client's project folder.

8. If not already removed, remove the safety recording tab(s) from Cassette tapes before playing. Do not discard the tab(s). Put each tab in a separate envelope. Identify the original location of each of the tabs. Seal the envelopes and document their contents and keep this in the client's project file.

Use the following guidelines whenever handling a client's audio evidence:

1. Use white cotton gloves whenever handling Forensics Audio

materials (like tapes or discs). Alternatively, you can thoroughly wash your hands prior to handling the materials. Only handle the material when absolutely necessary for examination, playback, or return to the client. Avoid any unnecessary handling of the material and be sure not to touch the actual media recorded surface with a bare hand or ferrous tool.

2. Create a direct digital copy of the material onto your DAW. Perform all of your analysis using the digital copy rather than working continuously with the original. Using the Diamond Cut recorder, transfer the recording with 24 bit resolution and a 96 KHz sampling rate. Keep a written record of the time that the recording was transferred and the file name assigned. Make a backup copy of the digital recording on a CD, DVD, USB thumb drive, or external hard-drive. Keep this copy in a secure location off-site.

3. Always be sure to keep magnetic media away from all sources of magnetic fields such as computer monitors, loudspeakers, permanent magnets, power amplifiers, backup power supplies, power tools, etc…

5. Make sure that the analog playback machine's heads and tape guides had recently been de-magnetized (degaussed). This is an operation that should be performed outside the forensics laboratory proper and at least ten feet away from all client magnetic based media.

   When you are not using the Audio Evidence, you should abide by the following rules:

   1. Limit access to the material to those in your laboratory who have a need to know based on their involvement with the project.

   2. Keep the material in a fireproof safe which is kept locked. Preferably, this safe should be located in the basement of your building near a 90 degree corner of a foundation wall.

   3. Assure that the material is kept in a constant temperature and constant humidity environment, keeping it stored away from any source of magnetic fields. Ideally, storage temperatures should be around 60 degrees F and the relative humidity should be 40% or less.

Quality Assurance for Forensics Audio Laboratories

System Maintenance and Calibration

The operators of Forensics audio laboratories are the solely responsible entities to assure the performance of their system and it is a good practice to assure that the software and its associated hardware are properly maintained and calibrated to a traceable standard. This is especially important in mission critical applications involving law enforcement or the legal system or other similar applications. The licensee is responsible to their customers to assure that their software is up to date; periodic updates and upgrades are available from the Diamond Cut Productions, Inc. website. It is the licensee's responsibility to periodically check for software updates and upgrades, and when they become available, these updates and upgrades should replace older versions. System hardware involved in the process of the use of the software should also be maintained properly. The licensee's Digital Audio Workstation System(s) (or equivalent) including its hardware and software should be calibrated by a recognized certified working standards laboratory and traceable to a government regulated standards organization to assure proper calibration of the system. This process should be performed at regular time intervals determined by the body of law governing the geographical area in which the software and hardware are to be used. An example of a Standards organization is NIST (National Institute of Standards and Technology) in the United States of America. Log books should be kept indicating maintenance events including preventative maintenance, updating or upgrading of the software and/or calibration at regular time intervals including the replacement of any hardware components of the system and their re-calibration.

Gain Riding Procedure

1. Highlight the portion of your .wav file during which you would like to "slew" (change) the gain up to or down to a new value. (If this is to be done on a relatively small portion of the .wav file, it may be useful to Zoom-In on the sector of interest.)

2. Press the Play button to audition your selection.
3. Click on either "Fade-In" or "Fade-Out." (Edit Menu)
4. Choose the type of slew curve you prefer, either Linear or Logarithmic.
5. Set the "Start Level" slider control to unity gain.
6. Set the "Stop Level" slider control to the desired new gain for the segment of interest. This can be up to + 6 dB (amplification) or up to - 96 dB (attenuation).
7. Click on OK. The gain "slewing" will be performed during the highlighted portion of the .wav file.
8. Highlight the portion of the .wav file that you desire to remain at the new gain setting. This will be the "dwell" sector for the gain riding procedure.
9. Again, click on either "Fade-In" or "Fade-Out."
10. Next, change the "Start Level" slider to the same value as the "Stop Level" slider. This will be your new gain setting (dwell).
11. Click on OK. The gain modification will be applied to the highlighted portion of your .wav file for its highlighted duration or dwell time.
12. Next, select the portion of your .wav file during which you desire the gain to slew back down to its original value (or another new value).
13. Again, click on either Fade-In or Fade-Out.
14. Change the "Stop Level" slider back down to unity gain (or whatever new gain you desire) leaving the "Start Gain" setting where it had been.
15. Click on OK. The gain will slew back to the unity gain value during the highlighted time interval.

Selective De-Clicking file sectors with the Impulse Filter and "Sync Mode" and Classic Edit Mode

1. Place DCArt10/DC Forensics10 into "Sync" Mode (View Menu).
2. Automatically De-Click the entire .wav file in the standard manner utilizing the Expert or EZ Impulse Filters.
3. Listen to the result in the Destination workspace. Listen for locations that contain pops or clicks which were too severe for the Impulse filters algorithms to conquer.

4. Zoom-In on the zone containing clicks that you desire to remove.
5. Raise the aggressiveness of the settings of the Impulse filter of choice and run the filter. Observe in the Destination Workspace to see if the increased sensitivity was able to remove the impulse noises in that file sector.
6. Keep repeating the above process until you are satisfied that the noises have been removed and replaced with a reasonable looking and sounding replacement set of waveforms/signals.

Impulse Noise Generation

(Forensics Version Only)

Sometimes it is desirable to be able to generate Random Impulsive Noise. One application involves room acoustical propagation delay testing. Another application is theatrical in nature; that is to say that a modern recording may need to have some "aged patina" added to give it the sound of a time gone by and contemporaneous with the time of a particular theatrical performance.

The following Presets can be found under the Multi-Filter for Random Impulse Creation:

- **White to Impulse Noise Converter 1** (45 RPM Low Cost Record Player Simulator)
- **White to Impulse Noise Converter 2** (33 RPM Low Cost Record Player Simulator)
- **White to Impulse Noise Converter 3** (45 RPM Hi-Fi Quality Record Simulator)
- **White to Impulse Noise Converter 4** (33 RPM Hi-Fi Quality Record Simulator)
- **White to Impulse Noise Converter 5** (Radio Static Simulator / Acoustical Test Signal)
- **White to Impulse Noise Converter 6** (78 RPM Record Simulator *)

The first set of 5 presets will allow you to generate Random Impulses using the DC Forensics version of the software. You will need to start with a "Makes Waves" file consisting of Monophonic, 44.1 kHz, 16 bit Random noise at the -10 dB level. Next, apply the resultant signal to

any of the first five presets in your Multi-Filter and the result will be randomized impulses of varying characteristics. This signal can be used to simulate an olde recording by "Paste Adding" it to a modern recording. (Please refer to the Bandpass Filter preset listing for additional theatrical sound simulations.) If you do not have the DC Forensics version of the software, download the DC Forensics / Forensics demo from our website located at www.diamondcut.com and use it to create the file that you desire. If you need more than a one-minute duration of random impulses, use the concatenate function in your DCArt10 software to elongate it.
*The sixth preset (78 RPM Record Simulator) requires a different stimulus signal that can be created using the following procedure:

1. Create a Make Waves file of Random Noise, - 10 dB, 16 bits, 44.1 kHz, Mono, 30 seconds long (which represents 39.00 revolutions of the record).
2. Copy that file to the clipboard.
3. Create a Make Waves file of Sine waves, 1.3 Hz, -30 dB, 16 bits, 44.1 kHz, Mono, 30 seconds long.
4. Paste Mix this file to the one on your clipboard.
5. Apply the White to Impulse Noise Converter.
6. Preview and/or Run that filter preset on the file that you had just created.

The resultant file, (or concatenated groups of files) when added to a musical source, will give you an excellent simulation of 78 RPM record crackle which can be useful if you are trying to create an artistic effect for theatrical purposes.

If you need to perform acoustical room propagation delay testing, use White to Impulse Noise Converter 5. If you only want one impulsive event every 10 seconds, use the Edit/Mute function to establish that relationship. If you need to modify the shape of the impulse(s), Zoom-In and use the Pencil Tool to make the desired changes.

Note: If you desire a stereophonic simulation of Impulsive Noise, simply "Make Waves" in stereo instead of mono. Then, use the file conversion filter to create stereo using a time offset value of 20 mSec before applying the stimulus signal to the White Impulse Noise Converter preset of choice.

Nudging the Highlighted portion of the Workspace

1. To Nudge the right-hand side of a highlighted portion of a DCArt10/DC Forensics10 .wav file, merely use the left and right arrow keys on your keyboard.
2. To Nudge the left-hand side of a highlighted portion of a DCArt10/DC Forensics10 .wav file, depress the Shift key while operating the left and right arrow keys.
3. To change the resolution of the Nudge feature (the number of samples per nudge) go the preferences section of the Edit Menu, and enter the desired value of samples for each nudge.

Stanton 500 RIAA Compensation Curve Preset for the Multifilter

Under the Multifilter you will find a preset called "Stanton 500 RIAA Compensation Curve". This preset is to be used in conjunction with a Stanton 500 cartridge and a "Flat" phono preamplifier. This preset will not only convert a flat transfer using a Stanton 500 to the RIAA curve, but will also compensate for deficiencies in the flatness of response of the phono cartridge itself. This was accomplished through the use of a test record made by "Hi-Fi News" in conjunction with the Diamond Cut Spectrum Analyzer. The Stanton 500 was chosen for compensation because it is very popular, readily available, relatively low in cost, and can be outfitted with LP and 78 styli. It is important to emphasize that this preset is not to be used with a transfer through an RIAA preamplifier, but only via a Flat preamplifier such as the CTP series.

Record Transfer to Hard Drive Technical Hints

The first order of concern when transferring records to your hard drive should be your system setup's electrical hygiene. Also, it is important to apply the proper EQ to the transferred file. Proper cartridge termination is also critical to success. Here are some things to consider in those regards:

- Your turntable, since it most likely will have a magnetic cartridge, should be kept at least three feet away from your monitor, because the stray electromagnetic fields created by the monitor's deflection circuits can be a source of noise entering your pickup.

- Another important consideration is to be sure that your turntable’s chassis is grounded to your pre-amplifier chassis through an independent grounding wire (not the phono cartridge shielded cables) in order to minimize hum pickup.
- Shielded cables should be used for all audio interconnections with the exception of the loudspeaker connections to your power amplifier.
- All power supply cords should be fed from the same wall outlet. This is done in order to minimize any possible ground loops that could occur if this technique is not followed. In other words, your computer including its monitor and printer, and your turntable, pre-amplifier, audio power amplifier and any other system accessories should all be plugged into the same multiple outlet strip. The outlet strip is then plugged into a source of power, such that all of your equipment is operating off of the same wall outlet. This is especially important when your audio equipment is of the three-wire variety (with a safety ground). This will be encountered more often with professional grade audio equipment as opposed to consumer grade equipment. If this technique is not followed, a ground loop may be created between your computer's ground connection and your audio system ground connection. When this occurs, a noise current will flow in your audio cable shields with a consequential noise Voltage drop appearing across the same. This can induce a hum and / or buzz into your recording. If it is not possible to operate all of your equipment on the same electrical outlet, then you should consider using an audio isolation transformer between your audio pre-amplifier output and your computer sound card input to break this ground loop.
- When transferring records utilizing a magnetic phono cartridge, it is useful to "equalize" the frequency response of the pre-amp system before transferring the sound to your hard drive, unless the original source material was acoustically mastered. There have been many different equalization curves (RIAA, Decca, etc.) employed by the various record manufacturers. The primary purpose of these various equalization schemes was to allow for a fuller bass response

without producing an over modulation of the record groove wall. The frequency below which the cutter system reverted to a "constant displacement" as opposed to a "constant velocity" is known as the turnover frequency. There are pre-amplifiers available which allow you to select an appropriate turnover (and roll-off) frequency depending on the brand of the record that you are transferring. Alternatively, you can use the Virtual Phono Preamplifier (VPP) to realize the proper EQ curve. The turnover frequency for most electrically recorded 78-rpm records generally falls in the 200 Hz to 500 Hz area. If this is not accounted for, your transfers will sound "thin" on the bottom end. In addition to turnover, LPs used some pre-emphasis of the frequencies above around 5 kHz. If a Roll-off is not utilized during playback, these records will sound harsh on the top end. The industry standard equalization that encompasses both a turnover and a roll-off characteristic is the RIAA curve. It was used almost exclusively on all LP recordings after 1955. Other equalization curves for LPs that were employed before 1955 include the NAB, AES, FFRR, and the Columbia contours.

- Although you may be transferring monophonic 78-rpm recordings to your hard drive, consider making the transfers in stereo mode. By capturing both groove walls of the recording, you can take advantage of some of the information captured in this way in order to reduce noise, and clean up the muddiness often found in the "bottom end" of old 78s. Use the file conversion filter to convert from stereo to mono L+R for laterals and stereo to mono L-R for vertical cuts.
- Make sure that the turntable is sitting on a surface having very little vibration coupling in from other mechanical sources. The best place to locate your turntable would be on its own table sitting on your concrete basement floor, or better yet, on the floor itself. When you make the transfer, turn off your sub-woofers and keep the speaker volume very low in order to minimize acoustical feedback (and the resulting resonance) from appearing on the digital recording. Remember, after the transfer, you can play it as loud as you prefer without affecting the sound quality. One of the great benefits of digital audio CDs as a media compared to analog discs is the lack of an acoustical feedback pathway having a direct effect on the

sound quality. Acoustical feedback is not a first order effect in the playback of digital recordings, but it is with mechanical analog records. However, we would not recommend placing your CD player directly on top of your loudspeakers, as this can cause errors or skips to occur at very loud listening levels.

- A very simple noise reduction technique involves conversion of the file to a mono file by adding the two files together and dividing the amplitude in one-half. This tends to cancel out rumble and low frequency noise and is easily accomplished via the file conversion filter.
- When transferring Hill and Dale (Vertically Cut) records such as Cylinder recordings and Edison Diamond Discs, no useful information can be extracted from a stereo representation, so save the disc space and transfer these monophonically. However, your pre-amplifier must be placed in subtractive mode in order to derive a decent signal for the transfer (left channel minus right channel). Since not all pre-amplifiers have this function, you may have to transfer to DCArt10/DC Forensics10 in stereo, and perform a stereo to mono L/R file conversion in software. Another method for obtaining the left minus right signal for the transfer of Hill and Dale recordings involves a slight modification to the tone-arm shell of your turntable. You can re-wire the stereophonic phono cartridge output terminal to be in series. The phasing must be arranged such that the two positive (hot) signals are wired together forming a node, and the actual output should be taken from the two negative (ground) terminals of the cartridge coils. The tone-arm shell should be connected to only one of the two cartridge negative terminals. Sometimes one of the two negative terminals is connected to a metal shield around the cartridge. This is the terminal that should be connected to the shielded conductor of one of the tone arm co-axial cables. It is important to note that when a stereo cartridge is connected in this manner, the output impedance will double and therefore the manufacturers recommended value of load resistance should also be doubled.
- If you transfer acoustically mastered recordings through a magnetic cartridge that is driving an equalized pre-amplifier, you will have to apply the “Reverse RIAA” curve found under the Paragraphic Equalizer. If you omit this step, you will

overly amplify the bottom end of the surface noise spectrum of the record. It will do very little to enhance the bass response of the transfer, and will cause the transfer to sound "muddy." When using a magnetic cartridge to transfer such records, the pre-amplifier equalization circuits must be disabled. If this is not easily accomplished, try using a Mic input for the magnetic cartridge, rather than the equalized magnetic cartridge input, but it should be noted that moving magnet magnetic phono cartridge usually requires a 47,000 Ohm termination resistance.

- The Mic input on a preamp is usually fairly high in gain, and flat in frequency response. However, if the input impedance is higher than the recommended load resistance for your phono cartridge it will be necessary to terminate the input to the Mic pre-amplifier with an additional parallel connected resistor. Use the following equation to determine the value for this input shunt resistor (this equation assumes that the input impedance specification for your input is actually represented by a simple resistance in the audio frequency band, rather than a complex impedance, which is generally the case):

Note: If the input impedance of your microphone pre-amp is less than the recommended cartridge load impedance, this technique will not work.

$1 / R_{shunt} = 1 / R_L - 1 / R_{in}$

whereby - - -

$1 / R_{shunt}$ = This is the reciprocal value of resistance that you must add in parallel with the input of your system, given in Ohms.

&

$1 / R_L$ = The reciprocal value of required load resistance specified by the manufacturer of your magnetic phono cartridge, given in Ohms.

&

$1 / R_{in}$ = The reciprocal value of input resistance for the existing microphone input which you will be using, given in Ohms.

- Also, something worth considering (but not as important as load resistance matching) is providing the proper value of load capacitance for your phono cartridge. If capacitance loading is incorrect, ticks and pops on the record may produce a ringing waveform that could make it more difficult for the Impulse Noise Filter detector to discriminate between sound transients and noise impulses. When analyzing the value of load capacitance, consider the cable capacitance as part of the total. For example, if the cartridge manufacturer recommends a 250 pF load capacitance value, and the cables already have 150 pF (6 feet of cable at 25 pF per foot), then you only need to add 100 pF across the load resistor at the amplifier input.

- Avoid the use of special effects such as reverberation, graphical equalization, notch filtering and so forth at this point in the sound restoration process. The added complexity of these signals will make your job of sound restoration more difficult, because DCArt10/DC Forensics10 will have a tougher time of separating the important signals from transients and noise. These special effects, if desired, should be added after the signal restoration process has been completed. And lastly, do not cut off the top end of the signal bandwidth at this early stage in the restoration process. Some of the algorithms make use of the fast rise times of the transient signals in order to perform their function. The use of analog Low-pass filters at this stage of the process can create severe distortions of the ticks and pops on the source material. The "ringing" created by these filters can make it more difficult for the Impulse filter(s) to perform their function. It is always easier to remove bandwidth from a signal source than it is to restore it, so this job is best left to the point in time where you are actually performing the sound restoration with the DCArt10/DC Forensics10 tool-set.

Removing a Lead Vocal from a Stereo Recording

The following procedure will very often be effective in removing a lead vocalist's performance from a stereophonic recording. This is useful when you desire to over-dub your own rendition of the performance onto the original orchestration. If there is a lot of ambient sound (reverberation) associated with the original artist, it may not be removed along with the vocal itself. This is seldom a problem, since the ambient sound is not very distinct, and will not drastically deter from your own over-dub performance. It is important to note that this technique will not work with a monophonic source. It is also important to note that this technique very much relies on the type of material you are using. It will produce better results on more acoustical types music and sometimes no success on music with heavy walls of instruments.

1. Open your desired stereo file.
2. Open the Channel Blender feature (Effects Menu).
3. Set Left and Right to 100% blend
4. Select Invert Phase on the left channel
5. Select Blend to Mono and choose “Below”.
6. Set the Blend to Mono adjustment to 160 Hz.
7. Click on the Preview button.
8. You should hear the Source .wav file with an attenuated lead vocalist, with perhaps a little bit of reverberation (echo) in the background.
9. Adjust one of the two-slider controls downwards first. Observe if the lead vocalist gets louder or softer. If it gets louder, readjust the control upwards, looking for a point on the slider control wherein the lead vocal is nulled out (minimized).
10. Click on "Run Filter."

Note: This technique works using the same methods as hardware boxes costing many times the price of this product. However, it has the same limitations as many of these hardware solutions in that the vocals to be removed must have been mixed in the exact center of the stereo field by the recording engineer. Many recordings are made this way, but some are not.

Restoring an Old 78 RPM Recording: In Depth

The following are the steps in the process that we have been using for the restoration of early disc and cylinder recordings. You may find it useful to learn from our experience before undertaking an audio restoration of an old phonograph recording on your own. Please note that although most of these steps are based on 78s, the principles can usually be applied to all types of record restoration projects, including LPs and 45s.

1. Backup First

When dealing with a one-of-a-kind or a very rare recording, it is strongly advised that you make a transfer of the recording before beginning any cleaning process. The reason for this is so that you at least have something to work with in the event that you inadvertently damage the recording in the cleaning process.

2. Clean the surface of the recording:

Use a machine designed for this purpose if you have one available. The type that deposits a "bead" of distilled water and then removes it with a stylus, string, and a vacuum system are probably the best for this purpose (like a Keith Monks RCM system or similar). If a system such as this is not available, clean your record with a lint-free cloth and distilled water. Avoid the use of solvents or wetting agents which are non-aromatic* as they have the propensity to leave behind a residue. These residues can attract particulate matter over time and clog the bottom of the record groove. If you are cleaning either wax cylinders or Edison Diamond Discs, or other records containing wood or paper cores, do not use water because of the potential for damage by the solvent. Use only a lint free cloth on these items. Also, be very careful not to get fingerprints on wax cylinders. The oil in your fingerprint will provide the "seed" necessary to trigger fungus growth on the wax surface. This will destroy the cylinder groove wall in time. Blue Amberol cylinders can be cleaned with a cloth that has been moistened with distilled water. However, be careful not to allow any water to come into contact with the plaster core, because it may swell up cracking the record surface.

Important Note:

Do not use solvents such as alcohol or acetone on acetate (transcription) recordings! These solvents will destroy the recording. We have seen grown men cry after utilizing this method of cleaning on acetates (and in one case the transcription was a one-of-a-kind recording of a once famous German opera star).

***Note 1:** Aromatic solvents such as ethanol, methanol, or isopropyl alcohol diluted with distilled water are sometimes used to clean vinyl recordings. Never use benzene, gasoline, naphthalene or any other similar hydrocarbon based solvents to clean the surface of any type of recording!

<u>Note 2:</u> Acetate recordings are often covered with a white coating that appears as a powdery substance on their surface. This material (hexadecanoic acid) is not soluble in H_2O. It is suggested that records with this problem be played "wet" using distilled H_2O for the best transfer. Do not attempt to use solvents to remove the acid. You will need to play it several times. The stylus will clean out the groove. Clean the stylus between plays. Record each play of the record, even if it is a poor and non-final transfer.

3. Play the record in a "dry run"

Play the record once on your turntable using one of your smaller tip styli (approx. 2.3 mils). Obviously, you do not need to be able to listen to the recording during this process, and so you may want to consider using a separate turntable that you will use exclusively for this purpose that is not necessarily connected into your sound reproduction system. The purpose for this step is to kick up any accumulated dirt located at the bottom of the groove before re-cleaning the record surface.

4. Clean the Record Surface Once Again

Clean the surface once again using the procedure outlined in step #2.

5. Set your Pre-amplifier to the Proper Mode

Set your pre-amp to stereo mode for lateral cut 78 RPM records or use a flat Diamond Cut CTP-xxxx preamplifier. You are going to digitize in stereo not because your 78 record was recorded in stereo, but the left and right groove walls will both be recorded independently utilizing this technique. Later, using some of the file conversion options available in DCArt10/DC Forensics10, you will make some decisions regarding the best way to use this left and right groove wall information. If you are transferring Vertical Cuts (Hill and Dales) like Edison Diamond Discs or cylinder records, you will either have to place your pre-amplifier in Stereo (Left - Right) mode (which may be difficult since not all pre-amplifiers have this feature) or you will have to record in stereo, and use a File Conversion feature to convert the signal to its vertical component using DCArt10/DC Forensics10. If your pre-amplifier does not have the feature, the record will sound a bit noisy and weak as you listen to it. However, often when the file conversion is performed via your software, the gain will increase and the signal-to-noise ratio will improve at that point in the restoration process. Make sure that your pre-amplifier is providing the full audio bandwidth to the digital recorder. Use no filters at this point in the restoration process. The DCArt10/DC Forensics10 program will need as much bandwidth later to perform its magic. Do not concern yourself with the theory of producing aliasing as some have suggested, since that problem has been solved by all reputable sound card or digital recorder manufacturers.

As mentioned, a flat pre-amplifier can be used such as the Diamond Cut CTP series. The Diamond Cut software Virtual Phono Preamplifier (VPP) can impart the correct equalization curve onto the recording after the transfer has been made, if that step is necessary. EQ is only required for electrically recorded records. Acoustically recorded records do not require any equalization. If an acoustically recorded 78 was transferred via an RIAA preamp, that incorrect curve can be reversed via the Diamond Cut VPP.

If you are using a RIAA preamplifier to transfer electrical 78s, you can use the Diamond Cut Virtual Pre Amp to reverse the RIAA curve and impart the correct Turnover curve to your transfer. Please refer to the section of this users guide pertaining to the Virtual Preamp (VPP) for details.

6. Verify that your Turntable Speed is Correct

Verify that the speed of your turntable is correct. Your Diamond Cut software includes a set of Strobe discs. Printable Stroboscope Disc Metafiles can be found in the install directory for your Diamond Cut software. The 50 Hz strobe disc is called "**Strobe50Hz.wmf**" and the 60 Hz version is called "**Strobe60Hz.wmf**". These strobes are found in the following directory:

<Users Documents>/DCForensics10/Strobe50Hz.wmf
or

<Users Documents>/DCForensics10/Strobe60Hz.wmf

Just print the desired disc and cut it out to be placed on your turntables platter. A fluorescent light will be needed to illuminate the disc properly. Most Victor, Columbia, and other 78s were actually recorded at 78.26 RPM, however, Edison laterals were recorded at 78.8 RPM. Edison Diamond Discs were recorded at 80 RPM. Use a strobe disc and a fluorescent or neon lamp operating from of your AC line for this purpose, provided that you live in an area where the power line frequency is accurate and stable. If the speed is incorrect, use the turntable pitch control to correct the anomaly. If you do not have a pitch control on your turntable, no need to worry. You can correct the speed later using the Diamond Cut Speed Change effect.

Note whether or not the record groove is rotating concentrically. If it is not, there will be a "Wow" effect on the recording transfer. This problem can be corrected if your turntable has a removable spindle. With the spindle removed, adjust the position of the record with respect to the turntable platter until the stylus tracks the record concentrically. If your turntable does not have a removable spindle, AND the record is not a priceless artifact, you might consider using a ream to enlarge its centering hole. Then, with the stylus playing the record, bump the edge of the record lightly with your finger until your tone-arm is operating concentrically. We must repeat - - - DO NOT use the ream method on an important artifact or a record that you care about!

7. Verify that you are Utilizing the Correct Equalization Curve

Verify that you are utilizing the correct equalization curve for the particular type and brand of record that you are about to transfer. Turnover and sometimes Roll-off are critical breakpoint frequencies that must be matched in a complementary manner to the recording process in order to preserve the "flat" response of the original recording session. Turnover frequencies for electrical recordings are between 200 to 629 Hertz. Acoustical recordings should always be transferred "flat" and "electricals" should be transferred with equalization that is the correct inverse of the recording equalization that was used in the mastering process. Also, it is important to have a system that has the ability to adjust the turnover (and roll-off frequencies for early vinyl) or to use the Diamond Cut Virtual Pre Amp to apply the correct EQ. Again, we recommend the use of a flat transfer via a flat phono preamplifier and to perform the EQ process via your Diamond Cut software. More esoteric EQs can be found in the preset listing under the Virtual Phono Preamplifier or the Paragraphic EQ filter. For more information on this topic, refer to the section entitled "Record Transfer to Hard Drive Technical Hints". Below is a list of common Turnover Frequencies for some of the more common brands of lateral cut 78-RPM records:

Type, Brand, or Process	**Turnover Frequency**
Acoustical Recordings	0 Hz
Columbia (1925 - 1937)	200 Hz
Victor (1925 - 1937)	200 Hz
Westrex	200 Hz
Decca (1935 - 1949)	250 Hz
EMI	250 Hz
English Columbia	250 Hz
HMV (1931)	250 Hz
EMI (1931)	250 Hz
London	250 Hz
Blumlein	250 Hz
Columbia (1938 – End)	300 Hz
BSI	350 Hz
Capitol	400 Hz
Mercury	400 Hz
Brunswick	500 Hz
Decca (1925 – 1929)	500 Hz
Edison Laterals (1929)	500 Hz
MGM	500 Hz
Parlophone	500 Hz
Victor (1938 – 1952)	500 Hz
“629 Curve”	629 Hz

8. Choose the Best Stylus

Choose the best stylus for the record transfer. In most cases, truncated elliptical styli are the best for transferring olde 78s, since a truncated stylus tip will not be in contact with any dirt and grit at the bottom of the record groove. The criteria for stylus selection should be based on two parameters. The first is signal-to-noise ratio (in more common terms it would be referred to as surface noise). The second is distortion. Keep in mind that it is always possible to improve the effective signal-to-noise ratio of a recording, but it is much more difficult to decrease the harmonic distortion content of a recording in any post-transfer process.

So find the stylus that produces the lowest distortion as the first criteria, and listen for the stylus that produces the best signal-to-noise ratio as secondary criteria. Styli are available in different diameters specified in mils, and in geometries such as spherical, conical, and truncated versions of both. Although there are charts which call out the best stylus to use for a particular record brand and era, you will probably find that the best one is always determined by trial and error, since the charts have no way to account for the wear pattern of the particular disc which you desire to transfer. However, if you prefer a more deterministic approach, you can use the following stylus list to get into the right ballpark:

1. **Edison 80-RPM Diamond Discs:** 3.7 mil spherical or non-truncated conical stylus.
2. **Wide Groove Acoustical 78 RPM Lateral Discs:** 3.8 mil truncated elliptical stylus.
3. **Edison White Wax, Brown Wax, Concert, and Gold Molded:** 7.4 mil Spherical stylus.
4. **Edison Blue Amberol Cylinders:** 3.7 to 4.2 mil non-truncated spherical stylus.
5. **Edison Wax Amberol Cylinders:** 4.2 mil spherical stylus.
6. **Pre-1935 Lateral Cut Electrical 78s:** 3.3 mil truncated elliptical stylus.
7. **Transcription Recordings:** 2.3 mil truncated elliptical stylus.
8. **Late 1930s Lateral 78 RPM Discs:** 2.8 mil truncated elliptical stylus.
9. **Narrow Groove 78s such as Polydor:** 2.4 mil truncated elliptical stylus.

10. **Standard Groove 78 RPM Discs:** 3.0 mil truncated elliptical stylus.
11. **Modern LPs:** 0.7 mil elliptical stylus.
12. **Early LPs:** 1.5 mil truncated elliptical stylus.
13. **1931 to 1935 RCA Pre-Grooved Home Recordings:** 5.0 mil spherical stylus (preferably truncated type).
14. **Pathé 78s:** 3.7 mil truncated conical stylus.
15. **Aluminum Instantaneous Discs:** 6.0 mil conical.
16. **Etched Label Pathé to 14 inches in diameter:** 8.0 mil conical.
17. **Etched Label Pathé over 14 inches in diameter:** 16 mil conical.
18. **CD-4 LPs (Quadraphonic LPs):** Shibata 0.2 mil, line contact (for 30 kHz subcarrier response).

Important Note:
Actual Groove widths can be measured with a 200X to 300X microscope equipped with a calibrated reticle. The location of the wear pattern can also be observed, so that you can choose a stylus having a dimension that will either ride above or below the groove-wall wear zone.

9. Fix Record Tracking Problems

If the record is severely warped and is having trouble tracking, try the coin trick that your grandparents used. Neutralizing the counterweight of the tone-arm, and applying some mass in the form of coins on the tone arm shell works wonders for warped, skipping discs. There is a "physics" basis for this technique. Ask your grandparents, and they will explain it to you. If they are not available, the technique involves the "second moment of inertia" as the culprit. The rear counterweight has little variable effect on this parameter. So, placing coins on the cartridge shell dramatically decreases the time constant of the second moment of inertia, allowing the tone-arm system to better track warped records without launching the tone arm into deep space. Pennies are used for mild cases, dimes are used for slightly tougher cases, nickels for more serious cases, and quarters are used for basket cases! Although it is best not to bottom out your stylus, do not worry too much if a basket case causes the pickup cartridge to do that. Bottomed-out cartridges will produce severe thumps in your stereo reproduction system. Most of these thumping artifacts can be removed later with the DCArt10/DC Forensics10 tool-set.

10. If your record skips, try Fractional Speed Re-Mastering

If your record still skips, try using this fractional-speed re-mastering technique. To perform fractional-speed re-mastering, you will need a turntable with wide-ranging speed variability. Set the turntable down from 78 RPM to 45 RPM. Then, use the Change Speed filter to correct the pitch. If you transfer the record onto your hard drive with the turntable running at 45 RPM, then, using the Change Speed filter, correct the pitch according to the following list:

1. 78.2 RPM record - Use +73.7% pitch increase (flat line contour)
2. 78.8 RPM record - Use +75.1% pitch increase (flat line contour)
3. 80 RPM record - Use +77.7% pitch increase (flat line contour)

When this procedure is utilized, equalization gets a bit messy. For example, if the correct turnover frequency was supposed to be 500 Hz, the setting on your pre-amplifier must be re-adjusted to 250 Hz to compensate for the fact that you are reproducing the disc at half its intended speed. If you are not using exactly half speed as your re-mastering speed, then you must use ratio-proportioning to determine the correct turnover frequency setting for your pre-amplifier. This is another argument in favor of flat transfers via a flat phono preamp. That way, EQ is not an issue since you will apply the correct Turnover curve after the speed has been corrected and it will fall onto the proper area of the audio spectrum. "Coin therapy" and/or half speed mastering should solve most warped record transfer problems. If these do not work, consider these two alternatives. The first involves utilizing a microscope to view the stubborn portions of the record. Look at the problem groove under a microscope with 50x, and with a very sharp tipped hobby knife; clear a pathway in the groove for the stylus to follow during reproduction. Minimize, but do not become overly concerned with groove damage because the software program can compensate (via interpolation) for this defect to a large degree later in the restoration process. Remember that if you are dealing with an acoustical recording, equalization speed compensation will not matter because you should be transferring with a flat pre-amplifier response, no matter what the value of re-mastering speed that you use.

* Warning! Listening to the Blues half-speed can be very, very, d e p r e s s i n g .

11. Adjust the Gain and Balance

Adjust the gain and balance of the audio input signal feeding your sound card. Play a portion of the song which you believe has the loudest crescendo or passage, and make sure that the system does not overload on any of the musical transients as indicated on the Level Meters. On the other hand, make sure that the gain is not so low that you are recording "in the mud". If you do, you will lose signal resolution and the signal-to-noise ratio will not be optimal.

<u>Important Note:</u>

Occasional overloading due to severe clicks and/or pops is allowable and will not adversely affect the sound restoration process. However, it is imperative to be sure that any overloading is due to noise transients, and not due to musical transients such as drum "rim shots", etc.

12. Choose the Appropriate File Conversion Technique

Choose the File Conversion that is most appropriate for your .wav file, if necessary. In a few cases this procedure is not necessary. For example, if you are starting with a monophonic .wav file, there is no need for a file conversion. Also, if you are starting with a stereo .wav file and intend to maintain it as a stereo .wav file, no file conversion will be necessary. However, in many cases, you will be dealing with a stereo-recorded .wav file of a monophonic source that must be converted to the cleanest format for further processing. This is the case with most 78-RPM laterals and some vertical (hill and dale) transfers wherein the A-B function was unable to be performed by your pre-amplifier. The method for determining the best file conversion for a 78-RPM lateral transfer is something you will have to subjectively judge for yourself. Use the preview function in the File Conversion feature found under "Filter". This will allow you to quickly hear the results of your selection.

First, try to listen to the material in stereo. This will be your baseline for comparison (reference). Next, listen to a file conversion from Stereo to Mono (Left Only). Compare the results of this audition to a file conversion to Mono (Right Only). This is essentially allowing you to compare the effects of the wear on the inner versus the outer groove wall of your recording. In many cases, these two will sound the same. However, in some cases with extremely worn recordings, one groove wall will sound much cleaner compared to the other. If this is the case, make a note of which of the two sounded cleaner (containing the least distortion).

The next comparison will be between the best of the two groove walls and the Mono (L+R) file conversion. In most cases, Mono (L+R) will be the cleanest version of all of the alternatives. This is due to the cancellation effect of the record vertical displacement component that contains record and turntable rumble when using this file conversion.

So listen carefully to the "bottom end" (bass) differences between the various conversion alternatives. Also, very often, some of the clicks and pops will diminish in intensity with the Mono (L+R) conversion. If you have transferred a vertical recording using a stereo cartridge, the only file conversion that generally makes any sense is the Mono (L-R) feature. This rejects the entire lateral component of noise signal from the transfer, preserving only the important vertical vector. After these decisions have been made, create a new file using the appropriate file conversion algorithm to be used in the next step in the restoration process. It is worth noting that sometimes the most effective de-clicking of the recording occurs when done before making a file conversion.
<u>Important Note:</u> When using the Stereo to Mono file conversion, it is imperative to verify that both gain controls are set to -6 dB in order to assure that you do not overload the Destination channel input. -6 dB is the DCArt10/DC Forensics10 default setting for this feature.

13. Filter out Residual Rumble
Filter out any residual "rumble" left on your Source File, by implementing the High-pass Filter. Cutting all frequencies below values in the range of 30 to 50 Hz with an 18 dB / Octave or 24 dB / Octave slope can be useful for this purpose. The Diamond Cut Virtual Phono Preamp (VPP) includes a 30 Hz Rumble filter feature making

this process simple. If you are dealing with acoustical material, you will probably want to set the frequency of the filter even higher to something in the range of around 100 Hz to 120 Hz since very little information generally exists below that range of frequencies. For this, you will have to use the High Pass Filter because the VPP rumble filter is fixed at 30 Hz. Don't be afraid to experiment and use "Preview" mode to help choose the best value for your material.

14. De-click using one or more of the Impulse Filter(s)
First, remove Big Clicks due to cracks in the record with the Big Click Filter. Next proceed to De-click the record using the EZ Impulse Filter (or, in extreme cases, the Expert Impulse Filter). Usually, the EZ Impulse filter will perform this function just fine. If your record is very "hissy" (lots of top-end noise as will be found in many early 78s) then it is sometimes useful to run a low-pass filter first set for around 15 kHz and 6 dB/Octave. The reason for this is that sometimes a lot of "hiss" can fool the algorithm into thinking that there are a lot of sibilant sounds on the recording, which will move the algorithm's threshold too far up to capture ticks and pops effectively. This filter should never be run at a frequency lower than 12 kHz at no steeper than 6db/Octave. (This rule <u>never</u> applies to vinyl LPs or 45s.) See the section on Low-Pass Filtering for details on its operation. If your record is not extremely "hissy", it is directly ready to be de-clicked by an Impulse Filter. If you are using the EZ Impulse Filter, start with all three control settings at 50 and adjust as necessary. If you are using the Expert Impulse Filter, for 78-rpm records, a good initial setting for the "Size" control is 5. Make sure that the filter is not in "Vinyl LP" mode. Set the Tracking adjustment all the way down (to a setting of 1) and perform all of your adjustments with the threshold control. Adjust the threshold downwards until the clicks are removed, but not so far down as to cause the process to distort the sound signal. Remember to utilize the preview feature to establish the best settings for your particular restoration job. Only in rare circumstances, as may be found on a very "high fidelity" 78 rpm record, will you have to employ the services provided by the tracking control, to move filter threshold out of the way on high-frequency musical transients. The use of the to the Expert Impulse Filter should be a last resort since the EZ Impulse Filter usually does a very effective job and is much easier to adjust.

Sometimes, two lightly applied applications of any of the Impulse filters will be more effective than one heavy application. The Multifilter can help with this process since multiple Impulse filters can be placed in tandem with that tool.

After running the Impulse Filters, listen to the entire recording and drop markers anywhere you find any clicks, pops, or thumps that the Impulse Filter did not eliminate. Try selective De-clicking utilizing "Sync Mode" (found in the "View Menu") and the Impulse filter or use one of the manual de-clicking processes (like the manual interpolation "I" key) to eliminate these. You can choose between the manual de-clicking process using copy and paste over, or you can use a much simpler process utilizing the interpolate feature. In extremely rare cases, it may be necessary to use the "Cut" feature when the anomaly is extremely long in time duration. Sometimes, the use of the Oterpolator (O) key will become a valuable tool to consider in some short impulse event cases.

Some of these techniques may not re-insert a signal as close to the original signal as does the Impulse Filter, so manual de-clicking should only be performed as a last resort (except for the use of the "I" key or the DSS). If there are a high number of impulse events that were not eliminated by the Impulse Filter, you should consider re-adjusting the filter's parameters and/or running it again before attempting to manually de-click the remaining noise artifacts. Multiple passes through the impulse filter with different settings for each pass is often useful.

When choosing a manual de-clicking technique, consider the following tradeoffs:

- The "Copy and Paste Over" method provides some attempt to re-insert a signal in the location of the transient, but it takes more operations to perform a single de-click with this method.
- The "Mute" method requires less steps to perform its function, but it replaces an impulse event with silence. This is OK when the clicks are very short in time duration. Your ear will integrate out the silence during the muted sector of the .wav file. However, longer events, when de-clicked with the Mute function, will

produce some inter-modulation distortion that may be noticeable.

- The “Interpolate” key is your best alternative in terms of ease of use and accuracy, so long as the problem is limited to a relatively short interval of time. The program will prompt you if you have chosen too large an area to be interpolated.

Note: Consider using the highlighter coupled with the “Snap Selection to Zero Crossing” (right-mouse) feature before manually correcting an impulse noise event in order to achieve the best manual interpolations.

15. De-Crackle the Recording

Many 78s have clean surfaces, and do not require this step in the restoration process. However, some very worn records, or records that were stamped utilizing low quality resins and fillers benefit from this step. So use the Narrow Crackle or the Median Filter to reduce "crackle" on recordings that need it. Crackle is essentially low amplitude clicks and which occur much more frequently than standard impulses. (It sort of sounds like “Rice Krispies”.) It is important to perform this step only after the initial "de-clicking" step has been completed, and before the de-hissing step is performed, if the best results are to be obtained. Usually, “Size” (in the case of the Narrow Crackle Filter) or "Sample" (in the case of the Median Filter) settings from 3 to 6 produce the best results. Using too many samples will create a distortion (sort of an "edge"), which is particularly pronounced on vocals. But experimentation is the only way to determine the most appropriate “Size” or “Samples”, since material varies widely in crackle content. Use the "Preview" feature to aid in the selection of the most appropriate setting.

Important Note:

Very early cylinder recordings may benefit from the application of the "Average" filter for de-crackling. Start with a small number of "Samples" and work your way up until you have achieved a good result. If you are not sure whether to use the Median or the Average filter, try both and decide which one works the best based on your own specific observations.

16. De-hiss the Recording

De-Hiss the recording using one of two methods. Even if you have a relatively quiet record, there will still be some residual hiss that you may wish to reduce. There are two alternatives to choose between. The Continuous Noise Filter (CNF) is the most effective at eliminating wide band noise all the way down to the bottom-end of the audio spectrum. Consider using the Artifact Suppression or Smoothing mode when performing this step. For details, please refer to the Filter section describing the Continuous Noise Filter.

However, the CNF has the possibility of introducing some digital artifacts onto your .wav file if it is not set correctly, particularly with high settings of the attenuator control. Start with a setting around 10 dB and try moving the control both directions. If you set this control too high, it will also reduce some of the "ambiance" of the recording with the tradeoff of providing a great deal of noise reduction.

Your other option is to use the Dynamic Noise Reduction filter. It will not produce as much noise reduction as the Continuous Noise filter, but it is easier to set up without introducing digital artifacts onto your Destination .wav file. So the choice will have a lot to do with the condition of your Source File, and your own taste with regard to the tradeoffs that were just outlined. Trial and error is a good method for sorting this one out.

17. Eliminate Line Frequency Hum and/or Buzz

Eliminate 50 or 60 Hz hum and their harmonics from your recording utilizing the Notch or Harmonic Reject filter. When "notching" out hum, be sure to set the bandwidth control to the smallest value that is effective in order to minimize any effects on adjacent frequency information. Similarly, when attenuating a “Buzz” with the Harmonic Reject Filter, minimize the number of harmonics required to do the job at hand. Some early recording had some hum induced by the amplification system used to cut the master. Two frequencies will sometimes be heard. One is often the line frequency fundamental of either 60 Hz on American made records, and the other is 50 Hz on European makes. Another frequency sometimes heard is at twice the line frequency and is usually due to faulty filter components in the full-wave rectifier dc power supply section of the master cutting-head amplifier. If you hear any of these noises, they can be greatly attenuated with the Notch filter. This filter can be set to have a very narrow bandwidth, and so will have very little sonic effect on the adjacent frequencies. Remember, if your recording should contain "Buzz" rather than “Hum”, use the Harmonic reject filter instead. You will learn through experience to discern the difference based on the sound that these two anomalies produce. And, usually, when a recording contains a lot of “Buzz” that usually means that your record transfer setup has a technical problem and the “Buzz” is not on the master. For example, an un-grounded turntable will often induce “buzz” onto your transfer. Low quality light dimmers are a common source of line induced “buzz’ into an audio system. It is always best to clean up your record transfer setup rather than trying to fix this kind of problem with the software post facto.

18. Provide a Fade-In and a Fade-Out Sequence

Provide a smooth Fade-In and Fade-Out sequence using the features with the same name. DCArt10/DC Forensics10 allow you to choose between a linear or a logarithmic gain vs. time curve. Experiment to determine which curve is the best for the material you are dealing with. After you are done setting up your Fade-In and Fade-Out process, use the "Mute" function to eliminate any extraneous noises or signals that precede the Fade-In and succeed the Fade-Out sequence.

19. Add Your Own Personal Touch to the Transfer

DCArt10/DC Forensics10 provides you with a 10 & 20 band Graphic Equalizer and a 10 band Paragraphic Equalizer that you can use to create a more pleasing tonal balance to your audio restoration. It can be found under the Filter Menu. Additionally, the Virtual Phono Preamplifier provides you with three tone controls, two of which are of the shelving type. Alternatively, as you listen to the restored .wav file via the DC Tune Library, you can consider feeding the signal through various analog signal-processing devices via the Multifilter in real-time to give it your own personal flare at that point in time. The Diamond Cut software systems such as paragraphic equalizers allow you to bring out sounds that you consider to be too understated, and you can attenuate sounds that you may believe to be overemphasized

Some people like to add some ambiance to the transfer with a little bit of reverberation, which can be accomplished with the Reverb tool, or the echo effect. Devices and effects such as Spectral Enhancers and Harmonic exciters (found within the VVA effect) can provide interesting effects especially on vocals. If your recording is missing too much of the "top-end" you can enhance it with the Virtual Valve Amplifier exciter. It will synthesize harmonics of the upper registers of the musical scale, and allow them to be mixed back into your original source material. If your material is exceptionally noisy, consider using the “High Noise” mode when applying the VVA to old 78s. Another interesting Effect to consider is the Punch and Crunch multiband Dynamics processor. This can be used to restore some of the original dynamics which may have existed during the recording session. Lastly, you have a Sub-Harmonic Synthesizer and an Overtone Synthesizer that may be used to create pleasing effects (when used in moderation).

If you are performing a sound restoration for commercial release, it is important to provide a product that will sound good on most audio systems to most people. For example, if you have a very low quality reproduction system, you may be providing equalization, which is really compensating for the fact that your sound system has a poor frequency response, rather than compensating for a lack of certain frequencies on the recording itself. On the other hand, if your system is the state-of-the-art, you may want to listen to your results on a cheaper sound system as well, to be sure that the recording does not "break up" due to excessive bass. With any reproduction system (and sound room), it is a good idea to "flatten it out" before making subjective decisions regarding your final EQ. Use a pink-noise generator and a Real Time (spectrum) Analyzer (RTA) as the stimulus / response system for the measurement. (Your Diamond Cut Software Spectrum Analyzer is capable of making these measurements.) Connect a graphic equalizer (preferably 1/3rd Octave, 30 band) between the output of your reproduction system pre-amplifier and its power amplifier. Flatten out the response of the system / room utilizing the graphic equalizer, moving the measurement microphone to various locations in your listening area. Flatten out each channel independently. Lastly, when making your final EQ decisions, get someone else's opinion as well as your own (and include both sexes). Since audio "quality" is a highly debated and subjective area of discussion, some "opinion averaging" is in order when engaging in commercial releases. If you do not seek the opinions of others at this phase, you may find that your customers will volunteer it to you, and at a point where it will cost a fortune to fix! (If the reviews have already gone out, it may be too late to fix it at all.)

When you are done with the restoration process, you will have to transfer the contents from your computer hard drive and into some other format. The most obvious method simply involves the use of the D-A converter in your sound card. However, this method will produce a poorer transfer compared to direct digital, since it will introduce a small amount of distortion and noise (hum) onto the recording. If the audio restoration you are performing is for commercial release, or you are simply interested in obtaining the highest possible transfer quality back to another medium, you should consider using a digital-to-digital transfer. There will be no generational losses if this technique is used. Another alternative for achieving a loss-less transfer of your Destination .wav files would be to use a CD-R drive. These units are readily available, low in cost, and can be mounted directly in one of your computer bays. Alternatively, you can purchase one that is USB based.

Restoring a Recorded Telephone Conversation

Recorded telephone conversations sometimes are very difficult to decipher due to gain variations and/or the "out-of-spectrum noise", "in-spectrum-noise", poor frequency response and high levels of distortion. This is particularly true of sound transfers made from Digital Communications Recording Systems. The following five-step procedure is useful for restoring such recordings. It is recommended that these steps be performed in the order outlined below. The process is arranged in an order that will require only the poorest recordings to require all of the steps outlined. Some recordings will require only one or two steps to yield adequate intelligibility. Keep in mind that you are not trying to tune for high fidelity here; improved intelligibility of the conversation is the only goal of this procedure.

1. If there is a variation in the gain (loudness) of the recording depending on which party is talking, or where you are in the conversation, use the DCArt10/DC Forensics10 Gain Riding Procedure first in order to even out the levels or run the Dynamics Processor ALC feature.
2. Next, apply the Band-pass filter. This filter is used to remove "out-of-spectrum-noise" from the signal. Use one of the two

Speech Filter settings indicated below, choosing the one that is most effective for the particular material you are dealing with:

Standard Speech Filter:
- Low Frequency - 300 Hz
- High Frequency - 3,000 Hz
- Slope - 12 dB / Octave

Steep Slope Speech Filter:
- Low Frequency - 250 Hz
- High Frequency - 3,500 Hz
- Slope - 18 dB / Octave

If none of these setting improves the signal-to-noise ratio of the signal, experiment with your own values for the Band-pass filter.

3. To reduce "in-spectrum-noise" next apply the Continuous Noise Filter (CNF). Before running the filter, highlight a sector of the conversation containing a slight pause between the two parties talking for use as the "sampled noise" baseline for the Continuous noise filter threshold setting. This sector should contain only background noise. Also try the AFDF (Adaptive Frequency Domain Filter) found under the CNF Mode selector. Run the one that produces the best results.
4. If the signal remains "muffled" or "garbled", try applying the Median filter. Start with the Samples control set to around 9, and increase to as high as 19 or more until the consonant sounds and the sibilant sounds become more pronounced and intelligible. Also, consider running either the Auto-Voice filter or the Spectral Filter found under the Forensics Menu.
5. Lastly, try applying the Graphic Equalizer to improve the overall frequency response of the conversation.*

*__**Note:**__ If there still remains a large variation in loudness between the two speaking parties, apply the ALC feature found in the Punch & Crunch Effect.

Rumble Reduction

Rumble is a low frequency random noise found on record recordings. Generally, the noise in the spectrum below 30 Hz is considered to be “rumble”. However, with very old 78 and 80-RPM recordings, rumble can often be heard with higher frequency content (as high as 120 Hz). DCArt10/DC Forensics10 provides you with three possible methods for substantially reducing rumble on a recording. But first, it is important to perform your transfers with a low rumble turntable to begin with. There is no sense in making the problem worse by using a cheap turntable (especially the types which use a thrust bearing which is of the ball-bearing race variety). Turntables whose platters are thrust supported on a single ball (or point) will offer the lowest degree of turntable rumble.

Method #1:
If you are transferring a lateral 78, or a lateral monophonic LP or Monophonic 45 RPM record, transfer it to your hard drive via a stereo cartridge. Then, using the File Conversions feature, convert the file from "Stereo" to "Mono L+R." Since most of the rumble on these types of records are contained in the vertical vector only, the Mono L+R algorithm will cancel out this noise signal, and only preserve the lateral (horizontal) component, which does not contain much rumble.

<u>Important Note:</u>
Method #1 will not work on Hill and Dale or Pathe´ (groove width modulated) or stereophonic recordings.

Method #2:
Use the Continuous Noise Filter as prescribed in an earlier tutorial. It is only important to realize that the signals you are dealing with will be those at about 60 Hz on down. So when you experiment with the threshold line position, only adjust it in the lowest octave to eliminate rumble. When adjusted correctly, you should be able to preserve the last octave of the audio band in terms of actual signal, while rejecting the lower signal level rumble. Use high values of fft and move the threshold line down to the bottom above 200 Hz.

Note: This technique will work on all types of recordings, not just monophonic lateral recordings.

Method #3:
Use the High-pass Filter, with settings of 60 Hz on down, and the slope control set to 18 dB / Octave or 24 dB / Octave. Experiment with

different frequency values and/or slopes until you achieve the desired results using Preview mode.

Note: This technique will also work on all types of recordings, not just monophonic lateral recordings. However, it is not as good as method number 2 because it will also eliminate some of the low bass of your recording as well as its rumble content.

Method #4:
Use the Channel Blender Effect with the "Vinyl LP Cancellation Filter" or the "Vinyl LP Bass Clarifier" preset enabled. Often, this method produces the optimum rumble reduction results on Vinyl LP Stereo recordings or Stereo 45s.

Simulate Stereo from a Mono Source – Method #1

1. Open the Monophonic File to which you desire to add "Stereo Simulation."
2. Click on "File Conversions", found under the Filter Menu.
3. In the "Source File" box, click on "Mono".
4. In the "Destination File" box, click on "Stereo".
5. Click on "Run Filter".
6. When the conversion process is completed, close the File Conversions dialog box
7. Under the File Menu, click on "Make Destination the Source."
8. When the "Save As" dialog box appears, click on "OK" if you are satisfied with the automatically assigned temp file name.
9. Under the Effects Menu, click on "Reverb".
10. Using the "Preview" mode button, listen to the effect that the reverb is having on the file. Adjust the various reverb parameters until you achieve the desired stereo effect.
11. When you are satisfied with the reverb settings, click on "Run".

Simulate Stereo from a Mono Source – Method #2

1. Open the Monophonic File to which you desire to add "Stereo Simulation".

2. Click on “File Conversions” found under the Filter Menu.
3. In the “Source File” box, click on “Mono”.
4. In the “Destination File” box, click on “Stereo”.
5. Click on “Run Filter”.
6. When the conversion process is completed, close the File Conversions dialog box.
7. Under the File Menu, click on “Make Destination the Source”.
8. When the “Save As” dialog box appears, click on “OK” if you are satisfied with the automatically assigned temp file name.
9. Under the Filter/EQ Menu, open the Paragraphic EQ.
10. Activate the Left (L) channel by clicking on the “L” button on the Toolbar (on the top of the Diamond Cut software program).
11. Find the Paragraphic EQ preset called Stereo Simulator Left Channel Comb Filter* and then click on “Run” in the Paragraphic EQ dialog box.
12. Make the Destination the Source.
13. Activate the Right “R” channel by clicking on the “R” button on the Toolbar and then click on “Run” in the Paragraphic EQ dialog box.
14. Make the Destination the Source.
15. Activate the Right channel by clicking on the “R” button on the toolbar.
16. Find the Paragraphic EQ preset called Stereo Simulator Right Channel Comb Filter* and then click on “Run” in the Paragraphic EQ dialog box.
17. Close the Paragraphic EQ and open up the File Conversion Filter.
18. Click on the “LR” button on the Toolbar.
19. Set up the File Conversion Filter for “From Stereo to Stereo” mode.

20. Set the Time Offset for 10 mSec and Preview the filter. Adjust the Time offset while previewing until you obtain the stereo effect that you like and then Run the filter.

***Note 1:** Paragraphic EQ Presets with a (W) suffix will provide a stronger stereo effect.

Note 2: Sometimes, you may want to combine methods 1 and 2 for the strongest stereo effect.

Manual Splitting & Recombining a Stereo Wave file

Though we have recently implemented a new automated Split and Recombine tool, some users may prefer performing this process manually.

1. Bring the desired stereo file up into the Source Workspace.
2. Under the Filter menu, select the File Conversions feature.
3. Click on the "From Stereo to Stereo" function.
4. Lower the Left Amplitude control to -96 dB (control all the way down).
5. Run the Filter
6. Make the Destination the Source and then rename the file. (It may be a good idea to add an extension to the file name denoting that it is the right channel only.)
7. Double click on the Icon just to the left of the File Menu selector to return you to the window that contains the original stereo .wav file located in the Source Workspace.
8. Double click on the Source Workspace.
9. Lower the Right Amplitude control to -96 dB (control all the way down).
10. Run the Filter again.
11. Make the Destination the Source again renaming the file. (This time you may consider adding an extension to the file name denoting that it is the left channel only.)
12. Perform whatever independent signal processing you desire on each of the two files (channels). If only one channel requires processing, just process that particular .wav file.
13. When you are done with the independent channel processing, bring up the final version of the right channel into the source window.
14. Under the Edit menu, click on "Copy".
15. Bring the final form of the left channel into the source window.
16. Under the Edit menu, click on Paste and then Paste Mix.
17. Click on OK - the resultant file in the Source Workspace is the re-combined stereo .wav file.

Using DCArt10/DC Forensics10 as an Audio Waveform Analyzer

DCArt10/DC Forensics10 can be used as a waveform analyzer. It is capable of both time and frequency domain analysis. This waveform information may be useful to know when trying to determine what useful frequency range is present in a .wav file. It also will allow you to measure resonance or acoustic feedback, and periodic noise sources such as 50 or 60 Hz Hum (or harmonics thereof). Once this information is known, it will be easier to set up the parameters for the Notch, High-pass, Low-pass or Band-pass Filters, so that you can retain useful portions of the spectrum, while rejecting unwanted noise signals. This information is also useful in Forensic applications for measuring the time between gunshots or other events of critical importance. The waveforms, either in the time or frequency domain, can be printed out using the Print Screen procedure.

Method #1

Using the time scale in the workspace to determine a signal's frequency:

1. Zoom-In on the portion of the .wav file containing the suspect noise signal, to the extent that you can make out the peaks and valleys of the waveform of interest.
2. Note the total time duration of the "frame" which you are looking at. It is located on the bottom of the DCArt10/DC Forensics10 window in the status bar. (This time is the difference between the frame "end" time which is indicated in the upper right hand corner of the workspace and the frame "beginning" time that is indicated in the upper left hand corner of the workspace.) This number will be in fractions of a second.
3. You will notice that the x-axis of the workspace is divided into 10 grids. Calculate the time duration of one grid by dividing the total frame time duration by 10.
4. Count the number of cycles (cyclic peaks, or cyclic valleys), which occur within one of the ten x-axis grids. Divide the total time duration of one grid (as determined in step #3) by the number of cycles that you have counted. This is the value

of the time duration for one cycle of the waveform you are observing (T_{cycle}).

5. Take the inverse of that time in seconds. This is the value of the frequency of the waveform you are observing ($F = 1 / T_{cycle}$).

Method #2

Using the Band-pass Filter as an Audible Wave Analyzer:

Note: This method only works well if your computer is fast enough to run the Band-pass Filter algorithms in real time in Preview Mode. Otherwise, the system will "stutter" making it difficult to interpret the results. Also, it is important to realize the subjective nature of this method of analysis. Your own ability to hear frequencies at the extreme ends of the audio spectrum is critical. You need to consider that most men begin to lose their high frequency response after the age of around 21, and most women begin to lose their high frequency response after the age of around 30 (although there are always exceptions). You may want to engage the opinion of your kids (whose hearing is generally exceptional, believe it or not) when performing these subjective tests. You will also need a very high quality sound system for these measurements. It is very useful to have one that utilizes a sub-woofer for making the low-end measurements.

1. You will be using the Band-pass Filter in "Preview" Mode.
2. Select a Slope value of 18 dB / Octave.
3. Set the High Frequency slider control all the way up to its highest frequency.
4. Place the Low Frequency control to around 7 kHz.
5. Listen to the Material in Preview mode through a high quality audio system.
6. If you can make out any audio signals containing information and not just noise, increase the Low Frequency setting in 1 kHz increments.
7. Repeat step 5 and step 6 until all you hear is noise. This will be your upper cutoff frequency value for the material you are dealing with. (Be careful in this evaluation because some very subtle sibilant sounds add a great deal of character to an audio source. So make sure you are only hearing noise when you decide that the frequency control has been set appropriately.)
8. Next, set the Low Frequency slider control all the way down to its lowest setting.

9. Place the High Frequency control to around 100 Hz.
10. Listen to the Material in Preview mode through your sound system.
11. If you can make out any audio signals containing information and not just noise, decrease the High Frequency setting in 10 Hz decrements.
12. Repeat step 10 and 11 until all you hear is noise (rumble). This will be your lower cut-off frequency value for the material you are dealing with. (Be careful in this evaluation, because some very low level bursts of bass in sync with the music may be important in the overall sound character of the source.)

<u>Method #3</u>

Using the Audio Spectrum Analyzer

1. Using the mouse, highlight the sector of the Source .wav file that you desire to analyze. Often this will be a lead-in groove or lead-out groove of a record, in order to analyze the record background noise. However, it could be a sector of a .wav file in which acoustic feedback occurred or Hum was observed, and you desire to measure its frequency, so that it can later be attenuated.
2. Click on the Filter Menu and select Continuous Noise.
3. Click on "Sample Noise."
4. A status window will pop-up indicating the "% Done" as the system performs its calculations.
5. When the calculations have been completed, the status box will disappear, and a graph will appear.
6. The graph of interest is shown in red. The graph will plot the Amplitude expressed in dB versus the Frequency of the selected portion of the .wav file. (The blue graph is used for a different application, and is not applicable for this function.)

<u>Method #4</u>

1. Bring up any of the filters or effects.
2. Under the View Menu click on the Spectrum Analyzer.
3. Set the Spectrum Analyzer controls to the desired settings.
4. Run "Preview" for the particular filter, and the post-filter frequency domain signal shall be presented on the graphical display.

5. To "freeze" the display on a particular signal, click the enable button off.
6. When running a filter, you can leave the spectrum analyzer on, but it will consume some resources, slowing down the processing of the file to a small degree. So it is best to shut down the spectrum analyzer during processing.

Ogg Vorbis Lossy Compression Tag Support

Your Diamond Cut software can both play and encode files using the Ogg Vorbis format. You can also rip files from a CD into this format as well as play then via the DC Tune Library. It also supports the following Ogg Vorbis tags:

TITLE: This field is the Track/Work name and it is initialized with the name of the file being used.

ALBUM: This field is related to the collection name to which the track belongs.

TRACKNUMBER: This field contains the track number of the piece if it is a part of a specific larger collection or if it is part of an album.

ARTIST: This field contains the artist generally considered responsible for the work. In popular music, this is usually the performing band or singer. For classical music it would be the composer. For an audio book it would be the author of the original text.

GENRE: A short text indication of music or track genre.

FLAC Lossless Compression Tag Support

Your Diamond Cut software can play and encode files using the FLAC (.flac) lossless file compression format. FLAC produces a reduction in file size of roughly one-half without any loss of sound quality. The "encoding / compression" control actually affects the degree of compression that the system will achieve and not the actual audio quality of the resultant file. However, higher values of "compression" will take longer for your machine to create due to the increased complexity of the math involved, but the resultant file size will be smaller. The software supports the following FLAC tags:

TITLE: This field is the Track/Work name and it is initialized with the name of the file being used.

ARTIST: This field identifies the artist or songwriters who are involved with the creation of the file.

ALBUM: This field is related to the collection name to which the file belongs.

GENRE: This field allows you to specify a type of music relating to the file.

TRACKS: (Numerical Value) This provides you with a field to identify the number of tracks involved in the file.

YEAR: Use this to identify the year of the file creation or the year of the audio restoration or re-mastering date.

mp3 Lossy Compression Tag Support

Your Diamond Cut software can supports the following mp3 lossy compression file tags:

NAME: Use this as a description field for the mp3 file.

ARTIST: Use this to describe the author or artist associated with the file.

GENRE: This field allows you to specify a type of music relating to the file.

TRACK: (Numerical Value) This provides you with a field to identify the number of tracks involved in the file.

YEAR: Use this to identify the year of the file creation or the year of the audio restoration or re-mastering date.

Section 4 – Tech Support

Trouble Shooting

This is your one stop shop for problems or questions (FAQs) that you may encounter while working with our software. This list of problems and solutions constitutes about 90% of the overall problems our techs deal with. Chances are…if you have a problem with our product, the solution is listed below. Please read this before contacting us.

This section lists common questions and problems users have with DCArt10/DC Forensics10.

Q: Onto how many computers am I permitted to install the Diamond Cut software based on one license purchased?
A: The Diamond Cut software is licensed for use by one person on up to two computers (typically a desktop workstation and a laptop). One license of the software does not permit installation on two or more computers to be used by two or more separate persons.

Q: How do I control the recording level of the audio signal?
A: In Windows, there is a speaker Icon in the lower right hand corner of the Taskbar. Double-click on this Icon to bring up the control panel for your sound card. There are level controls for the Mic or Line inputs of the sound card. For Sound Blaster type cards, double click the yellow speaker icon and then choose Options/Properties. Click on Recording, and then click OK. You'll now have your input mixer up. Also be sure that the correct input is selected (Mic, Line, or Aux) for your particular recording setup**.**

Some sound cards have their own audio control panels that are better suited to controlling the audio inputs. These are usually available from the Start Menu

Q: I'm trying to record and I hear the audio, but the Record meters are not jumping.
A: Most sound cards come with the Line Input not selected. This causes a "no recording" symptom. But first, make sure that the sound card is installed correctly. Just go to the Control Panel/Multimedia and

make sure that the sound card is selected as the default playback and recording audio device. If it is, the sound card is most likely working okay. Next, bring up the Preference dialog box and select the "Sound Card" Tab and make sure that the inputs and outputs are set to the correct sound card. If the system still is not recording, double click on the yellow speaker icon on the tray and choose Options/Properties and click on the Recording radio button. Click OK. This brings up the input mixer. Make sure that the slider labeled "Line Input" is selected (it usually will not be). When you do this, the meters in the Record window will start modulating immediately, indicating that the system is now working fine.

Q: Can I run multiple instances of the software at the same time on my computer:
A: Yes. To bring up another instance of the software, minimize the first and double-click on the software icon on your desktop and the next instance will appear.

Q: I get a Windows error message when I try to record that says something about "this is beyond the capability of this device" or "The specified format is not supported". What's this?
A: This just means that your sound card is not capable of recording with the sample rate or bit width you have selected. For example, if you try to record a 24-bit file with a 16 bit sound card, you'll get this message. Most sound cards will work with 44.1 kHz Sampling rate, 16 bit, stereophonic recording settings. This could also mean that Windows is assigning .wav files to be played by some device other than a sound card…like a modem. Make sure Windows is aware that you want .wav files assigned to your sound card's output.

Q: I click on Play and don't hear anything but the VU meters are jumping fine in the software program.
A: If the cursor is moving across the screen during playback and the VU meters are jumping, you can be pretty sure the audio is indeed playing. Usually, this just means that you don't have your speakers plugged into the soundcard output where the audio is appearing. Check under Edit/Preferences/Soundcard to see what the output device is set to and make sure your speakers are plugged into that set of outputs.

Q: My levels are low during recording.

A: Maybe not. You should normally not record all the way up to 0db (the red area) with digital systems. Remember, you'll likely need some "head room" in order to be able to increase volume with some of our tools such as the Paragraphic EQ. Peak levels of -4 or –6 db will give great results. If your levels are REALLY low, check to make sure you have the right type of signal going into your sound card. You cannot plug a turntable directly into a soundcard, for example - you'll need a preamp for this.

Q: My sound recording is distorted and monophonic despite a stereo line level signal input. Why is that happening?
A: It is likely that you have plugged your line level device into the microphone input of your desktop computer or laptop. If your computer has a line level input jack, simply re-plug the cord into that input. However, some computers only have one input jack. In many cases, the mic input can be converted into a stereo line level input via the computer sound card control panel.

Q: Is the shielded cable capacitance or characteristic impedance critical in the connection of the output of a line-level audio device to the line level input of another audio device?
A: Most modern op-amp based line level audio equipment have very low output impedance values (around 100 Ohms) and high input impedance values of around 10,000 Ohms. Furthermore, the wavelength of audio signals are very long so that characteristic impedance of the interconnecting cable has no impact on circuit performance. So, for most typical audio interconnections, the 10 to 25 pF / foot that most shielded audio cable exhibit will have no degrading effect on the top-end of your audio signal, unless you are running very long distances between components (4000 feet or more) in which case other factors become dominant, like ground loop noise injection. If the goal is to achieve a 20 kHz response using an unbalanced audio transmission system with a corner frequency (-3 dB), one would be able to run a 4000 foot coaxial line driven by 100 Ohm source impedance to reach the audio spectrum limit of 20 kHz. Long distances like this, however, should always be run in a balanced configuration to avoid noise problems.

Q: Is it necessary to use balanced (XLR or TRS type) connections between audio components in my system?

A: If you are performing Audio work professionally, we recommend the use of balanced lines between audio components, especially if they are located long distances apart (greater than 20 feet). Otherwise, unbalanced RCA or 1/8 inch phone plugs are acceptable for most setups.

Q: I am using unbalanced RCA or 1/8 inch phone connections between my audio components and can hear a slight hum or buzz in the recorded transfer. What can I do to minimize this short of going to a balanced audio signal distribution system?
A: Plug all of your audio components into the same power outlet strip. This will minimize ground loop interference. Also, make sure that your audio system power line branch circuit is not also supporting phase controlled dimmer light fixtures. A dedicated line is advisable for your Digital Audio Workstation (DAW) for the lowest noise situation. Make sure that you do not overload the outlet strip. If the branch circuit and/or outlet strip is rated for 15 Amps, as an example, never load it greater than 80% of its rating (which is 12 Amps in this example). In some cases, it helps to ground all of the audio components together via a set of short and heavy 14 AWG (green or green with yellow stripe) stranded set of cables (especially including a turntable chassis). This reduces the amount of ground loop return current present in the unbalanced shielded cables thus reducing induced hum and buzz.

Q: DCArt10/DC Forensics10 contains a Fast Edit mode and the Classic Edit (two-file) mode. Which one should I use?
A: Beginners tend to like the Fast Edit mode since it works more like other audio editing programs. It's intuitive, simple and fast. However, as you learn the program, we think you'll find the classic two-screen mode to be the most powerful audio restoration environment. Our "Source and Destination" way of working is optimized for those situations where you want complete control of the files being edited, created, and saved. This is especially useful for forensic situations where an audit trail is required.

Q: I clicked Fast Edit in the Preference box and I'm still in the Classic mode.

A: You may need to close the program and restart it for this change to take effect. Alternatively, use the Fast Edit icon to toggle between fast edit and classic edit mode(s).

Q: I am getting crashes when I try to play audio. What do I do?
A: DCArt10/DC Forensics10 uses both the newer Windows Driver Model (WDM) method of communicating with sound cards as well as the older MME method. WDM is new to many sound card manufacturers and some of these new sound card drivers may have bugs in them. If you are getting a crash when you try to play or preview a filter, please check to see if a new sound card driver is available from the manufacturer of your card. If this doesn't do the trick, look under Edit/Preferences/Sound Cards and check the box which says "re-initialize on play". This makes the sound card return to a known state every time you play and may help you. You can also select between WDM and MME in this area to coincide with the most reliable support from your sound card.

Q: Why does Preview mode sound like it is stuttering?
A: All of the filters in DCArt10/DC Forensics10 require a fair amount of processing power. If your computer cannot complete the processing fast enough to keep up with the audio stream, then the preview mode will stop and start in short bursts that sound like stuttering. This effect can be reduced or eliminated by increasing the number of "Preview Buffers" in the Edit/Preferences/Soundcard dialog box. Regardless of stuttering in Preview, all "Run" filter operations will be stutter free. Putting many filters in the Multi-Filter box will likely cause stuttering even on very fast machines. You may also want to check and see that you're using your sound card's most recent drivers.

Q: I noticed that a Live Mode is now included in DCArt10/DC Forensics10. What's this for?
A: This allows you to take a "live" audio stream and process it without first recording it to hard disk. With this function, you can make your computer an integral part of your home stereo system, ham radio station or SWL station. Imagine adding the sound of a $15,000 vacuum tube amplifier to all your audio without having to record and process it. As an example - take your CD player and plug its outputs into your sound card input. Take the sound card output and plug it into your home stereo receiver. Now run DCArt10/DC Forensics10 and

choose Multi-Filter. Put some enhancement modules and/or noise filters into the signal path and click on "live preview". All these tools will be applied to every CD you play in real time.

Q: I get an error when I try to use Live Preview mode. What's up?
A: Your soundcard must be able to play and record at the same time. If you have a cheap soundcard, it's time to upgrade!

Q: I have a Whackmaster 6000 sound card I bought in 1992. Is this OK?
A: Who knows? We don't remember that year at all. Just about all of the new sound cards include a new type of driver called WDM. This is Microsoft's effort to fix a lot of earlier problems with various sound card implementations. If your really old soundcard works, then you're OK. Again, you can try going to the Preferences/Sound Card area and selecting MME as your method of communication. In many cases, even the old Whackmaster is supported by this driver method.

Q: I have a 192 kHz / 24 bit sound card. Should I record using these settings?
A: Remember, your recordings will never be better than your originals. If you are recording vinyl records and are making CDs, we recommend 44.1 sample rate, 16 bit, Stereo. If you have high quality masters or are making DVD audio disks, then go with the higher settings. The DCArt10/DC Forensics10 Impulse Filters can benefit from 48 kHz or 96 kHz sampling rates, but higher sample rates and wider bit widths will result in larger files and more processing time.

Q: How do I record my records using a turntable?
A: A magnetic phono cartridge puts out a very tiny signal. You cannot plug a turntable having a magnetic cartridge directly into your soundcard since it doesn't amplify the incoming signal enough. You'll need a preamp. If you have a stereo receiver with a control labeled "phono" on the front, you're probably all set since this has a preamp built in. If your receiver doesn't have this (many modern ones don't) or your receiver is too far from the computer, get a stand-alone phono preamp and seriously considering the use of a flat response unit (like the CTP-xxxx series.

Q: How do I connect my computer into my existing stereo system?

A: On your receiver or master component, take the "tape out" or other line level output and plug it into the line level input of your soundcard. You can also take the line out of your soundcard and plug it into the "tape in" of your receiver though most folks just connect the soundcard output to their computer speakers.

Q: My computer crashes in Preview mode with the Multi-Filter.
A: If there is any weakness in your system, a Multi-Filter preview will likely find it. This can require very large numbers of calculations per second while audio is flowing. The audio can be presented to your sound card in a start/stop fashion as well (though the buffering on your sound card and in DCArt10/DC Forensics10 makes for smooth playback). Sound card designers seldom test for this environment and you can uncover bugs in the drivers when using the Multifilter. Make sure you have your latest sound card and video card drivers and try reducing the amount of video acceleration your video card is set for (this reduces the overall system overhead a bit).

Q: Will increasing the amount of RAM in my computer make DCArt10/DC Forensics10 run faster?
A: DCArt10/DC Forensics10 does not require huge amounts of RAM. If your computer system has 2 GBytes of RAM, then further increases will not appreciably speed up the program. The software uses disk-based processing so hard drive speed and raw processor speed will generally have a greater effect than increased RAM beyond a certain minimum. A faster processor **will** make a big difference in the number of simultaneous filters you can use in the Multifilter.

Q: I want to purchase a system optimized for use with the software. What sort of system will provide me with the fastest performance of all of the various DCArt10/DC Forensics10 filters, effects, and editing features?
A: You should purchase a computer with the fastest clock speed in your price range. Look for hard drives with fast access times and rotational speeds. Purchase a high performance sound card. Lastly, turn off all superfluous programs that may be running in background.

Q: How do I avoid producing dropouts during recording or playback?
A: Make sure that you have reviewed all of the following:

1. Make sure that you are using the latest drivers for your sound card. They can usually be obtained from the card manufacturers or Microsoft's web site.
2. Make sure that the screen saver and all power management functions will not kick-in during recording or playback. By default, the screen saver has a 1-minute timeout, so after 1 minute of no keyboard or mouse activity, the screen saver will kick-in. This flurry of disc activity will put a glitch on the recording or playback of .wav files.
3. Turn off all power saving features, or set their timers to a value of time greater than the longest musical selection that you want to record or play.

Q: I have a vinyl LP that is very noisy, and still has too many clicks after processing. What can I do?
A: Try running the Big Click Filter first, followed by the EZ-Impulse filter. If there is still some impulsive noise left behind, consider running the Expert-Impulse filter, as the third step in the process. Running it twice or more at moderate settings can be quite fruitful. When using the Expert Impulse filter, first run it with the Tracking control set to zero, and adjust the threshold control to remove just the largest clicks. When done, make the destination the source, and re-run the filter with the Threshold set back to 1, and adjust the tracking control to get the smaller clicks. Another thing to try is to use the File Reverse feature, and then process the vinyl recording through the Impulse Noise filter. When done, re-reverse the file. Also, consider trying HQ mode, which is especially good for detecting very small clicks when adjusted correctly (small size values around 5). If you still have clicks left, it's likely because the clicks are longer than the program is allowed to automatically identify and repair. Use the manual Interpolate function (‘I” key) to get rid of any remaining clicks.

Q: I have a record with one major scratch on it running from the center to the outside in a spiral pattern. How do I eliminate the loud "click" which I hear which occurs once per revolution of the record?
A: Try using the Big Click filter on it first. This filter will only remove the major clicks and not the smaller ones, so do not try to get everything with it. Then, you should consider using either the EZ

Impulse Filter or perhaps the Expert Impulse Noise filter to remove the remaining noises. Alternatively, you can try to use the Expert Impulse Filter in the following way. Set the Tracking control to its minimum value. Set the Threshold to its maximum value. Set the "samples" to about five. While in Preview mode, slowly decrease the Threshold control until you see the click counter increment once for each click, which is occurring. Do not increase the Threshold control any further than necessary. Next, increase the "Samples" control until the click is no longer audible. This technique is also useful for getting rid of the clicks produced by cracked 78-RPM shellac records. It is even possible to take a broken 78, glue it together, and then after transferring it, remove the clicks which occur at the breakage points with one of these techniques.

Q: I just transferred an old stereo recording to my hard drive for processing, but I am not sure whether or not the left and right channels are in their correct positions or transposed. How can I determine if they are correct?
A: If the material is contemporary pop, it is difficult to make this determination. However, if the music is orchestral in nature there are some obvious orientations to listen for. For example, the left channel should be dominated by any cymbal crashes or the tinkling sound of a triangle. Also, when Violins come in they should dominate the Left channel. Double basses should be more prominent on the right side of the soundstage. French horns should be heard behind the woodwinds, while trumpets and trombones should generally project from the right rear. If this is not the case, the channels are probably reversed and should be corrected by applying the File Conversion Filter Stereo Reverse feature.

Q: Some of my vinyl records sound more distorted as the play moves towards the end of a side. Why is this? What can be done?
A: This distortion can be due to several sources. A poorly mastered lacquer could have been the culprit. Not all mastering engineers were careful to "hold down" the dynamic range as the recording progressed, creating a great deal of groove wall over-modulation. However, a more likely scenario is that the record was played with a worn stylus. Have someone who is familiar with styli check it out under a microscope to be sure that it is not worn out or damaged. If it checks out to be okay, then your vinyl recordings may have been damaged in past times as a

result of being played many times with a defective stylus. One thing that can be done is to selectively apply a Low-pass filter to the last few cuts of the recording to reduce this annoying anomaly. Also consider applying the Filter Sweeper to compensate for this problem. Sometimes, the Continuous Noise Filter (CNF) operating in “Artifact Suppression” mode can reduce some inter-modulation types of distortion.

Q: How do I generate a simulated stereo Wave file from a monophonic Wave file?
A: Start with a monophonic file that has been de-noised, and convert it to stereo using the File Conversion Filters. Some stereo effect may be added here by applying a little "Time Offset" during this process. Next, make the destination the source, to get a new source file. Run the Reverb effect with a Small or Medium hall, setting the decay to a low number and the early reflections level nearly to zero.

Q: How can I simulate or emulate the sound of a magnetic tape recording?
A: There is a multifilter preset that emulates / simulates a 15 ips (inches per second) tape recording made on a mid-1950s professional tape deck using NAB equalization. The preset is called “Magnetic Tape Emulator 15 ips”.

Q: I get a DirectX Sound error when I go to play or record a file. What is the probably remedy?
A: The most likely reason for the error is the use of MME drivers for sample rates greater than 48 kHz and/or bit depths greater than 16 bits. MME drivers have a limitation of 48 kHz/16 bits. Usually, the solution is to switch over to the WDM drivers found under the Edit/Preferences/Soundcard selection.

Q: Does the order in which I process noise out of a Wave file matter?
A: Yes. Always remove impulsive type noises like clicks and pops with the Impulse Noise filter before de-hissing a recording using either the Continuous Noise filter or the Dynamic Noise filter. Never reduce the bandwidth of an audio signal before applying any of the Impulse Noise filter(s) (and this applies both in the analog side of your signal path as well as the digital side).

Q: I have an analog tape recording with clipping distortion due to over-modulation during the recording process. Is it possible to "soften" the clipping sound in order to reduce the harshness produced during the overloads?

A: Clipping distortion can usually be substantially reduced by applying the De-clipping filter. Analog de-clipping will require slope values greater than 0, however. The de-clipper is the preferred method for resolving this type of audio problem. However, clipping distortion can sometimes also be reduced by utilizing the Impulse Noise filter. If the clipping distortion occurs at the peaks of the waveform, set the tracking control set to its minimum value, and the threshold set to maximum. Highlight a segment of the recording that contains distorted and non-distorted material. In preview mode, adjust the Threshold control until the clipping distortion is reduced. In some cases, it may become necessary to run the Reverse NAB curve before following the above procedure. After the distortion has been reduced with the impulse filter, it will be necessary to run the NAB curve to re-correct the recordings equalization. These two curves can be found in the factory preset listing under the Paragraphic Equalizer filter. The reason for the above two steps is that the saturation overload occurs at the tape-to-tape head interface. The resultant overload is then phase shifted during playback by the NAB equalization circuit in the tape recorder. Utilizing the Reverse NAB curve places the clipping distortion closer to the peak of the waveform, where is actually occurred.

Q: I enjoy removing "clicks" and "thuds" manually from my recordings using the Interpolate feature. Sometimes these signals are so small that, although I can hear them, I can't see them in the waveform. What can I do to make them more visible?

A: In order to improve the view of these signals, first put the system in Sync mode which is found under the View menu. In order to see the "clicks" more clearly, run the High Pass filter with it set to 6 dB / Octave and 10 kHz on your .wav file. The "clicks" will now be more visible in the Destination window and they will be time aligned with the Source Window. Similarly, if you want to view "thuds", run the Low Pass filter with it set to 6 dB / Octave and 100 Hz on your .wav file. As with the clicks, the "thuds" will be more visible in the Destination Window.

Q: I've been using the product for extensive editing, it has always worked perfectly, but now I'm experiencing lockups and shutdowns. What's the deal?
A: Obviously, these problems are not common with this product, but one area you may want to check is your Temp files. DCArt10/DC Forensics10 has a maximum Temp file limit of 999, so if you've been doing extensive editing in Fast Edit Mode, you may have hit that limit and need to clean those temps file out before it will work properly.

Q: How much silence should I provide at the beginning and at the end each .wav file to be compatible with all players?
A: It is recommended that all .wav files have at least 1 second of silence (use the mute function or insert silence under the edit menu) as a lead-in and 1 second of silence as a lead-out so that all players can properly synchronize when sequential files are played. This should help assure glitch-less playback. It is also recommended that files be "Quantized for CD Audio" as found under the CD Prep menu.

Q: How do I determine which version of Direct X that I am running on my system?
A: Go to the windows "Start" button and click on Run. In the "Open" field, type in "dxDiag" and then click OK. A lot of data will be presented, some of which includes the version of Direct X that you are using.

Q: The highlighted display and play cursor appear to be visually "jittery" when editing. How to I eliminate this visual effect?
A: You probably have a second instance of your Diamond Cut software open which is competing for resources. Close the second (minimized) instance and the "jitters" will be alleviated.

Q: How do I clean-up mouth sounds on Voice Over recordings that get past my microphone Pop Screen?
A: Voice Overs (Voice-Over recordings) often suffer from mouth noises due to the close microphone technique used and other background noise anomalies'. These 8 stage Multifilter presets provides you with five special Multifilter systems to address this issue. They include the following choices:

Voice-Over Filter & Signal Conditioner1 (deep)
Voice-Over Filter & Signal Conditioner2 (male)
Voice-Over Filter & Signal Conditioner3 (female)
Voice-Over Filter & Signal Conditioner4 (child)
Voice-Over Filter & Signal Conditioner 5 (shallow)

The Multifilter line-ups are as follows looking from left to right on your Multifilter screen:

Notch 1: This removes phantom power 120 Hz ripple from the signal in 60 Hz power line situations. You will have to change this to 100 Hz for 50 Hz power mains applications.

High Pass 2: This attenuates P-Pops that get passed through the microphone pop screen.

Brick Wall Filter 3: This eliminates Seismic Noise which can be produced by road traffic and HVAC systems.

Continuous Noise Filter 4: This element reduces condenser J-FET random (hiss) noise (and mic preamp hiss) and other random background noises in the recording venue.

Dynamics Processor 5: This reduces sibilant "s" (ess) sounds from the recording.

Lo Pass Filter 6: This eliminates unnecessary high frequency response and thus unwanted high frequency noises.

Dynamic Noise 7: This increases any residual overall mid to high frequency noise.

Dynamic Processor: This acts as a final noise gate for the system.

These filters and their parameters are good starting points based on a basic description. You will need to normalize (normalize gain scaling found under the CD prep menu) to -3 dB first. You can then go into the various filter components and adjust them to provide the optimal result based on your needs. When you are done, you must re-normalize the gain to -3 dB again to be consistent with broadcast standards.

Q: Where do I find my exact registration email address, Serial Number and Registration Code in an earlier version of my Diamond Cut Software so that I can perform an upgrade to a newer version?
A: These data are found under the "Help About Diamond Cut xxx " (DC-Art menu item of your older software program.)

Q: How long is my support time period after purchasing my Diamond Cut Software Product?
A: Free support is provided for 1 year from the date of product purchase. Support is available via phone, the internet, or via the DCForum. Extended support time can be purchased from our website.

Q: Is additional support available for my Diamond Cut Software Product?
A: Yes. You can purchase additional support time from the Diamond Cut Productions website (www.diamondcut.com). The cost for additional support is 10% of the full version list price per year.

Q: Is there a good demo .wav file to show some of the overall features of the Forensics Version of the software?
A: Yes, there are several found under the File/Open Demo Wave Files. The primary demo file is called "Forensics Demo". A preset found under the Multifilter is already set up to fix the audio on this file and is called "Forensics Demo Clean-up Filter" which contains a line-up of multiple sub-filters.

Q: What characters can legitimately be used in my product registration "User Name"? (Legitimate characters)
A: The following characters are legitimate for use in a User Name:
abcdefghijklmnopqrstuvwxyzABCDEFGHIJKLMNOPQRSTUVWXYZ 1234567890~-_.!*(),

Q: What is the maximum file name number of characters allowed?
A: The maximum file length in terms of the number of characters is 260 total.

Q: What is the minimum number of characters required for a legitimate email address for registration of Diamond Cut Software Products?
A: 5 (five)

Q: Where is the printable form of the product documentation found in .pdf format?
A: Printable Documentation is found under the Diamond Cut Program grouping. The Forensics version is called DC Forensis10 Users Manual.pdf and the DCArt version is called DCArt10 Users Manual.pdf. For example, one pathway to the DCForensics10 documentation is as follows:

Go to the Windows Start button and click it.
Click on “All Programs”
Click on “Diamond Cut Audio”
Click on “Forensics10 Users Manual.pdf”

Q: Is the Diamond Cut Users Forum free and where can I find it?
A: The Diamond Cut Users Forum is free, but registration is required if you want to post questions. It is found at the following link:

http://www.diamondcut.com/vforum/

Q: Are educational and training programs provided for the use of Diamond Cut Software?

A: Yes. We have a line of DVD Videos for training at your leisure which can be purchased on our store at:

https://www.diamondcut.com/st3/product-category/videos/

We also offer three day training courses which are given throughout the USA which provides a certificate of course completion at the end of the program. Call us at 973-316-9111 to learn about the next one scheduled near you. Also, several colleges and universities include training in the use of Diamond Cut Forensics Software as part of their Forensics Science undergraduate programs.

Q: What are the various file paths used by the DC Forensics10 and DCArt10.8x Software?
A: All of the user data is consolidated into a directory called DC Forensic10 in the users "Documents" or "My Documents" directory. The file paths under this directory are as follows:

DC Tunes database:	\DCForensics10\DCTunDb.xml
Presets:	\DCForensics10Presets
Example Wavefiles:	\DCForensics10\Wavefiles
TempFiles:	\DCForensics10\Tempfiles

To get to these Files, use the following paths:

<u>For Vista and Windows 7 or Windows 8 or Windows 10</u>

C:\Users\<user>\Documents\DCForensics10

<u>For Windows XP</u>

C:\Users\<user>\MyDocuments\DCForensics10

Forensics version 10 also provides shortcuts which are located in the user's program menu to access these files.

Note: For DCArt10 versions, just substitute DCArt10 where you see DCForensics10 in the path.

Q: What is the support time period for my product and can it be extended?

A: 1 Year from time of purchase; extended support can be purchased from our store at www.diamondcut.com. for the shipping Forensics version only.

Q: Are formal classroom training courses available in the use of the Forensics software?

A: Three day training courses are available on an as needed basis. The courses include both classroom and laboratory modules. Contact us for details.

Q: Are training DVDs available pertaining to the software and it's use?

A: Yes. You can find several training DVDs on our website which you may find to be useful. Look in our store at www.diamondcut.com

Q: Where can I find additional technical information pertaining to these software products?

A: A knowledge base is located at the following link:

https://www.diamondcut.com/st3/knowledge-base/

Reporting a Problem

Your software distributor has operators available during most days during normal business hours. They provide technical support for registered users only. When you contact the distributor from which you purchased your software, please be prepared to provide the following information:

- Sound Card Type
- Version of the Software (Found Under Help\About Diamond Cut Software)
- Version of Windows
- Being specific as possible, outline the problem, how it occurs and its frequency
- Version of Direct X being used
- Your User Name (as entered into the software at first installation)
- Your Software Serial Number
- Your Software Registration Code Number
- Your email address that you used to register the product

Most distributors prefer email for an initial contact, in that it gives their technicians a chance to study the problem before they provide you with an answer, rather than experimentation while you're waiting on the phone. Our distributors normally will have an answer for you within one business day. If you need to communicate with the factory directly, here is the contact information:

Contact and Support Information

Diamond Cut Productions, Inc.
P.O. Box 305
Hibernia, NJ
07842
www.diamondcut.com
support@diamondcut.com

Telephone: 973-316-9111 Fax: 973-316-5098

Section 5 – Useful General Information

Glossary of Terms

Acoustical Impedance
Acoustical impedance is the total opposition provided by acoustical resistance and reactance to the flow of an alternating pressure applied to a system. More specifically, it is the complex quotient of the alternating pressure applied to a system by the resulting volume current. The unit of measurement is the Acoustical Ohm.

Acoustical Reactance
Acoustical reactance is the imaginary part of the acoustical impedance. Energy is not dissipated by acoustical reactance; it is only stored there. The unit of measurement is the Acoustical Ohm.

Acoustically Mastered
Acoustically mastered record recordings utilized only the energy of the sound waves created by the sound source to modulate the master cutting lathe stylus. This recording technique had none of the benefits which signal amplification can provide to the recording process. This is the method that was utilized from the time of the invention of the phonograph by Thomas Edison in 1876 up until around 1925, when vacuum tube amplifiers and microphones began to be employed in the mastering process. By 1929, all of the major record companies had switched over to the "electrical process" of record mastering.

Acoustical Resistance
Acoustical Resistance is the real term of the acoustical impedance relationship. This is the term responsible for the dissipation of energy. The unit of measurement is the Acoustical Ohm.

A-D Converter
A device used to convert analog signals into digital (discrete time) signals, so that they can be signal processed by a computer algorithm. The sound card in your computer contains an A-D converter and also a D-A (Digital to Analog) converter. To be compatible with DCArt10/DC Forensics10, it must have at least 16-bit resolution to realize the performance of the product. However, the software does support 8 through 24 bit resolution sound cards. In other words, your sound card must be able to divide the amplitude of audio signals into numerically sampled representations, the smallest division being one part in 65,536 (2^16). 16 bit audio has the same resolution as "red book" CD Audio.

A-law Compression
This is a slight variation of Mu-law compression, which is used in Europe. For more information, please refer to Mu-law Compression. DCArt10 & DC Forensics10 support this format.

ADPCM
(Adaptive Differential Pulse Code Modulation)
ADPCM is a data compression method in which an audio signal is quantized by the difference between the input reference signal and a prediction that has been made of that same signal. When the prediction between the actual and the predicted audio signal exhibits bits having a low variance, it is accurately quantized and fewer bits are required for digitization compared to standard PCM. Your software supports this file format.

AES
The **A**udio **E**ngineering **S**ociety
60 East 42nd Street, Room 2520
New York, NY
10165-2520
USA
1-212-661-8528 / http://www.aes.org

AGC or ALC
Automatic Gain (AGC) or Level Control (ALC). This is a set of algorithms, which can be found in the Dynamics Processor and Punch & Crunch effects that maintain the system gain at a relatively constant value independent of input signal level. Signals below a set threshold value are upwardly

expanded while signals above that threshold are compressed. This is particularly useful in forensics applications wherein the various parties who are communicating with one another are recorded at greatly varying relative levels.

AIFF
AIFF is the acronym for a PCM (Pulse Code Modulation) audio file format known as "Audio Interchange File Format". It is used primarily on the Macintosh platform. DCArt10 and DC Forensics10 facilitates conversions from this format to .wav.

Aliasing (Temporal Aliasing)
In sampled data systems, aliasing can occur on signals higher than the Nyquist frequency. These are not real representations of the source material. They could be looked at as parasitic signals or digital artifacts. For example, if an audio system is sampling data at 44.1 kHz, its Nyquist frequency is 22.05 kHz. If an audio signal exists at 23 kHz, an alias signal will be produced at the difference between those two (or 0.95 kHz). That 0.95 kHz signal would sound like a digital artifact to the listener (and be quite unpleasing). An anti-aliasing (brick wall) filter is usually applied to the signal before the A/D conversion process to avoid this problem. In the example above, the brick wall anti-aliasing filter would be set for around 22 kHz.

Algorithm
An algorithm is a step-by-step procedure for solving a mathematical problem.

Ampere
(I)
The unit of electric current that is equal to one Coulomb of electric charge flowing per second.

$I = V / R$,

wherein:

V = Voltage in Volts,

and

R = Resistance in Ohms

(also, see Ohms Law.)

Amplifier
An amplifier is an electronic system which enables an input signal to control power from a source independent of the signal and thus be capable of delivering an output that bears some relationship to, and is generally greater than the input signal. An audio amplifier performs this function producing a relatively linear relationship between the input signal and the output signal. For more information on audio amplifiers, refer to Pre Amplifier and Power Amplifier in the Glossary section of the Help File.

Amplitude
Amplitude is the loudness (or intensity) of a sound at any given moment in time, which is represented on the vertical axis of the Diamond Cut time domain workspace areas. Amplitude in audio terms is usually expressed in relative terms (the ratio of two levels) in dB (decibels), although sometimes it may be represented in absolute terms such as Volts or SPL (Sound Pressure Level).

Amplitude-Frequency Distortion
Amplitude-Frequency distortion is a deviation from a perfectly flat frequency response over a range of interest (usually 20 Hz to 20 kHz).

Amplitude Modulation (AM)
Amplitude Modulation is a technique for producing a radio carrier containing an audio signal in which the envelope of the carrier is varied in amplitude in proportion to the level of the audio modulating signal. With Amplitude Modulation transmission, the rate at which the carrier amplitude changes is proportional to the audio modulation frequency. AM carriers are ideally fixed (constant) in frequency.

Analog
An electronic system in which signals are represented, amplified, and processed utilizing continuous Voltages and/or currents (whose value could be expressed as an irrational number at any point in time) which are not quantized. DCArt10/DC Forensics10 utilizes many digital simulations of analog systems within its algorithms.

Aromatic
Solvents that are made up of cyclic hydrocarbons or their derivatives such that, after they evaporate, tend to leave little or no

residue on the surface on which they had been applied.

Attenuate
Attenuation is the process of signal reduction, which is the opposite of the process of signal amplification. Most filters attenuate signals outside of their pass-band and feed signals through with no attenuation (or amplification) within their pass-band. Some filters, such as parametric and graphic equalizers are configured to provide either amplification or attenuation at any given frequency. Devices such as volume control potentiometers, "L" Pads, "T" Pads and "H" Pads are used to attenuate signals independent of frequency, i.e. flat. "L" Pads hold either the input or the output impedance constant as the attenuation factor is modified. "T" Pads hold both the input and the output impedance constant as the attenuation factor is changed. "H" Pads perform the same function as "T" Pads, only for balanced line systems. See a table of resistance multipliers for a symmetrical (equal input and output impedance) “T” Pad attenuator in the ***Charts, Graphs and Other Useful Info*** section of this User’s Guide.

Audio Frequency Spectrum
The range of frequencies between 20 Hz and 20 kHz. A very high quality audio system capable of reproducing this frequency range should be able to do so within +/- 3 dB.

Azimuth
When analog magnetic tapes are recorded or reproduced, the gap of the respective head (recording or playback) should ideally be perfectly normal (perpendicular) to the direction of the tape movement. If, in either of the two mentioned processes, the respective head gap is off-normal (off-azimuth,) two types of signal degradation will occur. The first phenomenon results in the loss of the high-end of the audio spectrum frequency response. The second effect produces a phase shifting of one channel with respect to the other, thereby "smearing" the stereophonic image. A similar phenomenon occurs when a monophonic half-track reel-to-reel tape is reproduced on a quarter track machine. Azimuth problems can be corrected to some degree by utilizing the "Time Offset" feature found in the File Converter, which is found under the Filter Menu.

Band-pass Filter
A Band-pass filter allows only a range of frequencies to be passed from its input to its output without attenuation. A wide Band-pass filter is one in which an upper and a lower corner frequency need to be defined, and often several octaves will be passed in between without attenuation. A narrow Band-pass filter is one in which only a center frequency needs to be defined, and often has a bandwidth of an octave or less. The center frequency for a narrow Band-pass filter is sometimes referred to as its resonant frequency. There are different Band-pass shapes that can also be defined for narrow Band-pass filters.

Band-stop Filter
A Band-stop filter provides the inverse response of a Band-pass filter. You can find a Band-stop filter within the Brick Wall filter suite found under the Forensics menu of your software.

Berliner, Emile
Emile Berliner is widely known as the one who commercialized the lateral cut disc record format. He first introduced his products into the market place in 1895, although he had spent the previous 10-year period developing his system. However, Emile Berliner was not the inventor of the disc format or the lateral cut method for creating the undulations on a surface. These principles were outlined in the earlier Edison phonograph patents.

Bipolar Junction Transistor (BJT)
The bipolar junction transistor (BJT) is a three terminal solid state device having a base (b), emitter (e) and a collector (c). They come in two configurations, PNP and NPN which relates to their polarity and are made from doped germanium (early devices) or silicon. They are used in amplification, oscillator and switching circuits. They are often characterized in terms of their current amplification factor or beta (β) such that the current flowing in the Collector is controlled by a smaller value of current flowing in its base (given a common emitter configuration) per this relationship:

$$I_c = \beta \times I_b$$

Blast

"Blast" is a term that is used to describe a passage of sound on a recording which is disproportionately louder (and often distorted) compared to the rest of the recording. "Blasts" can be created by poor instrument placement on acoustic recordings, poor mixes on electrical recordings, or by poor planning of microphone placement in live recordings. The term "blast" was used by recording engineers at least as early as the 1920's.

Blue Amberol
Blue Amberols were the third (and final) generation of cylinder records which the Edison Company commercialized. They were about 4 minutes in length. These records were made of a celluloid recording surface mounted on a plaster of Paris core. They were an improvement on the Edison Gold Molded black wax cylinders, which were only two minutes in length. The rotational speed for Blue Amberols (and black wax cylinders) is 160 RPM.

Broadcast Wave (.wav) Format (BWF)
Broadcast Wave or BWF (.wav) is an audio file format that facilitates the inclusion of certain metadata within a .wav file header. It adds a "Broadcast Audio Extension" chunk to the basic Microsoft .wav format. It is a European standard per EBU document Tech 3285, version 1 (July 2001). BWF provides a seamless exchange of audio data between differing broadcast environments and equipment based on varying computer platforms. This format is supported by your Diamond Cut software program. You can create BWFs via the Save As feature found within the File Menu. You can edit the BWF header information via a dialog box within the Edit Menu. Complete information regarding this format is beyond the scope of this document and can be found on the internet.

Brown Noise
Brown noise (sometimes referred to as Red Noise) is a form of random noise exhibiting a Gausian distribution, which mimics Brownian motion. Its power spectrum is proportional to 1 / f^2, exhibiting a – 6dB / Octave slope (20 dB/decade). Your Diamond Cut software can produce Brown noise via two steps. First, create a random noise file using the Make Waves Generator. Then, bring up the LIVE/Multi-Filter and apply the preset called "White to Brown Noise Converter".

Buffer
A buffer is a memory sector that is used as a temporary storage location during input and output operations. The "preview buffer" length is programmable in DCArt10/DC Forensics10, and is found in the Preferences section of the Edit Menu.

Butterworth Filter
A Butterworth Filter produces a maximally flat amplitude characteristic in the pass band or the reject band (depending on whether it is used as a Band-pass or a notch filter). Its characteristic also applies to higher order (2nd order or higher, IIR type) high-pass and low-pass filters. It produces a critically dampened response at their corner frequency(s), having no ripple, and therefore it introduces little distortion into the signal that is feeding it. The Butterworth filter's poles of signal transmittance are uniformly spaced on a semicircle, having its center on the imaginary axis. Its half-power frequencies are those at which the circle intersects the imaginary axis. The low-pass, high-pass, and band-pass filters offer the Butterworth response as an option.

Buzz
Buzz usually refers to a series of harmonic noise signals related to the frequency of the AC power mains. It differs from "Hum" in sound, because it usually contains a large number of higher frequency harmonics (created by phase controlled lighting dimmers or other non-linear systems connected to the power line). Buzz is best attenuated using the Harmonic Reject filter, the Spectral Filter or either the Expert Impulse or Narrow Crackle Filter(s).

Byte
An eight-bit word. Each sample of a monophonic .wav file is generally represented by two eight-bit bytes. Two eight-bit bytes are used to represent all of the integer numbers between 0 to 65,535 (actually +/- 32,768), which is the total dynamic range of DCArt10/DC Forensics10 when it is operating in 16-bit mode.
1 Kilobyte (or 1 Kbyte) = 1,024 bytes

Capacitance
(C)

Capacitance is the ratio of the electric charge given to a body compared to its resultant change of potential. It is usually expressed in Coulombs of charge per Volt of potential change and its basic unit is the Farad. Energy is only stored (but not dissipated) in theoretical (ideal) capacitance. Time constants for audio filters are created with a combination of resistors and capacitors in various configurations. High-pass, Low-pass, Band-pass, and Notch filters can all be created with the appropriate combinations of resistors, capacitors, and amplifiers (usually op-amps). The corner frequency for a simple first order RC filter = 1 / 2 pi* (R x C). The principle of capacitance (and conservation of charge) is involved in the operation of condenser and electret microphones and electrostatic loudspeakers and headphones.

F = (Farad) and μF = micro Farad

μF = 1 X 10 ^ -6 Farads

pF = 1 X 10 ^ -12 Farads

*Note: pi = 3.141592654 (approximately)

Cassette Tape Equalization Time Constants

Compact Cassette tapes (which operate at 1 7/8 ips) commonly utilize one of the following two equalization time constants based on the tape type:

1. Normal (IEC Type 1) (Usually Ferrous Oxide based): 120 μSec
2. High (IEC Type 2) (Usually Chromium Oxide based): 70 μSec

Cent

1,200 cents = 12 semitones = 1 Octave in the scale of "Just Intonation". In other words, the interval between two tones whose basic frequency ratio is the twelve-hundredth root of two is the "cent".

Cepstrum (Power)

Power Cepstrum displays are a way of graphically representing speech patterns using a specialized transform of an audio signal. It is not a reversible transform. It produces a unique view of the information relating various frequency bands associated with a signal and is useful for separating vocal tract information from the exciting pitch in voiced speech. Mathematically, the Power Cepstrum of a Signal = $|F\{\log(|F\{f(t)|^2)\}|^2$. You can create cepstrum graphs using your DC Forensics10 version of this software package. The feature is found under the Forensics Menu under the Voice ID feature.

Cepstrum (Complex)

A Complex Cepstrum is a reversible transform and can be useful for the graphic display of the vowel sounds produced by the human voice. It is also useful in identifying echoes created by acoustics in a room. Mathematically, the Complex Cepstrum = FT (log)|FT(f(t))|) + j2πm) where FT is the signals Fourier Transform and m is the integer value needed to unwrap the imaginary component of the complex log function. The horizontal axis is called the quefrency which is related to the rate of occurrence of a particular harmonic contained within the signal. This transform is implemented in the Voice ID feature found under the Forensics Menu of the DC Forensics Audio Laboratory version of this software.

Charge (Coulombs)

Charge in units of Coulombs is the product of Voltage & Capacitance (Q = CV). C is given in Farads and V is given in Voltage. Also, refer to "Ampere" for the time relationship of the Coulomb.

Compact Discs

Compact discs are digital storage devices, typically 120 mm in diameter by 1.2 mm thick that are made of a polycarbonate compound. They can be used to store data, audio, video, and photographs. The active region of a compact disc begins at 46 mm from its center and runs to 117 mm. Reading of such discs is accomplished typically with an AlGaAs laser diode (producing light in the infrared region of 780 nM) in conjunction with a phototransistor sensor. There are a number of standards for data contained on Compact Discs. For a chart that describes these details, simply turn to our ***Charts, Graphs and Other Useful info*** pages.

Chebyshev Filter

A Chebyshev responding filter is one in which the pass-band produces a variation in attenuation between zero and its maximum value, and the band-stop requirement is only to increase monotonically to infinity. These filters display relatively steep roll-off characteristics, but, unlike the Butterworth filters, have ripple in their pass-band. Our IIR based Low-pass, High-pass, and Band-pass filters offer the Chebyshev response as an option.

Chirp Z-Transform
A Chirp Z-Transform (CZT) is used in the Diamond Cut High Precision Spectrum Analyzer to produce a higher resolution and signal processing speed compared to the Standard Precision Spectrum Analyzer. It is essentially a more advanced FFT technique or algorithm used to numerically calculate the Z-Transform of a sequence of a defined and limited number of samples relying on the fact that the values of the Z-Transform lie on a circular or spiral contour.

Classification of Amplifiers
Audio Amplifiers can be broken down into several classifications based on their devices degree of conduction relative to its input signal. The VVA Virtual Valve Amplifier utilizes two of the following classifications. The others are included in the description for completeness*:

<u>Class A:</u> The device or devices conduct for a full 360 degrees of the input signal. These amplifiers can be wired either in single-ended or push-pull configurations. Class A Audio amplifiers are usually used in pre-amplifier stages, or low power amplifier applications. This circuit has the poorest electrical efficiency, but produces predominantly even order distortion.

<u>Class B:</u> Two devices are operated out of phase with respect to one another. Each device conducts for only 180 degrees of the input signal. When the two amplified signals are combined, the full input waveform is represented, only amplified. This type of circuit is plagued by a phenomenon known as "crossover distortion" at low signal levels. This configuration is reserved for low performance PA amplifiers or AM (Amplitude Modulated) communications modulators. It is electrically efficient, but produces relatively large values of harmonic distortion especially at small signal levels.

<u>Class AB:</u> Two devices are operated out of phase with respect to one another, just the same as the Class B configuration. However, each device conducts for more than 180 degrees of input signal, but less than 360 degrees. This configuration produces a reasonable tradeoff between electrical efficiency and low distortion. This configuration is especially good at reducing crossover (notch) distortion (at zero crossing). It is commonly used in high power, high quality audio power amplifiers. Since the circuit is symmetrical, distortion levels can be quite low (especially even order products).

<u>Class C:</u> This configuration can consist of one or two devices which are conducting for anywhere between 90 to 180 degrees of the applied input signal. It is reserved for RF circuits only.
*Note: There are additional classifications of amplifiers involving tap switching, multiple rail, and pulse width modulation techniques, which have not been included in this listing.

<u>Class D:</u> Instead of operating power amplifier active devices in their linear range of operation, these same electronic devices can be used as state machines meaning that they are either on or off, and only in their linear region during transition (a parasitic state). Class D amplifiers operate in that manner, thereby minimizing power dissipation. The devices, which are acting as switches, are controlled by a pulse width modulator (PWM) and deliver power to the load via a Low-pass filter, which is used to attenuate the carrier or switching frequency. The rate of change of pulse width is proportional to the modulating audio frequency, while the magnitude of the change in pulse width is proportional to the modulating audio frequency amplitude.

<u>Class G & H</u>: These are “special case implementations” of classes A, B, and AB in which multiple power supply rails or variable rail voltages are used to minimize the power dissipation in the output devices while still allowing them to operate in their linear region.

Clipping
Clipping is a phenomenon, which occurs when a signal (or numerical value) exceeds a system's headroom. This concept applies to both analog and digital systems. The result of clipping is distortion. The amount

of distortion produced depends on the amplitude of the over-driven signal. In DCArt10/DC Forensics10, clipping will occur anytime a signal or calculation produces a numerical value greater than 2^15 (or 32,768 counts or LSB's) when using it in 16 bit mode of operation (one bit is the sign bit and that is why it is not 2^16). Clipping can be observed as a flattening of the slope (horizontal line) of a signal at its peak on the Source or Destination workspace displays.

CMRR (Common Mode Rejection Ratio)
CMRR, or Common Mode Rejection Ratio, is the ability of a balanced audio transmission system to reject in-phase signals that appear on the two input lines. It is usually expressed in dBu. For example, a CMRR figure of –60dBu means that 1 / 1000th of a common mode signal presented to the transmission line is converted into a normal mode signal (which is the signal of interest).

Co-Axial Cable
A coaxial cable is one constructed in a manner in which the signal conductor is located within the center of the return conductor with a dielectric located in-between. This provides three notable characteristics for the cable:

1. The center conductor is shielded from the effects of "E" fields that may be present. "E" field coupled current is returned back to signal ground with little effect on the signal itself.
2. The loop area formed between the two conductors is very small compared to other types of conductors thereby minimizing inductance and also susceptibility to "H" field coupling.
3. The cable exhibits a characteristic impedance, which is independent of cable length (after past a few wavelengths) which is of a constant value related to its ratio of distributed inductance and capacitance. This makes the cable suitable for carrying RF (radio frequency) signals over long distances.

Co-Axial cables are often used to carry low-level signals from one audio device to another because of the first two mentioned characteristics.

Codec
Codec is an acronym for Coder-Decoder and pertains to the process of data compression and decompression. Audio examples include such algorithms as the Mp3, A-Law, Mu-Law, and the ADPCM formats. Video examples of Codec's are the various MPEG formats.

Columbia LP Equalization Curve
This equalization curve pre-dated the RIAA curve and was in use by the Columbia label and others in the early LP days of the 1950's. Here are the key frequency inflection points for the early Columbia LP Curve:

1. 5310 uSec (30 Hz) (pull-out or shelf frequency)
2. 531 uSec (300 Hz) (turnover frequency)
3. 99.4 μS (1600 Hz) (roll-off frequency)

This curve can be decoded with the Diamond Cut Virtual Pre Amplifier.

Comb Filter
A comb filter (or Harmonic Reject filter) is a wave reject filter whose frequency rejection spectrum consists of a number of equi-spaced elements resembling the tines of a comb. This filter is useful for getting rid of "Buzz" type noise containing more than just the line frequency fundamental component. In DCArt10/DC Forensics10, it is called the Harmonic Reject Filter, and for more details, please refer to the same. The Spectral Filter found in the Forensics version of the software can also be used to create comb and inverse comb filters.

Compressor
A compressor is an electronic device that is used to reduce the dynamic range of an audio signal. They are often used to prevent overloading on certain mixer inputs (i.e. drums and vocals) in live performance applications. Radio stations often use compressors to make their signal "sound louder" when a potential listener is tuning across the radio band. This technique avoids violation of any FCC regulations regarding maximum % modulation or modulation index, yet still raises the perceived loudness of the station.

Corner Frequency

The corner frequency of a filter is the frequency at which the signal has been attenuated by 3 dB relative to the pass band region of the filter.

Crackle
Crackle is a term used to describe relatively low levels of impulse noise found on old phonograph recordings. It is very similar to impulse noise, only the peak amplitude is much smaller in comparison. Crackle sort of sounds like Rice Krispies just after you pour the milk into the dish. Crackle is usually caused by slight imperfections in the record-playing surface due to the use of coarse grain fillers in the record composition. Sometimes, crackle is caused by gas bubbles that occur in the surface as the record "cured" after the stamping process. Crackle can be filtered out most effectively with the EZ Impulse, Median, or Continuous Noise Filter. Very old acoustic recordings may be even more effectively de-Crackled (and de-Hissed at the same time) with the Average Filter.

Crest Factor
Crest Factor is the ratio of the peak value to RMS value of a signal (acoustical or electrical) over a defined time interval. For a 50 % duty cycle steady state square wave, this value is 1.0000. For a non-distorted sine wave, this value is 2^-2 (or about 1.4142). For audio signals, crest factor varies greatly depending on the material and the integration interval over which it is calculated. Classical music, for example, generally exhibits much higher values of crest factor compared to contemporary pop music when measured over an entire performance.

Crosstalk
Crosstalk is a figure of merit describing the degree to which one audio channel "bleeds" into another and expressed in dB. For example, a crosstalk figure of –60 dB indicates that the specified channel is affected by adjacent channels with an insertion ratio of 1000:1.

Dampening Factor (power amplifiers)
The dampening factor for a power amplifier to loudspeaker interface is the ratio of total load impedance divided by the closed loop power amplifier output impedance, or Zl / Zout(amp). The Zl term includes the speaker driver impedance at a given frequency and the cable impedance feeding it. Generally speaking, high dampening factors are desirable for frequencies below around 200 Hz to minimize system resonance and "muddy" bass responses. Dampening factors of around 10 to 20 are adequate for most applications. The law of diminishing returns comes into play at higher dampening factor values. Generally speaking, vacuum tube amplifiers exhibit lower dampening factors compared to their solid-state counterparts. High dampening factors relate dominantly to the amount of negative feedback used in the power amplifier. Greater values of negative feedback yields lower amplifier output impedance values, thus producing higher dampening factor values in a given system. But, line loss (resistive) ultimately will dominate the dampening factor performance of a system.

dB (decibel)
1/10 of a Bel. A Bel is the basic unit for the measurement of sound intensity. It is a log scale measurement system used for relating the ratio of two acoustical or electrical parameters. Since electrical Voltage, Current, and Power are used to represent sound through audio signals, the following mathematical relationships may be found to be useful when relating them in terms of outputs and inputs:

dB (Voltage) = 20 log V output / V input
dB (Current) = 20 log I output / I input
dB (Power) =10 log P output / P input

<u>**Note:**</u> A doubling of a Voltage or Current represents a 6 dB change. A doubling of Power represents a 3 dB change. For a table detailing the relationship between Voltage, Current and Power ratios in Decibels, turn to our Charts, Graphs and Other Useful Info section of this User's Guide.

dBm
dBm is the power level of a signal expressed in dB, and referenced to 1 milliWatt (0.001 Watt).

Since:

V = (P x Z) ^ ½

wherein:

V = Voltage in Volts

P = Power in Watts

and

Z = Impedance in Ohms

therefore:

In a 500 Ohm audio line distribution system,

V at 0 dBm = 0.707 Volts

dBu
Pro-level analog signals are often called out in terms of dBu and are at values of +4 dBu which is 1.228 Vrms or 3.47 Volts pk-pk (sine wave based).

dBv
dBv is the Voltage level of a signal expressed in dB, and referenced to 1 Volt rms sine wave. If a pure sine wave is the reference signal, its value would be 2.83 Volts peak to peak at 0 dBv. Most consumer audio equipment produces output levels of – 10 dBv (line level).

D-A Converter
A D-A Converter is a device designed to convert digital signals back into analog signals so that they will be compatible with analog sound reproduction equipment. DCArt10/DC Forensics10 requires a D-A converter with 16-bit resolution (2^16 = 65,536 counts). It also supports sound cards capable of up to 24-bit resolution (2^24 = 16,777,216 counts) .

DC-Art
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DC Offset
A DC offset is a fixed value of Voltage that may have been added to a signal inadvertently. It contains no audio information. It is the “b” term in the general expression y = mx + b. It can be eliminated by feeding the signal through the DCArt10/DC Forensics10 high-pass filter set to 10 Hz and a slope of 6 dB/ Octave. The High Pass Filter includes a Preset to eliminate DC Offset from a file.

De-Emphasis
The reversal of a pre-emphasis process. See Pre-Emphasis for more information.

De-Ess
De-essing is the process of decreasing the sibilance of over-modulated "ess" sounds on recordings produced by the human voice. The DCArt10/DC Forensics10 dynamics processor contains an algorithm for De-essing a signal containing this particular anomaly.

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Diamond Discs
The trade name for the records in the disc format produced by the Edison Company was "Diamond Disc". These records were cut vertically (hill and dale) and could only be played on Edison Diamond Disc phonographs designed for this purpose. They rotate at a speed of 80 RPM. To extract the vertical component of a signal provided by a stereo cartridge when transferring Diamond Discs, use the DCArt10/DC Forensics10 Mono (L-R) File Conversion feature.

De-Esser
A De-Esser is a non-linear system designed to attenuate the overly sibilant pronunciation of the letter “s” which can occur on some recordings. A De-Esser can also be used to reduce other forms of harmonic distortion which may be present on a recording. The Diamond Cut De-Esser is found within the Dynamics Processor routine and is enabled via a checkbox.

Differentiator
A Differentiator is a device, system, or process which evaluates the rate of change of one parameter with respect to another. In its generic form, it is expressed as f’(x) =

dy/dx. The reversal of the process of Differentiation is Integration. The Differentiator is found as one of the presets under the Diamond Cut IIR based High Pass Filter.

DIM (Dynamic Inter-modulation Distortion)
Refer to TIM

Distortion
Distortion is a general term used to describe the undesirable effects that an audio system or process can have on the source input signal. There are many types of distortion, with some of them listed below:

1. Harmonic Distortion
2. Inter-modulation Distortion
3. Clipping Distortion
4. Crossover (Notch) Distortion
5. Phase / Jitter Distortion
6. Transient Inter-modulation Distortion (TIM) which is sometimes referred to as Dynamic Inter-modulation Distortion (DIM) or Slewing Induced Inter-modulation Distortion (SID).
7. Amplitude – Frequency distortion

Dither
In control loops, dither is the addition of a useful oscillation or noise signal into the system to overcome friction or hysteresis. This improves the response of the control loop to very small changes of the system reference signal. This principle has been extended to digital audio. In this case it implies the addition of a random noise signal inter-modulated with the LSB of the audio signal, effectively increasing the resolution of the system. Dither (with various options) is included in the File Sample Rate Conversion feature.

Double-Ended Noise Reduction
Double-Ended noise reduction involves encoding an audio signal when recording the same by some form of compression and reversing that process on playback. There are many different schemes for performing this function, and you can even create your own using the various algorithms contained within DCArt10/DC Forensics10. The Dynamics processor or the Dynamic Noise Filter can be used to create various encoding / and or decoding schemes. It is important to match the encoding and decoding corner frequencies, thresholds, attack and release times in order to achieve good Double-Ended noise reduction results. The rest is left up to the creativity of the user.

Drive
Drive refers to the amplitude of a signal that is applied to an amplification device such as an electron tube or transistor. It represents the ac component, rather than the dc (or quiescent) component applied to the input. Since the above-described devices are intrinsically non-linear with regard to their transfer function, the larger the value of drive applied to the device, the greater will be the harmonic by-products. The DCArt10/DC Forensics10 Virtual Valve Amplifier allows you to adjust the drive level to the various amplifiers to vary the degree of "tube warmth" produced by the system. The program automatically compensates the output level, so that large values of drive do not produce substantial changes in overall system gain.

Dry
"Dry" is the term used to describe the signal output of a special effects generator (such as the Reverb) which contains only the non-processed signal. "Wet," on the other hand, refers to the effect signal alone. Like most special effect generators, the Reverb has an output mix control which allows you to transfer a signal from the effects generator which ranges from completely dry, to completely wet (no source signal), or to some mixture in between.

DTMF (Dual Tone Multi Frequency)
(Also known as Touch Tone)
DTMF is the dual tone encoding system used on the telephone system for dialing. Two frequencies are allocated for each number on a telephone keypad. For a chart that contains the Touch Tone dual frequency pairs, simply turn to the ***Charts, Graphs and Other Useful Info*** section of this User's Guide.

Dynamic Filter
A filter in which its corner frequency is varied as a function of another parameter associated with the signal content of a sound source. Most often the corner frequency is that of a Low-pass filter that is modulated by the rectified output of a High-pass filter,

although other schemes are possible. This sort of system changes bandwidth on the fly, and in co-ordination with the occurrence of high frequency content present in the source. It can be done either in a feedback or in a feed forward manner, with advantages and disadvantages attendant to each technique.

Dynamics Processor
An electronic device used to modify the characteristic dynamic amplitude response of an audio signal. These circuits or algorithms can compress, expand, and de-ess (remove overly sibilant "esses") audio signals.

Dynamic Range
The dynamic range of an audio signal theoretically is the ratio of its smallest to its largest resolvable level or value. For a digital system, each bit represents a doubling of the signal so its dynamic range is simply its number of bits x 6 dB. In practice, with very high-resolution digital systems, other parameters may come into play such as the value of the systems noise floor, which may render some of the systems dynamic range unusable. For a table of common values of audio system resolution and their associated dynamic range, simple turn to our ***Charts, Graphs, and Other Useful Info*** section.

Ear (hearing)
Your "ear" is the most critical piece of equipment that you will be using in the audio restoration process. It is important to realize that audio restoration is half science and half art. If you are only restoring audio for yourself, the audio restoration process is much less demanding as compared to situations where you may be performing the job for broad-based public consumption or for forensics purposes. In the latter cases, there are two very critical aspects of your "ear" which must be considered:

a. You must have a good sense of hearing. If you have a hearing deficiency, you may have a difficult time making the subjective judgments that are critical to the production of a commercially viable product, which will be acceptable to the "ear" of the general public. For example, if the "top-end" of your hearing is missing, it is more likely that you will produce restorations that seem harsh, hissy, and containing too many digital artifacts as far as the general public is concerned.

b. Even if you have an exceptional sense of hearing, you will need to develop a good "ear" for what the general public expects in terms of audio restoration. This requires good judgment, and a great deal of experience.

The standard hearing test evaluates your ability to hear pure tones between 250 Hz to 8000 Hz, at 8 discrete frequencies. If you are performing forensics audio work, it is advisable to have you hearing response evaluated/plotted by a professional audiologist routinely.

Edison, Thomas Alva
Thomas A. Edison was the inventor of the phonograph in 1877 at his laboratory in Menlo Park, New Jersey. Edison also invented the Carbon and the Condenser Microphone, and the "Edison Effect" which is the principle behind the early rectification and amplification devices that were used to develop the field of modern electronics.

Electrical Recording
“Electrical Recording” is the term given to a process which was commercialized around 1925 for mastering records in which microphones and electrical signal amplification was utilized to supply the energy required to modulate the cutting head stylus of the recording lathe. Prior to the invention of electrical recording, the acoustic energy of the various sound sources in the recording studio was the only source of energy that modulated the cutting head stylus. Electrical recording allowed more of the subtlety and detail of music to be captured on the wax master.

Electron Tube
The predecessor to the modern transistor was the Electron Tube (also sometimes referred to as an electron "Valve"). Dr. Lee DeForest invented the device around 1906 and called it the “Audion”. He took a Fleming diode (a derivative of the Edison Effect light bulb - 1883), and installed a grid structure between its cathode and anode. He observed that small Voltage signals applied to the grid with respect to the cathode produced large changes in the devices plate current. This device became known as the "triode", having three active elements within it. Thus was born the key device that

became the foundation building block for the development of modern electronics, as we know it today. Electron tubes are basically amplification devices, which can be used in a myriad of applications such as oscillators, mixers, detectors, etc. The Diamond Cut Virtual Valve Amplifier (VVA) uses the measured characteristics of real electron tube triodes and pentodes in various amplifier and rectifier circuit models to produce a versatile array of "tube-warmth" effects.

Elliptical Stylus
The shape of the tip of certain phonograph record playing styli is elliptical which improves the high frequency response as compared to standard conical styli.

Engineering
The art of managing engines *(Merriam-Webster).*

Envelope
The sampled peak amplitude values of a .wav file as displayed in the DCArt10/DC F9 workspace when zoomed-out.

Expander
An Expander is a device that performs the opposite function of a Compressor. These devices increase the dynamic range of an audio signal source. When the process of compression is used in the recording or transmission process, and the process of expansion is used in the playback or reception process, the technique is known as companding. It is sometimes employed because it increases the signal-to-noise ratio of the analog recording or transmission process.

FFRR (Full Frequency Range Recording)
This is the equalization curve used on records marketed by London (Decca) Records. It claimed a frequency response of 50 to 14,000 Hz as early as the mid 1930's. Details can be found under “Equalization Curves” in the Charts, Graphs and Other Info section of this User’s Guide.

FIFO
First **I**n **F**irst **O**ut or FIFO is a term used to describe the data flow in one form of a digital buffer. The first data into a FIFO buffer is the first data to exit out the other end of the buffer.

FIR (Finite Impulse Response)
FIR is a digital, non-recursive method for creating filters which can produce a phase linear response characteristic. FIR filters are always stable and are used in the Diamond Cut Forensics Brick-Wall filter suite.

Fletcher-Munson Loudness Contours
The Fletcher-Munson Contours are graphs developed in the 1930’s that identify the human perception of equal loudness as a function of frequency. Simply stated, the human ear provides the most perceptible flat response at very high loudness levels. At low loudness levels, the response falls off dramatically at the low end of the audio spectrum, and to a lesser degree, at its upper end. The curves indicate the region between 1,000 to 5,000 Hz are the flattest and most independent of loudness. The flattest region lies between around 600 to 1,500 Hz.

FLAC (Free Lossless Audio Codec)
FLAC is an audio file format that uses linear predictive coding techniques and provides lossless file compression. It reduces an audio file size roughly by 50%. FLAC is supported by Diamond Cut and is found under the File\Save As feature. Its file extension is .flac.

Flutter
Flutter is a relatively rapid frequency modulation of the information on a recording due to rapid changes in the velocity of the record, tape, or the soundtrack of the source. Flutter is the rapid counterpart to Wow, occurring at a deviation rate in the range of 6 to 250 Hz. This distortion could have been introduced in the mastering process, or the playback process, or a combination of both. DCArt10/DC Forensics10 is not capable of correcting this sort of problem in a sound recording at this time.

Fractional Speed Mastering
Fractional speed mastering is the process of transferring a record at a slower speed to a new media, and then converting it to the proper speed at a later time. This has two potential benefits:

1. It allows persons who do not own 78 or 80-RPM turntables to make transfers of those types of records using their 45-RPM speed

2. It allows severely warped records to be transferred without skipping.

DCArt10/DC Forensics10 supports fractional speed mastering using the speed change filter. Presets will be found which accommodate various fractional speed-mastering situations.

Frequency Modulation (FM)
Frequency Modulation is a technique for producing a radio carrier containing an audio signal in which the frequency of the carrier is varied in proportion to the amplitude level of the audio modulating signal. With Frequency Modulation transmission, the rate at which the carrier frequency changes is proportional to the audio modulation frequency. FM carriers are ideally constant in amplitude.

Frequency Response
Frequency Response is the range of frequencies that a system will pass through without attenuation. The frequency response of audio equipment is generally specified with the upper and lower corner frequencies defined at the -3dB points. For most high performance audio system electronics, the frequency response will be at least as good as 20 Hz to 20 kHz +/-3dB. However, loudspeaker systems rarely are able to reproduce that same spectrum within the specified linearity band.

Formants
Spectral concentrations of acoustical energy found in human vocal patterns produce resonances referred to as formants which can be used in the field of voice identification. Formants having the lowest frequency are designated as F0 and the sequence proceeds in an ascending rank up to Fn (usually between F1 to F4 in total number aside from the fundamental, F0). Comparing the relative amplitudes of these formants between a known reference and unknown samples can help establish the probability of a voice print match. Your Diamond Cut Forensics Audio Laboratory Spectrogram in conjunction with the Voice ID function (Find Formants) can be useful for plotting formant trajectories. These data are exportable for use in other software programs (like Excel) which may be useful for statistical analysis purposes.

Fourier Transform
A Fourier transform is a set of mathematical relationships which allow complex waveforms to be resolved into a series of fundamental frequencies, plus a finite number of terms which describe the waveforms harmonics. Fourier transforms are said to allow signals in the time domain to be represented in the frequency domain. Certain mathematical manipulations are more easily performed in the frequency domain as compared to the time domain, and DCArt10/DC Forensics10 takes advantage of this characteristic. After the mathematical manipulations have been completed, the resultants are re-converted back into the time domain via Inverse Fourier Transforms in order to re-create the processed version of the original time domain waveforms. For example, this method is utilized in the Diamond Cut Continuous Noise Filter (CNF).

Full-Duplex
In the context of audio restoration, the term “full-duplex” refers to a sound card that is capable of performing input and output functions simultaneously. For example, an analog sound card that has full duplex capability will be able to take an analog signal and convert it into a digital signal, at the same time that it is converting a separate digital signal into an analog form. The Diamond Cut Live Feed-through mode requires a full-duplex card in order for it to operate.

Gain
Gain is the amplification effect of an electronic system that is often expressed in decibels (dB). For example, an amplifier that has a Voltage gain of 20 dB produces an output Voltage signal that is 10 times greater in amplitude compared to its input. Many special effects audio processors produce "unity" gain. This implies that its output Voltage will be equal to the input Voltage (X 1 gain). Unity gain allows many signal processors to be placed in cascade without concern that the last processor in the chain will become overloaded due to the amplification build-up through each previous processor in the chain.

In general:

Voltage Gain = Av = Vout / Vin

or -

Voltage Gain in dB = Av (dB) = 20 log Vout / Vin

Total System Gain in dB = Subsystem #1 Gain in dB + Subsystem #2 Gain in dB + Subsystem #N Gain in dB (when the subsystems are connected in a cascaded configuration).

Note: If the subsystem gains are not given in dB, (but given in multiplication factors) the total system gain is the product of the various subsystem gain values. For example, the total Gain = (Subsystem #1 Gain) X (Subsystem #2 Gain) X (Subsystem #N Gain).

The gain (A) of an electrical system can be given in terms of any of the following:

1. Voltage: (Av = Vout / Vin) --- (Voltage Gain in dB = 20 log Vout / Vin)
2. Current: (Ai = Iout / Iin) --- (Current Gain in dB = 20 log Iout / Iin)
3. Power: (Ap = Pout / Pin) --- (Power Gain in dB = 10 log Pout / Pin)

Generational Loss
Each time an audio signal is transferred from one medium to another, it will suffer some degree of "generational loss." These losses include noise buildup, distortion, phase jitter, quantization errors, etc. In analog systems, generation loss is much more of a significant factor in signal degradation compared to that which will be found in digital systems. In practical terms, analog signal transfers should be minimized in audio restoration work. The best results are produced if the analog to digital conversion process is performed only once. Ideally, the only analog process would be the original A-D transfer. Once in the digital domain, all processing, including the final transfer to DAT or CD-R, can be performed by your computer and the DCArt10/DC Forensics10 program. The only future conversion back to the analog world would occur during the playback process of the CD or the DAT through your audio system.

Graphic Equalizer
A Graphic Equalizer is a signal processor in which the audio band is divided into smaller spectral bands (portions). Each spectral band can be adjusted in terms of either the gain or the attenuation of the frequencies that fall within that band. Most Graphic Equalizer are Octave based, and contain about 10 bands. However, some are 1 /3 Octave based and have about 30 bands. Octave based graphic equalizers (including the one contained within the DCArt10/DC Forensics10 application) typically break the audio spectrum down into bands with the following center frequency values:

31 Hz, 62 Hz, 125 Hz, 250 Hz, 500 Hz,
1 kHz, 2 kHz, 4 kHz, 8 kHz, 16 kHz

Your DCArt10 software program also includes 20, and 30 Band Graphic Equalizers. The DC Forensics version includes a 32,000 band EQ called the Spectral Filter.

Grey Noise
Grey Noise is a form of random noise which is related to how humans perceive sound level as a function of loudness. All frequencies are perceived having equal loudness when the source is Grey Noise) White Noise can be converted to Grey Noise at 0 dB SPL by applying White Noise to the multifilter preset by the same name. Also refer to Fletcher-Munson Loudness Contours

Ground Loop
A ground loop is a potentially detrimental conductive pathway formed when two or more points in an electronic system that are nominally at ground potential are connected by another conducting path. The term usually is employed when, by improper design or by accident, unwanted noisy signals are generated in the common return line of relatively low-level (audio) signal circuits by the return noise currents or by magnetic fields generated by relatively high-powered circuits or components.

Harmonic Exciter
A Harmonic Exciter is an electronic device or algorithm, which synthesizes odd and/or even harmonics of the upper end of the audio spectrum presented to it, and then re-inserts them back into the signal path. This device or system will "liven-up" olde recordings in which the upper musical registers are missing due to generational

losses or lack of response to begin with. It can also be used to enhance vocals, or stringed instrument recordings. The Exciter is found under the Virtual Valve Amplifier (VVA) system located under the Effects Menu. It uses real models of Electron Tube rectifier and amplifier circuits to accomplish its synthesis.

Harmonics
Harmonics are the odd and even multiples of a fundamental frequency. In music, it is the distribution of these harmonics that provides the characteristic (or timbre) that gives each musical instrument or human voice a unique sound.

Harmonic Distortion
Harmonic Distortion results from the interaction of a non-linear transfer function of a system on a signal. The non-linearity of the system creates undesirable harmonic products (except in rock and roll) that modify the sound of the original signal. Devices like transistors, vacuum tubes, microphones, phonograph cartridges, loudspeakers, and A to D converters all have non-linearities to some degree. In some cases, feedback is used to correct for non-linearity and in other cases using the device only in a very limited portion of its total dynamic range is the method used for minimizing the production of harmonic distortion. For information on the measurement of this parameter, see THD (Total Harmonic Distortion) which can be measured using the Diamond Cut Spectrum Analyzer.
DCArt10/DC Forensics10 can produce signal distortion when one of the algorithms attempts to drive the system to full scale or beyond. It is therefore necessary to be careful when applying the Gain Change or the Graphic Equalizer algorithms, both of which can increase the gain of the system causing signals to exceed the programs dynamic range. The distortion produced as a by-product of this mechanism is called clipping.

Harmonic Reject Filter
Please refer to "Comb" or "Multiple Notch Filter.

Helmholtz Resonator
A Helmholtz resonator is the most basic resonant system found commonly in musical instruments and consists of an enclosed volume of air and an opening (or aperture). It is the acoustical analogue to the electrical LC resonant tank circuit.

Hertz (Hz)
Hertz is a unit for the measurement of frequency. 1 Hertz = 1 cycle per second (CPS).

Heterodyne
When two frequencies mix with one another through a non-linear system, sum and difference signals are produced. These signals are referred to as heterodynes and are also sometimes referred to as "beat frequencies". In 1918, Major Edwin Armstrong patented the use of heterodyning radio signals against a local oscillator into a single intermediate frequency (IF) to create greatly improved radio receivers. Also, early audio oscillators constructed before 1935 used the heterodyne technique for producing output signals in the audio range by "beating" two RF (radio frequency) oscillators against one another. They were referred to as Beat Frequency Oscillators (BFOs,) and were eventually replaced with a gain-stabilized form of the Wien Bridge oscillator. Spurious heterodyne signals are produced on the various AM radio bands due to adjacent channel interference. In the US, these signal are 10 kHz, and in Europe, they are 9 kHz. The DCArt10/DC Forensics10 notch filter can be used to remove them.

Hill and Dale
See Vertical Cut

High-pass Filter
A filter that attenuates all frequencies that fall below its corner frequency. The degree of attenuation of a signal outside of the filters pass band depends on the frequency of interest, and the corner frequency and slope (order) of the High-pass-filter. This type of filter is often used to reduce rumble, muddy bass and wind noise on a recording.

Hiss
Hiss is random noise located at the top end of the audio frequency spectrum. Generally this is considered to be the random noise that is heard above 5 kHz. A good example of "hiss" is the sound you will hear if you tune a FM tuner to the top or bottom end of the band where there are no stations transmitting with the "mute" button

disabled. (This is a form of limited bandwidth "white noise".)

Hum

Hum is noise introduced into a recording or sound system that is harmonically related to the power line frequency. In the US, this will be 60 Hz and in Europe, this will be 50 Hz., and in both cases, it will include harmonics of the line frequency. The most common "hum" frequencies are the fundamental (usually due to ground loops) and /or its second harmonic (due to defective power supply filter capacitors in electronic equipment). To attenuate Hum on a recording, use the DCArt10/DC Forensics10 Notch filter set to either 50 or 60 Hz, depending on the hum frequency. Start with a bandwidth setting of around 0.2 Octave. Adjust the bandwidth to the minimum value required to effectively attenuate the Hum. This will minimize the Notch filters effect on all other frequencies. Hum on cockpit voice recordings will be at 400 Hz (and harmonics thereof).

Human Voice Frequency Range

See "Voice Frequency Range"

IIR

IIR is the acronym for "Infinite Impulse Response" which is a form of recursive filter. It employs, in the simplest example, a simulation of at least one resistor and one capacitor in a circuit, which are represented by a pair of recursive equations. The types of filters employing IIR techniques in the "Filter" menu are such functions as the Low-pass, High-pass, Band-pass, Notch, and Slot filters as well as the Graphic and Paragraphic Equalizers.

Impedance

(Z)

Impedance is the total opposition, including resistance and reactance, which a circuit element(s) offers to the flow of an alternating current, measured in Ohms.

Z = ((R^2) + (Xc^2) + (Xl^2)) ^ ½

Wherein:

Z = Impedance in Ohms

R = Resistance in Ohms

Xc = Capacitive Reactance in Ohms

Xl = Inductive Reactance in Ohms

Some standard Input and Output Impedance values that you will encounter are as follows:

1. 1 Ohm - The basic unit of measurement for Electrical Resistance, Impedance, or Reactance
2. 2 Ohms (sound re-enforcement systems), 3.2 Ohms (antique audio), 4, 8, 16, and 32, Ohms, (standard Loudspeakers)
3. 8 Ohms – The most common loudspeaker impedance found in the US in 2014.
4. 32 Ohms – Standard earbud impedance.
5. 50 Ohms - Standard Unbalanced Co-Axial impedance for RF signal transmission
6. 75 Ohms - Standard Unbalanced Co-Axial impedance for Television and FM signal transmission
7. 100 Ohms – Output resistance of modern op amp based line level audio equipment
8. 300 Ohms - Standard Balanced impedance for Television and FM signal transmission
9. 377 Ohms - Impedance of Free Space
10. 500 Ohms - Standard Balanced Microphone impedance
11. 600 Ohms - Standard Telephone Exchange Audio line impedance
12. 10,000 Ohms – Input resistance of modern Line Level Op Amp based audio equipment.
13. 2,000 Ohms - Antique Audio (headphones & 1920's vintage horn loudspeakers)
14. 20,000 Ohms - Common single ended input impedance found on vintage Professional Audio Equipment
15. 47,000 Ohms - Common Magnetic Phono Cartridge Loading Impedance
16. 50,000 Ohms - Standard Unbalanced High Impedance Microphone Impedance.

17. 100,000 Ohms - Common Input Impedance on Audio Equipment
18. 1 Meg Ohms - De-Facto Standard, Oscilloscope Input Impedance
19. 10 Meg Ohms - De-Facto Standard, True RMS Voltmeter Input Impedance

Impulse
Mathematically, an impulse function is an event of infinite amplitude, and infinitesimal time duration. In DCArt10/DC Forensics10 terms, an impulse is a transient that begins and ends within somewhere between 50 μS to several mSec, with amplitudes which are generally higher than the rectified average or RMS program material in a .wav file.

Inductance
(L)
The inductance of a circuit component (most often a coil) is the rate of increase in magnetic linkage with an increase of current. The unit of measurement of inductance is the Henry which corresponds to a rate of linkage increase of 10 ^ 8 Maxwell-turns or one Weber-turn per Ampere of current. Energy is stored (but not dissipated) in theoretically ideal inductors. The principle of inductance is a strong element in the operation of electronic transducers such as loudspeakers, magnetic phono cartridges, dynamic microphones, and transformers. Resonant circuits can be created utilizing a combination of capacitors and inductors. The basic resonant frequency of such a circuit is given by Fr = 1 / 2 pi (L x C) ^1/2. This principle can be used to create narrow Band-pass and notch filters.
The unit of measurement of inductance = H (Henry)
Note: pi = 3.141592654 (approximately)

.ini files
This is where the initialization constants, factory presets, and user presets were stored by the software the earliest versions of the software. Because of the 64 Kbytes total .ini file length limitation; a special presets directory is presently used for this function allowing a much larger number of total presets. The old .ini file extension used to be found in the Windows directory. DCArt10/DC Forensics10 uses the system registry for all other settings storage.

Instantaneous Sound Recordings
Instantaneous sound recordings are ones in which playback can immediately follow the recording process. The first sound recording made by Edison was “Instantaneous”. The following is a brief history of this form of recording technology:

- Wax Cylinders (1877 forward)
- Aluminum Discs (mid 1920s)
- Pre-Grooved Plastic Discs (early 1930s)
- Acetate covered Aluminum Discs (mid 1930s)
- Acetate covered Glass Discs (early 1940s)
- Magnetic Wire (mid 1940s)
- Magnetic Analog Tape (late 1940s forward)
- Digital Audio Tape (mid 1970s forward)
- CD/R’s (early 1990s forward)
- .mp3 recorders

Integrator (∫)
An Integrator is a device, system or process which evaluates the area under a curve of one function vs. another. It is the inverse process of Differentiation (also refer to “Differentiator”). The Diamond Cut Integrator is found as one of the presets under the Low Pass Filter.

Inter-modulation (Distortion)
Inter-modulation distortion is a very specific type of distortion that results from the amplitude modulating effect which one frequency has on another in a non-linear system. This form of amplitude modulation results when one frequency appears as the “carrier” and the other as the “modulating” signal. Since IM distortion is due to non-linearity’s in the transfer function of a system creating an AM effect, this results in the production of frequency components that are equal to the sums and differences of integral multiples of the components of the original complex wave. A two-tone test source is used to measure this parameter, typically consisting of (approximately) a 60 Hz test signal added to a 7,000 Hz test signal feeding the device or system under test. The low frequency signal is applied to the system being tested with an amplitude value much higher than the upper test frequency, typically by a factor of four or more to one. Using 60 and 7000 Hz test signals as an

example, vestiges are measured at 6,940 and 7,060 Hz and divided by the value of the 7,000 Hz fundamental in order to determine the level of Inter-modulation Distortion.

Inertia
The property by which matter that is at rest will tend to remain at rest, and matter that is in motion will tend to remain in motion (in the absence of friction).

I/O
Input / Output refers to the ports into which electronic signals are fed to an electronic device and the ports from which electronic signals are derived from an electronic device. DCArt10/DC Forensics10 allows you to choose between several I/O ports, provided you have the sound cards to support the feature.

IPS (Inches per Second)
The linear velocity of magnetic tape moving past a recording or playback head is referred to in terms of its IPS (inches per second) value. For a table of common tape deck speeds, please refer to our ***Charts, Graphs and Other Useful Info*** page.

kHz (kilo Hertz)
The unit used in the measurement of frequency equal to 1000 Hertz. In earlier times, this term was Kilocycles (per second).

kOhms
The unit used in the measurement of electrical resistance equal to 1000 Ohms.

Latency
Latency is the delay time encountered when operating in Preview mode or in "Live" feed-through mode. Maximizing the speed of your computer system minimizes latency.

Lateral Cut
A record recording technique in which the groove modulation (undulations) occurs in a side-to-side direction, as opposed to up and down. This technique was popularized by Emile Berliner with his “Victrola” phonograph.

Launch
Starting a program. This is accomplished by double clicking on the appropriate program Icon (usually found on your desktop).

Least Significant Bit (LSB)
The smallest quantified increment which an Analog to Digital or Digital to Analog Converter can resolve an analog Voltage or current. (LSB's are sometimes referred to as "counts.) In DCArt10/DC Forensics10, this value is 1 part in 65,536 (or 1 part in + / - 32,768), for 16 bit system applications. For 24 bit system applications, this value increases to 1 part in 16,777,216 (or 1 part in + / - 8,388,608). For other sampling depths (values of resolution) use the formulae:

+ / - LSB's = (2^(SD)) / 2

wherein SD = Sample Depth (or resolution) in bits

Limiter
A Limiter is an electronic circuit or system consisting of non-linear elements that will not allow signals above a certain threshold value to pass through to its output. An upward compressor will produce this effect when its ratio is set to a high value. The Dynamics Processor can be used as a signal limiter when used in compressor mode when high values of "ratio" are selected. It also includes “Limiter” presets to make setup easier for the user.

Line Level Signal
A Line Level analog signal is one that develops -10 dBv nominally without clipping and is usually associated with consumer grade audio equipment input and output ports. This translates into 0.316 Vrms or 0.894 Volts pk-pk (based on a sine wave). They are usually unbalanced (often using RCA type interconnections).

Note – Pro-level analog signals are often called out in terms of dBu and are at values of +4 dBu which is 1.228 Vrms or 3.47 Volts pk-pk (based on a sine wave). They are usually balanced (XLR or TRS type interconnections).

Lissajous Figures
When two sine waves are displayed on an X-Y display, with one applied to the X-axis and the other to the Y-axis, the interacting vectors of the two waveforms are displayed. These waveforms are referred to as Lissajous figures. Signals having the same frequency but of differing phase (other than 180 degrees) will form elliptical patterns,

the phase of which can be calculated from the intercepts of the waveform with the display axis. This technique if often used to adjust the azimuth of tape recorder recording and playback heads. A properly aligned tape head will produce no ellipse, but only a 45-degree line with a positive slope. This can be somewhat compensated for using the Time Offset feature found in the File Conversions Filter in conjunction with the X-Y plotter found under the View menu.

Loudness
The loudness of a sound is the perceived magnitude due to the auditory sensation produced by an acoustic signal. It is a function of frequency and signal amplitude. A 50 Hz tone requires sound field intensity 250,000 times larger than a tone at a reference of 1,000 Hz to achieve the minimal level of perception. The unit of measurement for loudness is the Sone. See Sone for more details.

Low-pass Filter
A Low-pass Filter is one that attenuates all frequencies that fall above its corner frequency. The degree of attenuation of a signal outside of the filters pass-band depends on the frequency of interest, the corner frequency, and slope (order) of the low-pass filter. This type of filter is often used to reduce the hiss on a recording. However, low-pass filters will also attenuate the "highs" on a recording at the same time, which make them generally undesirable for this application.

Magnetic Phono Cartridge
A magnetic phono cartridge is a device for converting the mechanical motion of a record stylus into electrical signals utilizing the properties of magnetic circuits. There are three types of magnetic phono cartridges. They are:

1. Variable Reluctance (early magnetic cartridges)
2. Moving Magnet (the most commonly used)
3. Moving Coil (quite expensive and having costly stylus replacement)*

* The output impedance of moving coil (MC) cartridges are in the 10 to 100 Ohm range. Therefore, they require special matching transformers or pre-pre-amplifiers in order to be able to drive a conventional magnetic cartridge input on an audio pre-amplifier.

Mbyte
One million Bytes. (Sometimes 1024 kBytes for disks)

Median
The median value of a series of numbers is the number that resides in the center of the sorted string. For example, in the series of numbers 2, 4, 7, 3, 8, 0, 7, 1, 9, the median value is **4.** (Sort = 0,1,2,3, **4** ,7,7,8,9)

Microphone
A microphone is a device (or transducer) that converts acoustical signals to electrical signals. There are many variants. In order of development starting with the earliest they are as follows: Dynamic (Bell Telephone), Carbon Button (Edison), Condenser, Ribbon (a special form of a Dynamic type), Piezoelectric, and Electret (a special form of a Condenser type). Any of these technologies can be created with various polar pickup patterns (such as omni-directional, cardioid, unidirectional and others).

Milliseconds
The unit used in the measurement of time equal to 1/1000 of a second.

Mils
The unit used in the measurement of distance equal to 1/1000 of an inch. The diameter of phonograph styli are generally specified in "mils". (If the stylus is elliptical in shape, the larger of the two dimensions is generally given.)

Modulation
An electronic process in which one source modifies the characteristics of another signal source. For example, an audio signal may be used to Amplitude, Frequency, or Phase Modulate a sine wave signal (called a carrier). The result would be an Amplitude Modulated carrier in the first case (AM). In the second case, the result would be a Frequency Modulated carrier (FM). In the last case, the result would be a Phase Modulation (PM) carrier. These are techniques used for transmitting radio, television, and data. Sometimes, in audio, one refers to the undulations on a record as record groove "modulation".

Monophonic
An audio signal or a .wav file that contains only one unique channel of aural information is sometimes referred to as being monophonic.

MOSFET (Metal Oxide Silicon Field Effect Transistor)
A MOSFET is a three terminal device in the transistor family. The MOSFET is a three terminal solid state device having a gate (g) source (s) and a drain (d). They come in two configurations, N and P channel which relates to their polarity. They are used in amplification, oscillator and switching circuits They are often characterized in terms of their trans-conductance (amplification factor) mu (μ) such that the current flowing in the Drain is controlled by a Voltage applied between its Gate and Source.

Mu (μ)
The small signal amplification factor that a device exhibits in a circuit, often associated with Electron Tubes (and Junction **F**ield **E**ffect **T**ransistors or MOSFETs).

Voltage Amplification = $\mu \times Rl / (Rl + Rp)$
Where u (mu) = Tube Amplification Factor
Rl = Plate Load Resistance in Ohms
Rp = Plate Resistance in Ohms

Mu-law (μ-law) Compression
This is a form of data compression primarily used in telephonic applications. It takes advantage of the logarithmic nature of the sense of human hearing to accomplish its task effectively. It compresses a 16-bit signal into 8 bits using logarithmic signal mapping, producing a 2:1 compression ratio while maintaining 13 bits of dynamic range. The bottom 3 LBS's of precision are dropped. Typically, this is used in 8 kHz Monophonic formats. DCArt10 supports this format.

Multi-path Distortion
Multi-path distortion is a phenomenon that can occur during FM broadcast reception. It occurs when the receiving antennae picks up two (or more) signals from the same transmitter. This dual pickup consists of the direct signal from the transmitter (usually a line of sight trajectory) and a second parasitic signal arriving at the antenna some time later. The second signal is a reflected signal off of a mountain, building or other object, and arrives at the antennae sometime after the main signal had arrived. The time shift between the main signal and the reflected signal creates phase distortion of the de-modulated audio signal when these two signal mix together. This phase distortion manifests itself in the last two octaves of the audio spectrum and sounds like "slurring" of the pronunciation of the letter "s" and general harshness. It will sound worse on a stereo broadcast than on a monophonic one. There are several cures for this problem. Purchase a directional antennae (one with a high front to back ratio) and install it as high as possible, aiming it towards the transmitter of interest. Secondarily, you can minimize the problem by switching over to monophonic during a particularly distorted broadcast. And lastly, when all else fails, you can reduce the distortion by utilizing the "de-esser" found in the Dynamics Processor or apply one of the appropriate presets found in the Diamond Cut Channel Blender effect.

Multiple Notch Filter
This is the term used in the DCArt10/DC Forensics10 program used to describe a comb filter. A comb filter is a wave reject filter whose frequency rejection spectrum consists of a number of equidistant elements resembling the tines of a comb. This filter is useful for getting rid of "Hum" type noise containing more than just the line frequency fundamental component. This type of noise is line frequency related noise and sometimes described as "Buzz." This results from the interaction of non-linear systems with the finite output impedance presented by the power line sine wave Voltage waveform, adding harmonics to the same. Buzz can also be introduced into and audio system through non-sinusoidal current waveforms producing "H" fields which couple into noise sensitive loop areas (or ground loops) in audio systems. The Diamond Cut Harmonic reject filter is one example of a Multiple Notch filter.

Musical Scale
There are two relatively common musical scales. They are the "Scale of Just Intonation", and the "Scale of Equal Temperament". The Scale of Just Intonation requires at least 30 discrete frequencies for each octave, making it relatively impractical to build musical instruments with fixed

tones to play in the Just Scale. Therefore, the scale of Equal Temperament containing only 12 notes per octave is the one in general use.

For a chart which displays the frequencies of four octaves of the tempered scale, simply turn to our ***Charts, Graphs, and Other Useful Info*** section.

NAB Equalization Curve
(**N**ational **A**ssociation of **B**roadcasters)
The NAB Curve is a set of equalization frequency response contours, which are used by manufacturers of analog tape recorders to compensate for the inductive nature of a tape head. The equalization time constants specified depend on tape speed. One pair of time constants are specified for 1 7/8 ips (inches per second) and 3 3/4 ips. Another pair of time constants are specified for 7 1/2 ips and 15 ips. The low frequency breakpoint for all speeds is 50 Hz. The high frequency breakpoint for 1 7/8 and 3 3/4 ips is specified as 1770 Hz. The high frequency breakpoint for 7 1/2 and 15 ips is specified as 3180 Hz. These curves can be found as presets within the Diamond Cut Paragraphic Equalizer. Additional curves can be found on the presets sharing section of the Diamond Cut User's forum.

Neper (Napier) (Np)
Nepers are units of ratios of measurement like the dB, except using base e (~2.71828183) rather than base 10. For details, please refer to "dB" in this glossary. To convert between Nepers and dB, use the following relationships:

1 Np = dB x 0.115129255

or

1 dB = Np x 8.685889638

Noise
Noise are unwanted disturbances superimposed upon a useful signal that tends to obscure its information content. Also, refer to Signal-to-Noise ratio for more information.

Noise Gate
A noise gate is an electronic device, which turns off a signal path when an input signal is below a predetermined threshold value. The Dynamics Processor produces a noise gate effect when you check the Expander/Gate function. You must set the ratio to the highest number for the best noise gate effect.

Notch Distortion (crossover distortion)
A discontinuity in a signal waveform sometimes produced by power amplifiers is referred to as "Notch Distortion". It is always found in class B Amplifiers and sometimes found in inadequately biased class AB Amplifiers. It is usually associated with audio power amplifiers. High levels of "Notch" distortion results in a raspy sound at low signal output levels.

Notch Filter
A notch filter is one which attenuates all frequencies close to the center frequency of the filter setting. The degree of attenuation and the range of frequencies which are attenuated by this filter are determined by the filters Q or bandwidth. This type of filter is often used to minimize hum or acoustic feedback from a recording. This type of filter is sometimes referred to as a "band reject filter."

Octave
An octave is a group of eight musical notes and also a doubling of frequency. For example, the range of frequencies from 440 Hz to 880 Hz is 1 octave. The next octave will end at 1,760 Hz. Note that in two octaves, the frequency has increased by a factor of four.

Offset
A fixed or DC value of Voltage or current added into a circuit to shift the quiescent operating point of a device or display. Offset is used in DCArt10/DC Forensics10 to allow small features to be seen in a signal when the detail exists towards the top or bottom of the signal workspace display area. DC offset can be removed from a .wav file by applying the high pass filter set to 10 Hz, Butterworth response, and 6 dB/octave.

Ohm (Ω)
(R) (or the Greek Letter Omega□)
The Ohm is a unit of electrical resistance in which a potential difference in a circuit of 1 Volt produces a current flow of 1 Ampere.

Ohms Law
V = I x R

wherein:

V = Voltage in Volts,

I = current in Amperes,

R = resistance (in Ohms)

Over-modulation
When an audio signal is applied to an audio device, which is greater than the device can handle in a linear transfer manner, this creates a condition of "over-modulation". It results in a distorted sound in the output of the device being over modulated. Sometimes, this condition is referred to as "clipping," meaning that the amplification devices of an electronic system are either cutting-off or saturating due to overdrive.

Overtones
Overtones are multiples of a fundamental frequency. The Diamond Cut Overtone Synthesizer is capable of creating signals of this type (even multiples) at the upper end of the audio spectrum (above 6000 Hz). The Virtual Valve Amplifier (and its Exciter) are also capable of creating overtones (odds and evens) referenced to various fundamental frequencies.

Paragraphic Equalizer
This is a Diamond Cut invention in which a parametric equalizer is coupled to a graphic display of its transfer characteristic. It yields the best of both worlds produced by a graphic EQ and a parametric EQ.

Parametric Equalizer
A variable electronic filter in which the following three parameters may be adjusted on each parametric channel:

1. Frequency
2. Level (attenuation or amplification)
3. Bandwidth

Parametric equalizers are usually equipped with several parametric channels, which can all be used simultaneously or each one can be individually bypassed. DCArt10/DC Forensics10 includes a 10 band Paragraphic equalizer, which is a combination of a parametric and a graphic equalizer.

Pathé
Path**é** Freres Phonograph Company was a European based record and phonograph company, who utilized a somewhat unique groove modulation technique. Their method produced a vertical stylus displacement (like Edison Hill and Dale Diamond Discs and Cylinders) however; this was accomplished by a different mechanism. The groove on these recordings is "width" modulated, and so when a conical stylus interacts with these groove width modulations, a vertical displacement is thereby produced. If you are transferring a Path**é** 78 rpm recording with a stereophonic pickup cartridge, you will need to use the DCArt10/DC Forensics10 Mono (L - R) file conversion algorithm.

Pentode
A Pentode is an electron tube (or valve) containing five elements. They include a cathode, anode, control grid, screen grid and a suppressor grid (or beam deflector electrode which replaces the suppressor grid in beam power pentode configurations). They are most commonly used in audio power amplifiers, but are sometimes found in microphone pre-amplifiers. Typical beam power pentodes listed in ascending power levels include types 6BQ5/EL84, 6L6GC, 5881, 7591, KT-66, 6CA7/EL34, KT-88, and 6550. Some pentode tube types are available in the Diamond Cut Virtual Valve Amplifier (VVA).

Phantom Microphone Power
(Sometimes referred to as P48)
High performance microphones (such as large diaphragm condenser types) require an external source of power to drive their internal circuitry. The Voltage of this power source is generally between 12 to 48 VDC (most commonly 48 VDC). The negative side of this power source is connected to pin 1 (the shield side) of a 3 pin XLR style connector. The positive side of this source is connected (via two ~ 6800 **Ω** resistors) to pins 2 and 3 on the connector with one resistor located in series with each leg at the source end.

Phase Inversion
Phase inversion is the phenomena that occurs when one of two signals has become

180 degrees phase shifted with respect to the other. This sometimes accidentally occurrs on vinyl stereo recordings because the input leads to one of the two cutting lathe driver heads became "swapped" in location. This can be corrected by using the File Converter, using the Left or Right Phase-Invert feature.

Phon
The Phon is a measurement of the perceived loudness of a sound which takes into account the non-flat frequency response of the human sense of hearing. The perceived loudness of a sound in Phons is equal to the sound intensity in dBs of an equally loud 1 kHz pure tone signal; 1 dB SPL @ 1 kHz = 1 Phon. Also, refer to the glossary topic "Fletcher-Munson Loudness Contours".

Pi (π)
Pi (Greek Letter) is the symbol that relates the ratio of the circumference to the diameter of a circle.

$\pi = C / D$

wherein:

C = Circumference of a Circle

D = Diameter of a Circle

π =

3.1415926535897932384626433832795029

(approximately because it is an irrational {non-repeating} number.)

Pink Noise
Pink Noise is random noise, which is characterized as containing equal energy per unit octave. When viewed on an octave based spectrum analyzer, it will produce a flat horizontal line on the display. Pink Noise is useful for characterizing the frequency response of electronic systems and for analyzing room acoustic transmittance and resonance. Pink noise can be created through a two-step process using DCArt10 or DC Forensics10. First, create white (random) noise with the Makes Waves generator function. Next, process the signal through the Multifilter using the factory preset labeled "White to Pink noise converter, 20 kHz." You can also convert Pink Noise to White Noise by applying the Multifilter preset called "Pink to White Noise Converter" to a Pink Noise File.

.pkf files
These are "peak files" (.pkf) which represent the .wav files for display purposes in the source and destination windows. They are a small subset of the .wav file consisting of one sample for every 200 .wav file samples. The data point selected is the peak value of the .wav file found within that 200 sample window. Each .wav file is stored with an attendant .pkf file that is created by the software at the time when the file is recorded or modified or when the "rebuild peak file" command is used.

Potentiometer
A potentiometer is a three-terminal passive electronic device consisting of a resistance element and a wiper arm. They can be of the slider type or the rotary type and come in various resistance tapers. They produce a variable Voltage Divider effect on a signal in a circuit. They are used in the analog portions of audio circuits to vary the gain of a particular circuit (like volume or tone controls). The most common application of this component is as a Volume Control, but they are also found in Graphic Equalizers and other analog signal processing systems. The term "Pot it up" comes from this component (on mixing boards). Pot it up is an early radio term for "turn it up" or "make it louder".

Power
Power is the time rate for the transfer of energy in any system. In other words, Power = Energy / time. In electrical terms, power is given in Watts and has the following relationships to Voltage, Current, and Resistance (given a sine wave):

$P = V \times I (\cos \theta)$

wherein:

P = Power in Watts

V = Voltage in Volts

I = Current in Amperes

θ = the displacement (phase) angle between the Voltage and Current Waveforms (assuming that they are both sine waves) also (for purely resistive situations),

$P = (I ^ 2) R$

and

$P = (E ^2) / R$
wherein:

R = Resistance in Ohms

Power Amplifier
(Power Amp)
A power amplifier is a device that provides power amplification of an audio signal. Generally, this is the device that is used to drive a loudspeaker, the cutting head of a record lathe, or an audio transmission line, and is the final stage of amplification in an audio system. Audio power amplifiers generally develop somewhere between 10 to 1000 Watts of output power, depending on make and model (although shake table audio amplifiers and AM radio transmitter modulators can be found which produce well over 50,000 Watts).

To minimize power loss in the transmission process, and to maximize the systems dampening factor, it is important to minimize Voltage drops across loudspeaker distribution cables. Poor (low) dampening factor can produce an ill-defined (or resonant) bottom-end (bass). Long distances between your power amplifier and your speaker system will require larger diameter cables. To determine the correct cable for your application, refer to the Wire Table provided in the Charts, Graphs and Other Info section of this User's Guide.

Power Cepstrum
See "Cepstrum"

Pre-Amplifier
(Pre-amp)
A Preamplifier is a device that provides Voltage amplification of an audio signal. Sometimes these devices also include equalization networks and/or tone (bass, treble, loudness, etc.) controls. Pre-Amplifiers generally produce about 100 milliwatts (mW) of output power and require a power amplifier connected in cascade in order to be useful for driving loudspeakers or very long audio transmission lines.
Pre-Emphasis
Pre-emphasis is the intentional added amplification which is sometimes applied to the top end of the audio spectrum during a recording or radio transmission process in order to raise the signal level at high frequencies substantially above the noise (hiss) level of the system. This process is reversed during the reproduction process of the signal in order to recreate an overall flat frequency response. The result of this process is an improvement in the signal-to-noise ratio of the system. For example, the third specified time constant of 75 μSec associated with the RIAA equalization curve is pre-emphasis. Also, FM broadcast transmission utilizes a 75 μSec (or sometimes a 25 μSec) pre-emphasis to improve its signal-to-noise ratio. This process is reversed at your receiver (de-emphasis). The Paragraphic equalizer contains 75 μSec pre-emphasis and de-emphasis preset curves.

Presets
Most of the Diamond Cut filters and effects have a plethora of descriptive presets. Usually, the most efficient place to start when using a particular filter or effect would involve selecting one of the factory presets, and then tweaking the parameters to fine tune the system to your own personal taste. If you desire to keep a separate copy of your presets on external media, they can be found in the Diamond Cut directory under this pathways: * \DCForensics10Presets

Pure Tones (as related to Noise Laws)
A pure tone occurs when the noise level in any one Octave weighted frequency band exceeds those in an adjacent frequency band by 3 dB or more. The High Precision Analyzer found in the DC Forensics version has the characteristics needed to evaluate compliance with the various state and federal laws in this regard when used in conjunction with a calibrated / certified / NIST traceable microphone.

Q
(Quality Factor of Resonant Systems)
Q is the ratio of the reactance of a system (or filter) to its resistance (or losses). Q determines the systems bandpass width. Higher "Q" values producing sharper (narrower and more selective) responses.
Q = Reactance (@ system resonance) / Resistance.

Quefrency
Quefrency is the unit of measure associated with the horizontal (X) axis of a complex ceptstum graph. The units are not frequency or time in the standard sense of those terms. Rather it is the ratio of the sample rate divided by the number of samples that a peak persists. For example, if a signal persists for 50 samples and is sampled at 48,000 samples per second system, its quefrency will be 48000 / 50 = 960 Hz.

Quiescent Point
The Quiescent Point (or operating point) of an amplification device like an electron tube or a transistor, refers to the bias established on its linear portion of the transfer function curve when the device is "at rest" (i.e. no signal input applied). For example, the Virtual Valve Amplifier allows you to adjust the Quiescent (operating) point of Class A amplifiers anywhere from near cutoff to near saturation.

RAM (Random Access Memory)
RAM is a digital electronic device for storing binary information temporarily. RAM performance is generally characterized in terms of its size in Mbytes, and its access time in nanoseconds. Your computer will need a minimum of 2 GBytes of RAM to run the DCArt10/DC Forensics10 application correctly.

Real Time
A system that can process a signal and output the signal at the same rate at which it is being fed into the system is said to be a real-time processor. The DCArt10/DC Forensics10 algorithms can process signals in real-time or faster provided your platform is a 500 MHz Intel Pentium or higher.

Real Time Analyzer (RTA)
A Real Time Analyzer is a form of spectrum analyzer used for the analysis of audio signals. Unlike conventional spectrum analyzers, it does not use a single filter in a scanning mode to produce an Amplitude vs. Frequency display, which is a relatively slow process. Instead, it processes audio signals in parallel, so that all frequency bands are displayed simultaneously. Generally, RTA's have 31 bands (in 1 / 3 octave increments) covering the frequency spectrum from 20 Hz to 20 kHz. They usually come with a calibrated electret microphone and a built-in pink noise generator for making acoustical measurements. Your Diamond Cut Spectrum Analyzer has 1/3rd Octave-based RTA capability.

Rectified Voltage
This is a process wherein an alternating current signal is converted into a direct current, amplitude modulated envelope representation of the source signal. Often, some smoothing is applied to this signal with a set of time constants referred to as "attack" and "release". This signal is used in such devices as dynamic filters, companders, compressors, expanders, spectral enhancers, etc. and is digitally simulated in some of the DCArt10/DC Forensics10 algorithms.

Residue
The residue of a filtered signal is the algebraic difference between the filter output and its signal input. DCArt10/DC Forensics10 allows you to hear the "residue" of two of its filters by enabling the "Keep Residue" function. Several filters that include this feature are the Continuous Noise Filter and the Harmonic Reject Filter. This feature has been included because in some cases, it may be useful as an aid to hear what you are filtering out of the signal source. This is particularly useful when adjusting the Harmonic Reject Filter when attempting to remove "Hum" or "Buzz" from a recording.

Resistor (Resistance)
(R)
(Ohms) (Ω□)
A resistor is a basic electrical device that has electrical resistance, and is used to control the amount of current that flows in a circuit. The unit of measurement for a resistor is the Ohm.

$R = E / I$

wherein:

R = Resistance in Ohms

E = Voltage in Volts

and

I = Current in Amperes

Resonance (Electrical)
Resonance occurs in a system in which two elements are operated in quadrature (each element operating at ¼ the signals period) to produce a minimization or maximization of said signal.

Fr = 1/2 π (L x C)^1/2

wherein:

Fr = Resonant Frequency in Hertz (Hz)

L = Inductance in Henries

C = Capacitance in Farads

Resolution
Resolution is the minimum amplitude increment into which the A-D converter of a discrete time system can divide an analog signal. The resolution of DCArt10/DC Forensics10 is usually 16 bits, which is 1 part in 65,536. However, with the appropriate sound card, DCArt10/DC Forensics10 does support up to 24-bit I/O resolution. Resolution can also refer to the minimum "time slice" into which a sampled data system is divided or displayed.

Reverse RIAA Curve
DCArt10/DC Forensics10 is equipped with a family of reverse RIAA curves, allowing you to use a standard RIAA phonograph pre-amplifier to perform your mastering of old acoustical and 78-RPM recordings. A straight reverse RIAA curve is supplied for acoustical recordings, and a number of reverse RIAA curves with varying values of turnover frequency are supplied for electrically recorded 78s. These reverse curves can be found as several of the Paragraphic equalizer factory presets. An RIAA curve encoded recording can also be decoded using the Diamond Cut Virtual Phono Preamp.

Reverberation
The process whereby the acoustical reflections of a room or concert hall are reproduced artificially, with devices such as tapped delay lines working in conjunction with mixing and phase shifting devices or algorithms.

Rheostat
A rheostat is a two terminal device used to vary the resistance in an electrical circuit. A potentiometer can be used as a rheostat if only the wiper and one of the other terminals are used to vary the circuit behavior.

RIAA
Equalization Curve (**R**ecord **I**ndustry **A**ssociation of **A**merica)
The RIAA Curve is a non-linear frequency response contour which was utilized by some manufacturers of LP records after around 1950 and was made pretty much the industry standard by 1955. It’s purpose was to improve the S/N of the disc recordings. It specifies three R*C time constants to be used by playback pre-amplifiers in order to reverse the record cutter equalization. The three time constants and their corresponding breakpoint frequencies are as follows:

1. 3180 uS (50 Hz) (shelf or pullout frequency)
2. 318 uS (500 Hz) (turnover frequency)
3. 75 uS (2120 Hz) (roll-off frequency)

This curve can be de-coded using the Diamond Cut Virtual Phono Preamp if you are using a flat hardware preamplifier.

RIAA / IEC Equalization Curve
The RIAA / IEC equalization curve is defined in terms of the same time constants as the RIAA curve, with one additional time constant added of 7960 μS. This provides 3 dB of attenuation at 20 Hz rolling off at -6 dB / Octave thereafter. Below is a listing of all of the time constants associated with the RIAA / IEC Equalization Curve:

1. 7960 μS (20 Hz) (low frequency roll-off)
2. 3180 μS (50 Hz) (pull-out frequency)
3. 318 μS (500 Hz) (turnover frequency)
4. 75 μS (2120 Hz) (roll-off frequency)

Right Mouse Button
DCArt10/DC Forensics10 implements the following six functions with the right mouse button:

1. Play From Here
2. Preview From Here
3. Copy

4. Cut
5. Paste Over
6. Paste Insert
7. Mute
8. Zoom-In
9. Zoom-Out
10. Zoom-Out Full
11. Add a Marker
12. Label a Marker
13. Delete a Marker
14. Clear All Markers in Selected Area
15. Undo Last Edit
16. Select All
17. Snap Selection to Zero Crossing (Q)

Note: A pop-up menu will appear when using the right mouse button containing the above-mentioned items that you can select from when they are active. Items that are not active will be grayed out.

RMS (root mean square)

The RMS value of a Voltage or Current is the dc equivalent heating value of an alternating current waveform into a resistive load. For a series of sampled V(t) or I(t) data, it is (with V &/or I shown as X in this general expression):

Xrms = ((1/n (X1^2 + X2^2 + Xn^2))^1/2

Note: RMS power is a misnomer, as is commonly found associated with audio power amplifier specifications. What is really being presented is the average power calculated into a resistive load using a true RMS Voltmeter per the equation Pav = Vrms^2/R.

Roll-off
In the record industry, roll-off usually refers to the amount of attenuation in dB @ 10 kHz which must be applied during record playback in order to achieve a flat response on the high end of the audio spectrum. It is a form of de-emphasis. For example, the roll-off for the RIAA curve is –13.7 dB and –12 dB for the AES curve.

Roll-off Frequency
For a Low-pass filter or for an equalization curve (such as the RIAA curve), the upper cutoff frequency is sometimes referred to as the Roll-off Frequency.

RPM (Revolutions Per Minute)
Some common record speeds are 33.33 RPM for LPs, 45 RPM for records with the same name, 78.26* RPM for most so called electrically recorded lateral 78s (like Victor), 78.8 RPM for Edison Lateral's, 80 RPM for Edison Diamond Discs, and 160 RPM for most Edison Cylinder recordings. Additional speeds such as 16.66 RPM will occasionally be encountered. A comprehensive listing of record speeds can be found in the ***Charts, Graphs, and Other Useful Info*** section of this User's Guide.

Rumble
Rumble is a low frequency noise signal, typically below 50 Hz, which is often found on records. This phenomenon can be caused by seismic effects during the mastering process or during playback. On low quality or worn turntables or cutting lathes, it can also be produced by irregularities in the main thrust-bearing race. To attenuate turntable rumble using DCArt10/DC Forensics10, use the High-pass Filter. Start with settings of 30 Hz and 18 dB / Octave (or steeper), and adjust the frequency upwards or downwards until you are satisfied with the results. Acoustical recordings will benefit from settings as high as 125 Hz, depending on the material.

Sample Rate
The rate at which an analog signal is converted to discrete numbers by an A-D converter. For audio systems, sample rate is expressed in kHz. DCArt10/DC Forensics10 supports any number of standard sample rates including:

1. 11.025 kHz
2. 22.05 kHz
3. 44.1 kHz
4. 48.00 kHz
5. 88.2 kHz
6. 96.00 kHz
7. 192.00 kHz
8. Your entered choice of a numerical value from 100 Hz up to 210.00 kHz*

*If your sound card supports intermediate sampling rates, you can also enter the numeric value of any sample rate you desire, between 100 Hz (0.1 kHz) to 192 kHz for recording purposes.

Sampling Theorem

In a sampled data system (like the environment in which your DCArt10/DC Forensics10 program is operating), sampling theorem tells us that regularly spaced sampling must occur at least at the Nyquist rate, which is twice the frequency of the highest frequency signal or noise component that is expected to be resolvable by the system (without aliases). In other words, in a system expected to exhibit a frequency response up to 20 kHz, the minimum sample rate will have to be 40 kHz. Because it is impossible to construct an ideal Low-pass filter, the sampling rate will have to be somewhat larger than 2X the desired maximum frequency response value. In practice, a 44.1 kHz sampling rate is generally used in 20 kHz frequency response audio applications (although sometimes 48 kHz and 96 kHz are also used).

Shielded Cables
Shielded cables are special cables which are designed to minimize stray noise fields (particularly E fields) from entering an audio system through the interconnection wiring from component to component due to extraneous sources. Most often shielded cables are of the co-axial type so that loop area is also minimized, resulting in a minimization of "H" field pickup. However, some systems use a balanced pair of shielded wires which further minimizes pickup, provided the appropriate terminating transformers or differential amplifiers & line drivers are used on each end of the cable.

SID
Slewing-Induced Inter-modulation Distortion. This is similar to TIM. See TIM for details.

Signal-to-Noise Ratio
The ratio of signal-to-noise (Voltage, current, or acoustical sound pressure level) that is expressed in dB. Signal-to-Noise ratio in dB = 20 log (signal / noise).

SINAD (Signal, Noise And Distortion)
Essentially, SINAD is the reciprocal value of a THD measurement and is expressed in dB.

SINAD = 20 log(signal rms value + noise & distortion) / (rms value of noise & distortion)

Single-Ended Noise Reduction
Single-Ended Noise Reduction are the processes wherein noise is removed from an un-encoded audio signal. Algorithms like the Impulse, Continuous, Dynamic, Harmonic Reject, and Notch filters are examples of Single-Ended Noise Reduction tools.

Slew Rate
Slew rate is the maximum dv / dt (and sometimes the maximum di / dt) that an audio system component can react to. Parasitic Miller capacitances in operational amplifiers are generally the cause of this parametric limitation (collector to base or drain to gate in the case of transistors). When an audio component is confronted by a signal exceeding its maximum slew rate capability, the system reverts to slew rate-limited mode of operation. During this time interval, all audio information is lost resulting in severe distortion of the applied input signal.

Slope
In the context of DCArt10/DC Forensics10 and audio filter terminology, slope is the linear rate of change of amplitude vs. frequency of a filter past its corner frequency. This is expressed in dB / Octave or dB / Decade. 6 dB / Octave = 20 dB / Decade, 12 dB / Octave = 40 dB / Decade, etc.

Slot Filter
A slot filter is the compliment to the "notch" filter. It is a variable narrow Band-pass filter; capable of greater selectivity than a typical Band-pass filter. It is often used in Forensics work for isolating particular sounds like the ringing of a telephone on a recording in a crowded noisy bar situation, or anything similar. By allowing only a very narrow "slot" of frequencies through the system, one can observe the "slotted-band" with a much improved signal to noise ratio compared to the wideband signal. The DCArt10/DC Forensics10 slot filter can be found under the Notch filter and is activated by checking the appropriate box. Multiple slot filters can be run via the Multi-Filter. If the slots that are desired are harmonically related, you could use the Harmonic reject filter in "keep-residue" mode to produce up to 500 slots in one pass.

Sone

A Sone is a unit of measurement for sound loudness. A simple tone of a frequency of 1 kHz and at a level 40 decibels above a listener's threshold of perception represents a loudness of 1 Sone. A loudness of any sound which is judged by a listener to be “n” times greater than that of the 1 Sone tone is defined as “n” Sones.

Sound Level
Sound Level is a weighted sound pressure level obtained by the use of a metering system and any of three weighting standards as established in the American National Standard Specification for General Purpose Sound Level Meters. The reference pressure is 2 X 10 ^ -5 Newton per meter ^2. The two most common standards are the "A" and the "C" weighting factors. The "A" weighting characteristic responds mostly to frequencies in the area of the greatest sensitivity of the human ear in the 500 to 10,000 Hz range. The "C" weighting characteristic is nearly uniform over most of the audio spectrum. The 0 dB reference sound pressure level (SPL) for a sound level meter is 0.0002 microbars using a simple tone of 1000 Hz.

For a chart showing several common sound sources and their Acoustic Power and Sound Power levels (from 10 meters distance to the source), simply turn to the ***Charts, Graphs, and Other Useful Info*** section.

Sound Wave Velocity
Sound Wave Velocity in air as a function of temperature is given by the following:

c = 33,100 (1 + 0.00366t)^1/2

wherein:

c = Sound Wave Velocity in air in centimeters per second

and

t = temperature in degrees centigrade

Therefore at 70 degrees C, sound will travel at 37,098.6 centimeters per second, or around 830 miles per hour.

Sound Wavelength
The Wavelength of a sound wave is given by the following equation:

$\lambda = c / f$

wherein:

λ (lambda) = wavelength in centimeters

and

c = Sound Wave Velocity

and

f = frequency in Hz (cycles per second)

S/PDIF
S/PDIF (Sony / Philips Digital Interface Format) is a serial digital signal format used to connect digital audio devices together. It is a subset of the IEC 60958 (or AES/EBU) standard. It is electrically characterized as a coaxial system terminated with 75 Ohms on each end and typically using RCA (orange in color) or BNC connectors.

Speech Filter
A filter which typically has a Band-pass only between the frequencies of 300 Hz to 3 kHz, and which is used to improve the basic intelligibility of speech. Often, this type of filter uses slopes of -12 dB / Octave. This characteristic can be replicated with the Diamond Cut IIR based Band-pass filter. An alternative speech filter that is sometimes useful is called the Steep Slope Speech filter. Its response is 250 Hz to 3.5 kHz with a slope of 18 dB / Octave.

Speed
See RPM

Spectrogram
A Spectrogram is a system for presenting audio data in a graphical form and is a special case of a spectrum analyzer coupled to an oscilloscope. The horizontal (X) axis represents time, the vertical (Y) axis plots frequency and the gray scale brightness (or color) (Z axis) represents the signal intensity. Audio Spectrograms are used forensically for spectrographic voice recognition (sometimes referred to as voiceprints) and acoustical analysis applications. The software includes a Spectrogram feature, which is found under the Forensics menu.

Spectrum

Spectrum is band or range of frequencies as in the audio spectrum, the light spectrum, or the electromagnetic spectrum.

Spectrum Analyzer
A spectrum analyzer is a device for analyzing and displaying the Amplitude versus Frequency characteristic of a portion of a spectrum. They fall into two general categories:

1. Swept Band-pass Filter (a serial process of analysis).
2. Real Time Analyzer (a parallel process of analysis).

Spectral Enhancer
An electronic device or system which is used to expand the dynamic range of the upper and/or the lower octaves of the audio frequency spectrum, leaving the mid-band portion of the spectrum unprocessed. This has the effect of increasing the "definition" of a recording without continuously amplifying hiss and rumble which may be present on the source material. It is a form of dynamic filter which uses the principle of "upward expansion" to improve dynamic range. The Diamond Cut Dynamic Noise Filter contains a Spectral Enhancer mode of operation, which can be enabled.

Spectral Subtraction
Spectral subtraction is a method for reducing the unwanted noises associated with a recording using the amplitude elements of the frequency domain information contained in a .wav file. A noise fingerprint is taken, and this frequency domain information is then subtracted from the rest of the file and then re-converted back to the time domain. This technique is primarily used in forensics audio applications, and can be found as one of the CNF Mode options in the Continuous Noise filter.

Stroboscope
A stroboscope is a device which indicates the RPM speed of a turntable by creating an optical illusion of the slowing-down, freezing, or speeding-up of a pattern when illuminated by a pulsating light source operating at a known frequency. You can create your own stroboscope disc by dividing a circle evenly into black and white segments. Use the following formulae to calculate the number of segments required per 360 degrees (1 rotation of the disc) into which the disc must be marked:

60 Hz power systems: # of segments = 7,200 / RPM*
50 Hz power systems: # of segments = 6,000 / RPM*

For example, assume that you want to construct a strobe for use in the United States where the power system operates at 60 Hz in frequency. We want to design it "to freeze" at 78.2 RPM. 7,200 / 78.2 = 92.07.Round the number to 92 segments. Divide your circle into 92 evenly spaced segments, and voila, you have your strobe. Because of the rounding error, the strobe you constructed will be in error by 0.08 %. Your strobe will have to be used under a fluorescent or neon light connected to the power line in order to function. Incandescent lamps will not work because of the long thermal time constant of their filaments.
For a chart that will help you create your own strobe using common line frequencies and RPM values, go to our ***Charts, Graphs, and Other Useful Info*** section.

Note: DCArt10/DC Forensics10 provides two bitmaps that you can print and use as phonograph strobes covering the important speeds. These can be found in the Diamond Cut Directory at “Strobe50Hz.wmf” and “Strobe60Hz.wmf”.

Sub-harmonics
Sub-harmonics are fractional multiples of a fundamental frequency. The Diamond Cut Sub-harmonic synthesizer is capable of creating certain signals of this nature in the bass end (below 75 Hz) of the audio spectrum.

Square Wave
A square wave is a signal consisting of a fundamental frequency and the sum of all of the odd harmonic components of that fundamental frequency on the spectrum up to an infinite number of harmonics. An ideal square wave contains approximately 43% Total Harmonic Distortion (THD).

Styli
Styli are the devices used to transmit the mechanical undulations of a phonograph recording to its pickup or reproducer. Early styli were made from cactus needles or steel.

Later, Osmium (osmiridium), Sapphire and Diamond became the norm for phonographic styli use. A chart of styli (sizes and shapes) for various record types can be found under the ***Charts, Graphs, and Other Useful Info*** section of this User's Manual.

Tape Head
A tape head is an electromagnetic device used in a tape recorder to apply and read magnetic signals onto (and from) magnetic tape media. It consists of a coil mounted on a magnetic structure having a "gap" where the tape comes in contact. The tape head gap width in conjunction with the magnetic particle size on the tape media determines the frequency response of the system. This process follows Faradays Law of electromagnetic induction:

$\mathbf{V}_{\mathrm{head}}$ **= - N dø / dt**

wherein

$\mathbf{V}_{\mathrm{head}}$ = Tape Head Voltage

N = Number of turns of wire on the tape head structure

dø / dt = The time derivative of magnetic flux

Since dø / dt is proportional to and increases with frequency, this tape head Voltage signal increases at a rate of 6 dB / Octave and must be compensated for to prevent tape saturation. The process to provide this compensation is called tape equalization.

Tape Head Gap
The Tape head gap is the discontinuity in the magnetic pathway (or circuit) formed in the tape head structure which runs perpendicular the direction of the tape movement. Ideally, the playback head gap should be smaller than one half the Lambda (or wavelength as it relates to the speed of the tape) of the highest frequency signal to be recorded. The following is a listing of typical Tape Head Gaps by Tape Head type:

Playback Heads: 1 to 5 microns (micro-meters)
Recording Heads: 3 to 13 microns (micro-meters)
Erasure Heads: 25 to 150 microns (micro-meters)

Tape Recorder Speeds
See IPS

THD (Total Harmonic Distortion)
THD is a figure of merit as to how much non-linearity a system is imposing upon an audio conduit. The Spectrum Analyzer has the ability to measure the THD of an item under test when used in conjunction with either the Make Waves generator or an external hardware equivalent. %THD = Amplitude of the Harmonic Content of a signal / Amplitude of the Fundamental Component. When using hardware to make this measurement on very low distortion equipment, it is necessary to account for the Generator THD. Therefore, one must measure the test Generators THD at each test frequency as well as the System Measured THD (of the whole system). Then, one must apply the following equation:

Actual THD = (((System Measured THD)^2 – (Generator THD)^2))^1/2

THD + N (Total Harmonic Distortion plus Noise)
This is what is actually measured by most THD meters, including not only the harmonic distortion created by the device being tested, but its noise as well.

Thermal Noise (Floor)
Any electrical conductor produces a random noise Voltage as long as it exists at a temperature above 0 degrees K and/or has an electrical resistance greater than zero Ohms. The following formulae can be used to calculate the Root Mean Square value of the thermal noise Voltage of a terminating or source resistance:

E = (4RkT x Δf) ^1/2 or E = √(4RkT x Δf)

Wherein:

R = Resistive Component in Ohms (Ω)
k = Boltzmann's Constant = 1.38 x 10^-23 Joules / Kelvin (1 Joule = 1 Watt x Second)
T = Absolute or Thermodynamic Temperature in degrees Kelvin
Δf = Bandwidth of the system in Hertz (Hz)
E = Root Mean Square (RMS) Noise Voltage

T (in degrees Kelvin) = Temperature in Degrees C + 273.15

Example: Assume an audio mixer/microphone preamplifier is terminated with a 50K Ohm Resistance. It is operating at 40 degrees C internally, has a 60 dB Voltage Gain and exhibits a usable flat response from 20 Hz to 20 kHz. What is the RMS noise floor of the output of the mixer?

60 dB Voltage Gain = 1000 : 1 = 1000

Eout = (4RkT x Δf) ^1/2 x 60 dB = ((4 x 50,000) x (1.38 x 10 –23) x (30 + 273.15) x (20,000 – 20)) ^1/2 x (1000)

Eout = 0.004 Volts RMS = 4.0 Millivolts RMS

It is important to note that this is the Noise Floor for the defined system and can't be made to be any quieter than this number unless special techniques are employed. Cryogenic cooling techniques are sometimes used on first-stage amplification devices of specialized amplifiers to improve the noise performance of highly sensitive systems.

TIM (Transient Inter-modulation Distortion)
This is a form of distortion that occurs when a system enters into a "slew limit" mode of operation during a fast audio transient. The result is the loss of all sonic information during the "slew interval". It is usually the result of poorly designed amplifiers having slow error amplifiers or insufficient high frequency current drive provided to power amplifier output transistors. This type of distortion is sometimes referred to as SID (slewing induced inter-modulation distortion) or DIM (Dynamic inter-modulation distortion).

Time Constant
Time constants are exponential amplitude vs. time functions, which are realized with resistors and capacitors, or resistors and inductors and are often called "Tau"..

Tau = R x C for Resistor/Capacitor circuits

or

Tau = L / R for Resistor/Inductor circuits

wherein:

Tau = time constant in seconds

R = resistance in Ohms

C = capacitance in Farads

and

L = inductance in Henries

The relationship between a simple first order filters corner frequency (Fc) and time constant is as follows:

Fc = 1 / (2 x pi x Tau)

Note that the higher the value of time constant, the lower the corner frequency created. Some common time constants found in audio applications are as follows:

25 μSec - Dolby based FM de-emphasis
70 μSec - Type 1 (Normal Bias) Cassette Tape Eq.
75 μSec - Standard FM Broadcast de-emphasis
120 μSec - Type 2 (High Bias) Cassette Tape Eq.

Additional audio time constants can be found under RIAA and NAB in this glossary.

Time Derivative
This is the instantaneous rate of change of a parameter (such as Voltage, amplitude, or sound pressure level) with respect to time. (i.e. dV / dt, dP / dt, etc.)

Tremolo
Tremolo is the amplitude modulation of a musical note. For example, the tremolo control on a guitar amplifier modulates the gain of its signal by way of a low frequency sine wave oscillator in the 1 to 10 Hz range. A Tremolo effect can be observed easily using the Diamond Cut Spectrogram.

Transformer
A transformer is an alternating current device used to impedance match transducers and electronic circuits to one another. Sometimes, these devices are used with a unity turns ratio to provide isolation from one circuit to another rather than to

impedance match the two. Unity coupled transformers (1:1 turns ratio) are useful in audio applications when it is necessary to break a ground loop source of noise in a system.

Triode
A Triode is an electron tube (or valve) containing three elements. They consist of an anode, cathode, and a control grid. Small changes in grid Voltage produce large changes in values of current in the plate circuit (the ratio of delta plate current to delta grid Voltage is its gain in transconductance or mu.) They are most commonly used in audio pre-amplifier, and other low-level applications. Typical triodes found in audio applications include the 12AX7 and 6SL7 high mu (gain), and the 12AU7 and 6SN7 medium mu devices. All of the devices listed are "dual" (two in one envelope).

Tube
See Electron Tube and/or Valve

Turnover Frequency
Turnover (frequency) is the frequency in a phonograph equalization curve below which the master was recorded with the cutting head operating in constant displacement mode rather than in constant velocity mode. This is used to limit the excursions of the cutting stylus so that bass notes do not cause the cutting stylus to break through to the adjoining groove wall. For a chart listing the most common Turnover Frequencies utilized by 78 RPM records, refer to our ***Charts, Graphs, and Other Useful Info*** section.

Valve
Valve is the British term for an electron tube. It arises out of the valve like effect that a grid has on the flow of electrons between the devices cathode and anode (or plate). Also, refer to Electron Tube.

Vector Quantity
Any physical quantity, like the displacement of a record stylus, whose specification involves both magnitude and direction and which obeys the parallelogram law of addition. The Diamond Cut X-Y display provides a visual indication of the Vector Quantity consisting of one audio channel plotted against the other.

Vertical Cut
(Also known as “Hill and Dale”)
A record recording technique in which the groove modulation (undulations) occur in an up-and-down direction as opposed to side-to-side. This technique was used by Thomas Edison in his original invention of the phonograph, and was maintained as the recording method used by his companies cylinders and Diamond Discs.

Vibrato
Vibrato is the frequency modulation of a musical note. It results in what is perceived as a pulsating change in timbre. A typical vocal embellishment of this type created by singers will occur at a rate of change in frequency occurring around 6 to 8 Hz. This effect can be observed via the Diamond Cut Spectrogram.

Violet Noise
Violet Noise is the inverse of Grey Noise and exhibits a +6dB/Octave slope and has a density proportional to f^2. White noise can be converted to Violet noise via the multifilter by applying the appropriate preset to White (random) noise.

Voice Fundamental Frequency (F0)
The fundamental frequency of a human voice is sometimes referred to as formant F0. For adult females, that range runs from 165 to 255 Hz and for adult males the range runs from 85 to 180 Hz. Speech articulation uniqueness is the result of the ratio of formant frequencies F1 through F4.

Vocal Ranges
Human Vocal ranges (relative to middle C = C4 = 262 Hz) can be broken down from lowest to highest as follows:
Bass: E2 to E4
Baritone: F2 to F4
Tenor: C3 to C5
Contralto: F3 to F5
Mezzo-Soprano: A3 to A5
Soprano: C4 to C6
The notes in these ranges can be created via presets found in the Make Waves Generator.

Voice Frequency Range
The fundamental component frequency range for mature adult, healthy human voices are as follows:

Female: 165 to 255 Hz (Harmonics up to 10 kHz)
Male: 85 to 155 Hz (Harmonics up to 8 kHz)

Volt
(V)
(Voltage)
The unit of measurement of electrical potential difference (or electromotive force) equal to the difference in potential which occurs in a conductor which is carrying 1 Ampere, and the power being dissipated in the conductor is 1 Watt, with the resistance of the conductor being 1 Ohm.

Vorbis (Ogg Vorbis)
Vorbis is a lossy audio compression technique (codec) that is supported by your Diamond Cut software. It is especially useful in low bit rate audio applications (less than 128 kbit/sec). Its file extensions are .ogg and .oga.

VOX
VOX is an acronym for “Voice Operated Transmit” which is derived from the half-duplex two-way radio field. In DC terms, it applies to the automatic activation of the recording function when a signal is detected by the system. The VOX system also ceases recording after the signal vanishes for more than a user settable period of time.

Watt
(P)
(Power)
The Watt is a measurement unit of electrical power equal to the ability to do work at the rate of 1 Joule* per second.

P = V x I wherein P = power in Watts, V = Voltage in Volts, and I = current in Amperes.

*1 Joule = 1 Watt x Second

Wave file (.wav)
Wave (.wav) files are the primary and native PCM (**P**ulse **C**ode **M**odulation) sound file format that DCArt10 & DCArt10/DC Forensics10 supports. This (.wav) is the standard Windows file format.

Wet
“Wet” describes the signal output of a special effects generator (such as the Reverb), which contains the modified (processed) signal. "Wet" refers to the effects signal alone. The non-processed signal from such a generator is referred to as "Dry". As with most special effect generators, the Reverb has an output mix control which allows you to transfer a signal from the effects generator, which ranges from completely dry to completely wet (no source signal), or to some mixture in between.

White Noise
White Noise is random noise that is characterized as containing equal energy per unit frequency (Hertz). White Noise is sometimes referred to as Johnson, shot, or thermal noise. White noise derives its name from the analogous definition of white light. Audio white noise can be created using the Diamond Cut Make Waves function using the function by that same name. White Noise can be converted to Pink Noise and other noise distributions via various Multifilter presets. Please refer to the Pink Noise section of this Glossary for more information..

Window Weighting
Window weighting is a concept that pertains to systems, which involve fast Fourier transforms (FFT's). Signals, which are observed for finite intervals of time, may contain distorted spectral data in the transform due to the ringing of the Sin(f)/f spectral peaks of a rectangular window. This distortion is minimized by the use of a window-weighting function, which is applied before the DFT is performed. The window weighting functions used some of the FFT based DCArt10/DC Forensics10 algorithms is proprietary. The window weighting shapes in the frequency domain measurement tools are user selectable.

.wma File Format (Windows Media Audio)
Windows **M**edia **A**udio format is Microsoft’s implementation of compressed audio files and uses the file extension .wma. You can create files of this type using the Diamond Cut “Save As” command found under the File Menu. Both lossy and lossless compression routines are available.

Wow
A slow periodic change in the pitch or low frequency flutter which may be present on

phonograph, tape, or soundtrack recordings due to a non uniform velocity of the recording medium. Wow is generally a frequency modulating effect that occurs at a deviation rate between 0.5 to 6 Hz. The "Wow" could have been introduced in the recording process, the playback process, or a combination of both. Wow found on record recordings is usually caused by a non-concentric spindle hole. Wow found on tape recordings is generally caused by warped take-up or supply reels. DCArt10/DC Forensics10 is not capable of correcting audio problems of this nature at this point in time.

Wow and Flutter
Wow and flutter is the combined FM effect of both mentioned parameters. The frequency spectrum in which this rate of frequency deviation is made is in the spectrum that exists between 0.5 to 250 Hz.

X-Axis
This is the horizontal axis of a graph. In DCArt10/DC Forensics10, it contains the time information for your .wav file that is divided up into ten equally spaced grids.

Y-Axis
This is the vertical axis of a graph. In DCArt10/DC Forensics10, it contains the amplitude information for your .wav file that is divided up into four equally spaced grids.

Z-Axis
This represents the gray scale intensity or chroma modulation level that you find in the spectrogram/spectrograph display.

Charts, Graphs and Other Info

Additional Technical Information

We've recently added a section to our web site with specific Application Notes for making use of our products. These App Notes provide stimulating specific information on varied uses of this software. These currently include the following Application Notes:

- AN1 – Using DCart or DC Live/Forensics as an Audio Signal Generator
- AN2 – Using DCart or DC Live/Forensics to test your sound cards performance.
- AN3 – Using DC5 or DC Live/Forensics To Create 3 Audio Test CDs
- AN4 – Using DC Live/Forensics: The Electric Network Frequency Analysis
- AN5 – Audio Archival Recording
- AN6 – Speaker Cable Power Loss Wire-chart
- AN7 – Power Line Frequency by Country (hum)
- AN8 – The New Way of recording LPs
- AN9 – Land Line Telephone Interface Circuit
- AN10 – Bringing Acoustic Recordings Back to Life
- AN11 – Proper Use of Stereo Phono Cartridges to Transfer Lateral and Vertical Cut Recordings
- AN12 – Measuring the Dynamic Range of Your Hearing
- AN13 – Digitizing Records using a DCP-47K-F Flat Response Magnetic Phono Cartridge Preamp
- AN14 – Preserving the Proper Phonographic EQ curve with Fractional or High Speed Re-mastering
- AN15 – Theoretical Frequency Response of Records by Format
- AN16 – DIY Vinyl Record Cleaner Formulation
- AN17 – Virtual Phono Preamp Supported Phono Equalization Curves
- AN18 – Comprehensive Feature listing for DCart and DC Forensics 10.xx
- AN19 – Forensics Audio Laboratory Security Set-up and Procedural Tips
- AN20 – Audio Restoration Terminology

- AN21 – Audio Related Charts, Graphs and Tables of Values
- AN22 – Dynamic Bass Processor
- AN23 – Getting Started Guide
- AN24 – Transferring Vertical Cut Records (to your hard drive)
- AN25 – Generating Pink Noise from White Noise
- AN26 – Selection of Stylus for Edison Diamond Disc Records
- AN27 – DCArt10.75 Feature Listing taken from its User's Guide
- AN28 – A New Make Waves Generator

They are located at the following link:

http://www.diamondcut.com/st3/application-notes/

The Engineers at Diamond Cut Productions also maintain a very active BBS / Forum where users intermingle with the programmers and ask questions about the products. As of 11-2018, there are roughly 5,300 topics covered via 28,500 posts along with a search engine. You may find a visit here to be very informative. You can join for free, but you must register in order to create postings.
http://www.diamondcut.com/vforum/

A Presets Sharing Forum is available for your use at:

http://www.diamondcut.com/vforum/forum/preset-exchange-forum/preset-exchange-for-dc8-dc7-dc6-dc5-and-live-forensics-7-live6-and-live6

Download Demo's and Patches can be found at the following site:

http://www.diamondcut.com/st3/support/

The Diamond Cut Productions, Inc Home Page URL is:

http://www.diamondcut.com/

Hot Key Index

Here is a list of all known factory-based Hot Keys/Combos and their real world equivalent. These keyboard accelerators (sometimes just referred to as “Accelerators”) are listed below:

"1"	Play selected area + 0.25 seconds on each side
"2"	Play selected area + 0.5 seconds on each side
“3"	Play selected area + 1 second on each side
"4"	Play selected area + 2 seconds on each side
"A"	Select displayed area (same as double click)
CONTROL+"A"	Select entire file
CONTROL+"C"	Copy selection to clipboard
"D"	Select destination
CONTROL+ "E"	Select/deselect the pencil (need to be zoomed in far enough to see waveform)
"I"	INTERPOLATE Full File (Bi-Modal {BM})
"J"	INTERPOLATE LEFT channel only {BM}
“SHIFT + I”	INTEROPLATE LEFT channel only {BM}
"K"	INTERPOLATE RIGHT channel only {BM}
CONTROL + “I”	INTERPOLATE RIGHT channel only {BM}
“O”	INTERPOLATE (Time Domain Mode)
"L"	PLAY LOOPED
"M"	DROP MARKER
CONTROL+"M"	MUTE
"N"	GO TO NEXT MARKER
CONTROL+"N"	NEW File,
SHIFT+"N"	GO TO PREVIOUS MARKER
CONTROL+"O"	OPEN File
"P"	Open PREFERENCES
Plus Sign (+)	Zoom In X2
Minus Sign (-)	Zoom Out X2
CONTROL+"P"	PRINT
“Q”	Activates "Snap Selection to Zero Crossing"
CONTROL+"R"	Brings up Record screen
"S"	SELECT SOURCE
CONTROL+"S"	FILE SAVE
CONTROL+"U"	UNDO LAST EDIT
CONTROL+"V"	PASTE
ALT+BACKSPACE	UNDO

SHIFT+DELETE	CUT
END	REWIND TO Start
F1	Launch HELP file system
SHIFT+F1	CONTEXT SENSITIVE HELP
F6	NEXT PANE
SHIFT+F6	PREV_PANE
HOME	FORWARD TO END
CONTROL+INSERT	EDIT COPY,
SHIFT+INSERT	EDIT PASTE,
LEFT ARROW	NUDGE RIGHT edge of selected area to the left
RIGHT ARROW	NUDGE RIGHT edge of selected area to the right
SHIFT+LEFT ARROW	NUDGE LEFT edge of sel. area to the left
SHIFT+RT ARROW	NUDGE LEFT edge of sel. area to the right
RETURN	Default button (this is the cancel button on all filters)
SPACEBAR	Play File or Record/Pause (toggle) when the Recording dialog box is active.
UP ARROW	Increments the selected parameters
DOWN ARROW	Decrements the selected parameters
CONTROL+"X"	CUT
"X"	ZOOM OUT
"Z"	ZOOM IN
CONTROL+"Z"	ZOOM OUT FULL
CONTROL+"B"	Paste Bleep
ALT+"S"	Toggles between Spectrogram& Normal Mode
Esc	Exits from the Spectrogram Mode
CONTROL+"T"	Paste Insert

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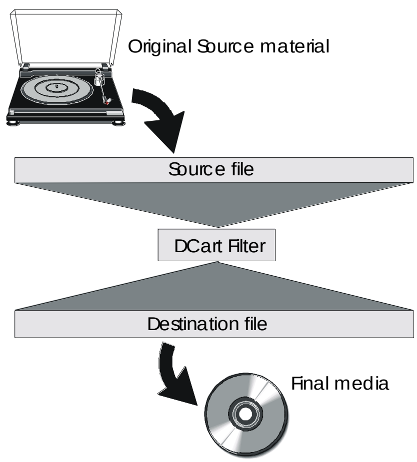
MOUSE WHEEL	Zoom In & Zoom Out

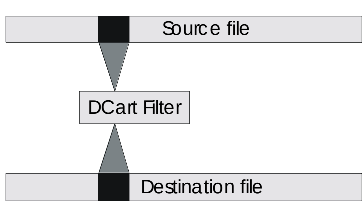
Note: You can customize some of the keyboard accelerators. To do so, go to the View Menu / Toolbars and Docking Windows / Customize / Keyboard

Sync Mode/Non Sync Mode Explanation Process Diagram

The following diagram illustrates the standard filtering process of DCArt10/DC Forensics10 using Classic Edit mode.

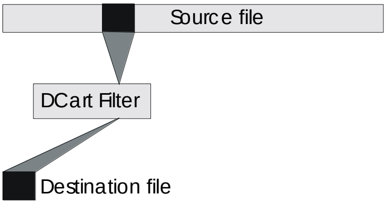


Sync Mode



Sync mode is the default mode of operation for the Classic Editing screen of *DCArt10/DC Forensics10*. In Sync mode, both the Source and Destination files track each other. If you zoom into a section of the Source file, the Destination file will zoom to the same section. When you process the Source file using a *DCArt10/DC Forensics10* filter, the program reads the Source file, processes it, and writes it to the Destination file at exactly the same position as the Source file. This means that if you want to reprocess a section in the middle of a song, just highlight the section in the Source file that needs processing and run the filter again.

Non-Sync Mode



In Non-Sync mode, the highlighted section of the Source file is read and processed by the *DCArt10/DC Forensics10* filter. The processed section is then written to the Destination file, starting at the beginning of the file. If a Destination file already exists, it will be overwritten (a prompt warns you of this). This mode is useful when only a section of the Source file needs to be extracted, or for testing a filter's settings before processing an entire file.

Function Finder Table

In a program that has as many tools as both DCArt10 and DC Forensics10, we thought it might be helpful if you could look up the function to help you more easily find the tool you need.

Functions Unique to DC Forensics10 are shown in Italics

Function	**Feature**	**Feature Location**
3 Band EQ	Phono Pre Amp (VPP)	Filter Menu
10 Band Graphic EQ	Graphic EQ	Filter Menu Under “EQ” under “Graphic EQ”
20 Band Graphic EQ	20 Band Graphic EQ	Filter Menu Under “EQ”
30 Band Graphic EQ	*30 Band Graphic EQ*	*Filter Menu Under “EQ”*
32,000 Band Equalizer (FIR Based)	*Spectral Filter*	*Forensics Menu*
33 RPM Record Click Filter	Expert or EZ Impulse Filter	Filter Menu
45 RPM Record Click Filter	Expert or EZ Impulse Filter	Filter Menu
78 RPM Record Click Filter	Expert or EZ Impulse Filter	Filter Menu
80 RPM Record Click Filter (Edison Diamond Disc or Pathé)	Expert or EZ Impulse Filter	Filter Menu
A-Law Compression	A-Law To .wav Conversion	Open/Save File As x.y
About DCart	About DCart	Help Menu
Acoustical Recording Transfer	Phono Pre Amp	Filter Menu
Adaptive Filter	*Adaptive*	*Forensics Menu*
Adaptive Noise Rejection	*Forensics AFDF*	*Continuous Noise Filter*
Add File to End of Existing File	Append To End	Edit Menu/Paste
ADPCM Compression	ADPCM to .wav Conversion	Open/Save File as x.y.
AIFF	Saving a file in the AIFF Format	File\Save As Menu
ALC (Multiband)	Punch & Crunch	Effects Menu
ALC (Wideband)	Dynamics Processor	Effects Menu
Ambience Enhancer	Reverb	Effects Menu
Ambience Reduction	CNF Spectral Subtraction	Continuous Noise Filter under Filter Menu
American 78 RPM Turnover Curve	Phono Pre Amp	Filter Menu
Amplifier Non-linearity Reversal	*Polynomial Filter*	*Forensics Menu*
Amplifier Non-linearity Simulation	Virtual Valve Amplifier	Effects Menu
Amplify Display	Right Slider Control	Next to Right Side of Display
Amplitude Measurement (Relative in dB)	VU Meter	View Menu
Amplitude Measurements and Statistics	*Waveform Statistics*	*Forensics Menu*
Annotate Printed Document	Page Setup	File Menu
Automatic Noise Reduction	EZ Clean Filter	Filter Menu
Automatically Assign Names To Tracks Using CD Internet Database	CD Data base (Internet)	File Menu
Averaging Filter	Averaging Filter	Filter Menu

Average Phase Angle Management	XY Display	View Menu
Background Noise Amplification	*Whisper Enhancer*	*Forensics Menu*
Band Pass (FIR Based)	Brick Wall Filter	Forensics Menu
Balance (audio sound level)	Virtual Phono Preamp	Filter Menu
Balance (gain controls)	File Conversion Filter	Filter Menu
Band Pass (IIR Based)	Band Pass Filter	Filter Menu
Band Pass Filter (Butterworth Response)	Band Pass Filter (Butterworth Checkbox)	Filter Menu
Band Pass Filter (Chebyshev Response)	Band Pass Filter (Chebyshev Checkbox)	Filter Menu
Band Stop Filter (FIR Based)	Brick Wall Filter	Forensics Menu
Bass Control	Phono Pre Amp	Filter Menu
Bass Sound Enhancement	Fat Bass	VVA Under Effects Menu
Batch File Editor	Batch File Editor	Filter Menu
Batch Processing	Batch File Editor	Filter Menu
Binaural DSS – Left Reference	*Continuous Noise Filter*	*Filter Menu*
Binaural DSS – Right Reference	*Continuous Noise Filter*	*Filter Menu*
Both Channels Processed	L/R Icon	Toolbar
Break a File into Smaller Pieces at Markers	Chop File into Pieces	CD-Prep Menu
Brick Wall Filters	Brick Wall Filters	Forensics Menu
Broadcast .wav (BWF)	Create a File having the Broadcast .wav format	File\Save As Menu
Burn a CD ROM	Burn a CD	CD Prep Menu
Buzz Reduction	Harmonic Reject Filter or Impulse Filter	Filter Menu
Cascade Multiple Filters	Multi-Filter	Filter Menu
CD Creation	CD Burner	CD Prep Menu & DC Tune Library
Cell Phone Noise Interference Reduction	*Cell Phone Noise Filter*	*Forensics Menu*
Cepstrum Plots	*Voice ID*	*Forensics Menu*
Change Resolution of File	Change Sample Rate / Resolution	Edit Menu
Change Sampling Rate of File	Change Sample Rate / Resolution	Edit Menu
Channel Mixing / Blending	Channel Blender	Effects Menu
Channel Toolbar Activation	Channel Toolbar	View Menu
Chop File into Pieces at Markers	Chop File into Pieces	CD-Prep Menu
Clone a File	Create a replicate of the opened file	File Menu\Clone Source
Clipping Repair	De-Clipper	Forensics Menu
Columbia LP Curve (Early)	Phono Pre Amp	Filter Menu

Comb Filter 1	Multiple Notch Filters / Multi-Filter	Filter Menu
Comb Filter 2	*Spectral Filter*	*Forensic Menu*
Combine 2 Mono Files into a Stereo File	File Split and Re-Combine	Edit Menu
Comparing File Histogram Statistics	*Comparative Histogram*	*Forensics Menu*
Compress File Size	Save as .mp3 or .wma or .ogg or .oga	File Menu
Compressor (Instantaneous)	*Polynomial Filter*	*Forensics Menu*
Compressor (multiband)	Punch & Crunch	Effects Menu
Compressor (wideband)	Dynamics Processor	Effects Menu
Context Sensitive Help	Point At Feature	Hit F1 Key
Convert a Stereo .wav file to Monophonic	File Conversions	Filter Menu
Convert CDs to MP3s	CD to MP3 Conversion	File Menu
Convert Monophonic .wav file to Stereo	File Converter	Filter Menu
Convert Random to Brown Noise	Multi-Filter (preset)	Filter Menu
Convert Random to Pink Noise	Multi-Filter (preset)	Filter Menu
Convert Random to Seismic Noise	Multi-Filter (preset)	Filter Menu
Convert Redbook Audio on a CD to a .wav file.	Rip CD Feature	File Menu
Convert Stereo .wav file to Stereo Reverse .wav file	File Converter	Filter Menu
Convert .wav file to AIFF Type	Save As	File Menu
Convert .wav file to MP3 Type	Save As	File Menu
Copy a Portion of a .wav file	Copy	Edit Menu
Corner Frequency vs. Time	Filter Sweeper	Effects Menu
Cracked Record Click Remover	Big Click Filter	Filter Menu
Create a Playlist	Open / Create Playlist	File Menu
Create Silence	Mute	Edit Menu
Create Your Own Filter	Multi-Filter	Filter Menu
Cross fade Two .wav files	Paste - Cross fade	Edit Menu
Data Base of Files (DC Tunes)	Open/Create Playlist	File Menu/View Menu
Data Disc Burner	Burn A Data Disc	File or CD Prep Menu
DC Offset Eliminator	High Pass Filter	Filter Menu/Preset
DC Offset Generation	*Polynomial Filter*	*Forensics Menu*
Deep Bass is Missing	Sub-harmonic Synthesizer	Effects Menu
De-Esser	Dynamics Processor	Effects Menu
Delay or Advance timing of	Time Offset feature in File	Filter Menu

One Channel compared to the other	Conversions	
Delete .wav files	Delete Files	File Menu
De-verb (reverberation reduction on spoken word)	Dynamics Processor in Expander Mode (use Presets)	Effects Menu – Dynamics Processor – Expander Mode
Differentiator	High Pass Filter (Preset)	Filter Menu
Disguise a Voice	*Voice Garbler*	*Forensics Menu*
Display Colors	Preferences / Display	Edit Menu
Display Setup	Preferences / Display	Edit Menu
Distortion Reduction	*Polynomial Filter*	*Forensics Menu*
Distortion Synthesizer	*Polynomial Filter*	*Forensics Menu*
Dithering	Change Sample Rate / Resolution	Edit Menu
Draw Waveform Segment	Pencil Tool	Toolbar
DSS – Delay Reference	*Continuous Noise Filter*	*Filter Menu*
DTMF Filter	Paragraphic EQ	Filter Menu under "EQ"
Dynamic Bass Processor	Dyna Bass	Effects Menu
Dynamic Enhancer	Dynamic Noise Filter	Filter Menu
Dynamic Noise Filter	Dynamic Noise Filter	Filter Menu
Dynamic Spectral Subtraction (DSS)	*Continuous Noise Filter*	*Filter Menu*
Dynamics Compressor (multiband)	Punch & Crunch	Effects Menu
Dynamics Compressor (wideband)	Dynamics Processor	Effects Menu
Dynamics Expander (multiband)	Punch & Crunch	Effects Menu
Dynamics Expander (wideband)	Dynamics Processor	Effects Menu
Echo (reverberation) reduction	Dynamics Processor – Expander Mode - Presets	Effects Menu
Echo 1	Reverb	Effects Menu
Echo 2	Time Offset Feature in File Converter	Filter Menu
Echo Chamber	Echo Effect	Effects Menu
Editing History	Fast-Edit History	View Menu
Enhance Audio Quality	EZ Enhancer	Effects Menu
Enhance Sibilance	Overtone Synthesizer	Effects Menu
European 78 Turnover Curve	Phono Pre Amp	Filter Menu
Evens Harmonic Reject	Harmonic Reject Filter	Filter Menu
Exciter (Harmonic)	Checkbox under the Virtual Valve Amplifier	Effects Menu
Exit the Program	Exit	File Menu or "X" on the Top Right of screen
Expander (Instantaneous)	*Polynomial Filter*	*Forensics Menu*
Expander (multiband)	Punch & Crunch	Effects Menu

Expander (wideband)	Dynamics Processor	Effects Menu
Export Presets	Manage Presets	Edit Menu
Extract Audio From AVI Video	Open Video Files	File Open
EZ Clean Filter	Automatic Impulse, Hiss and Hum Filters	Filter Menu
EZ Enhancer	Complex Audio Enhancement Effects	Effects Menu
EZ Forensics Filter	Adaptive Forensics Audio Filters	Forensics Menu
EZ Impulse Filter	Multiple & Adaptive Impulse Type Filters	Filter Menu
Fade In	Fade-In	Edit Menu
Fade Out	Fade-Out	Edit Menu
Fast Edit Mode	Preferences / General	Edit Menu
File Conversions	File Conversions Filter	Filter Menu
File Information	File Information	View Menu
File Time Reversal	Reverse File	Effects Menu
File Toolbar Activation	File Toolbar	View Menu
Filter Toolbar Activation	Filter Toolbar	View Menu
Find Peak Value	Spectrum Analyzer (Checkbox)	View Menu
Forensics Adaptive Noise Reduction	*Auto Voice Filter*	*Forensics Menu*
Forensics Adaptive Noise Reduction; Frequency Domain	*AFDF Mode in the CNF Filter*	*Continuous Noise Filter – choose AFDF Mode*
Forensics Adaptive Noise Reduction; Time Domain	*Adaptive Filter*	*Forensics Menu*
Formant Identification	*Voice ID*	*Forensic Menu*
Frequency & Amplitude vs. Time	Spectrogram	Forensics Menu
Frequency & Amplitude vs. Time – High Resolution	*High Resolution Spectrogram*	*Forensics Menu*
Frequency Domain Measurements	Spectrum Analyzer	View Menu
Frequency Doubler	*Polynomial Filter*	*Forensics Menu*
Gain Change	Gain Change	Edit Menu
Gain Change vs. Time	Gain Change	Edit Menu
Gain Normalizer	Normalized Gain Scaling	CD-Prep
Gain vs. Time	Gain Change	Edit Menu
Generate Overtones (Evens)	Overtone Synthesizer	Effects Menu
Generate Overtones (Odds and Evens)	Virtual Valve Amplifier	
Generate Sub-harmonics	Sub-harmonic Synthesizer	Effects Menu
Harmonic Reject Filter	Harmonic Reject Filter	Filter Menu
Hear Removed Signal or Noise	Keep Residue Checkbox	In Filter Dialog Boxes where appropriate

High Pass (FIR Based)	Brick Wall Filter	Forensics Menu
High Pass (IIR Based)	High Pass Filter	Filter Menu
High Pass Corner Frequency vs. Time	Filter Sweeper	Effects Menu
High Pass Filter (Butterworth Response)	High Pass Filter (Butterworth Checkbox)	Filter Menu
High Pass Filter (Chebyshev Response)	High Pass Filter (Chebyshev Checkbox)	Filter Menu
High Resolution Frequency Response Contouring	*30-Band Graphic EQ*	*EQ Menu*
Highlight an Area of the .wav file	Left Mouse + Drag	Mouse
Hiss Reduction 1	Continuous Noise Filter	Filter Menu
Hiss Reduction 2	Dynamic Noise Filter	Filter Menu
Hiss Reduction 3	Hiss Filter	Hiss Filter in EZ Clean Filter
Hum Reduction	Notch Filter	Filter Menu
Import Presets	Manage Presets	Edit Menu
Impulse Filter (Easy)	EZ Impulse Filter	Filter Menu
Impulse Filter (Expert)	Expert Impulse Filter	Filter Menu
Insert a Segment into a .wav file	Paste - Insert	Edit Menu
Insert File at Beginning of Another File	Insert at Start	Edit Menu/Paste
Instant Audio Review	Flashback	Multi-Filter
Integrator	Low Pass Filter (Preset)	Filter Menu
Inter-modulation Distortion Reduction	CNF used in Artifact Suppression Mode	Filter Menu
Interpolate a Portion of a .wav file	Paste - Interpolate	Edit Menu
Interpolate Both Channels	“I” Key on Keyboard	Paste Interpolate or Keyboard
Interpolate Left Channel Only	“J” Key on Keyboard	Keyboard
Interpolate Right Channel Only	“K” Key on Keyboard	Keyboard
Last 4 Files Opened	Listing Near the Bottom of the Menu	File Menu
Lateral Cut Record to Monophonic Conversion	File Conversions / Presets	Filter Menu
Lead Vocal Removal	Channel Blender	Effects Menu
Left Channel Process Only	L Icon	Toolbar
Limiter	Dynamics Processor Presets	Effects Menu
Log to Disc	Multi-Filter / Log to Disc Button	Filter Menu
Lossless File Compression	FLAC (.flac)	File Menu (Save As)
Loudness	Volume Control	View Menu
Loudness Maximizer	Punch & Crunch in Compressor Mode	Effects Menu

Low Pass (FIR Based)	Low Pass Filter	Forensics Menu
Low Pass (IIR Based)	Low Pass Filter	Filter Menu
Low Pass Corner Frequency vs. Time	Filter Sweeper	Effects Menu
Low Pass Filter (Butterworth Response)	Low Pass Filter (Butterworth Checkbox)	Filter Menu
Low Pass Filter (Chebyshev Response)	Low Pass Filter (Chebyshev Checkbox)	Filter Menu
Marker (Add)	Right Mouse Click on Display	Right Mouse Button
Marker (Annotate or Label)	Right Mouse Click on Label Marker	Right Mouse Button
Marker (Clear All)	Clear All Markers	Marker Menu
Marker (Clear an Individual Marker)	Right Mouse Click on Delete Marker	Right Mouse Button
Marker (Drop)	Drop a Marker	Marker Menu
Marker (Got Next One)	Got Next Marker	Marker Menu
Marker (Got Previous One)	Got Previous Marker	Marker Menu
Marker (Highlight in between two)	Double Left Mouse click between 2 markers	Left Mouse Button
Marker (Highlight in between two)	Highlight Marked Area	Marker Menu
Marker (Move)	Drag with Mouse	Left Mouse button
Markers (Lock all in Place)	Lock Markers	Markers Menu
Median Filter	Median	Filter Menu
Medium Resolution Frequency Response Contouring	20-Band Graphic EQ	EQ Menu
Mids Control	Phono Pre Amp	Filter Menu
Mix two .wav files together	Paste - Mix	Edit Menu
MME Drivers Setup	Preferences / Soundcard	Edit Menu
Monitor Input vs. Output	Bypass (Checkbox)	All Filter and Effects Dialog Boxes
Mono DSS – Delay Reference	Continuous Noise Filter	Filter Menu
Move .wav file from Destination to the Source Display	Make Destination the Source	File Menu
MP3 Encoder Setup	Preferences / MP3 Encoder	Edit Menu
Mu-Law Compression	Mu-Law To .wav conversion	Open/Save File As x.y
Music from Speech Separator	*DSS in the CNF*	*Filter Menu*
Narrow Crackle Impulse Noise	Narrow Crackle Filter	Filter Menu
Narrowband Noise Rejection	*30 Band EQ*	*EQ Menu*
Noise Gate	Dynamics Processor Presets	Effects Menu
Noise Reduction	Continuous Noise Filter	Filter Menu

(Wideband)		
Non-linear Transfer Function	*Polynomial Filter*	*Forensics Menu*
Normal Continuous Noise Filter	Continuous Noise Filter	Filter Menu
Normalize Loudness between Multiple .wav files	Auto Leveling Feature	Batch Processor
Notch (IIR Based)	Notch Filter	Filter Menu
Notch Filter vs. Time	Filter Sweeper	Effects Menu
Odds & Evens Harmonic Reject	Multi-Filter	Filter Menu
Odds Harmonic Reject	Harmonic Reject Filter	Filter Menu
Offset Display	Left Slider Control	Next to Right Side of Display
On Line Help	Help, Contents	Help Menu
Open a Playlist	Open / Create Playlist	File Menu
Open a .wav file into Destination Display	Open Destination	File Menu
Open a .wav file into Source Display	Open Source	File Menu
Overtone Synthesis (x 2)	Overtone Synthesizer	Effects Menu
Paragraphic EQ	Paragraphic EQ	Filter Menu under “EQ”
Parametric EQ	Paragraphic EQ	Filter Menu under “EQ”
Paste as a New .wav file	Paste as New File	Edit Menu
Paste Over a Portion of a .wav file	Paste - Over	Edit Menu
Paste Tone	Paste Bleep	Edit Menu/Paste
Pause Play	Play / Record Toolbar	Pause Button on Toolbar
Peaking Filter(s)	Paragraphic EQ	Filter Menu
Pencil Editing	Pencil Icon	Toolbar
Pencil Tool	Pencil Icon	Toolbar
Phase Inversion 1	File Conversions	Filter Menu
Phase Inversion 2	Channel Blender	Effects Menu
Phase vs Time Plot	View Phase vs Time	Forensics Menu
Phono Equalization Curves	Paragraphic EQ Presets	Filter Menu
Pitch Shift	Stretch & Squish Effect	Effects Menu
Place Markers Automatically on Silent spots	Find and Mark Silent Passages	CD-Prep Menu
Play a CD	DC Tune Library	File Menu\DC Tune Library
Play Controls Activation	Play Controls	View Menu
Play File	Play / Record Toolbar	Play Button / Spacebar
Play in a Loop	Loop Play	Loop Play Button on Toolbar
Playlists	DC Tune Library	File Menu/View Menu
Preview Filter or Effect	Preview Button	All Filters and Effects
Print Document	Print	File Menu
Print Preview	Print Preview	File Menu

Printer Setup	Print Setup	File Menu
Process in Batch Mode	Batch File Editor	Filter Menu
Quantize for CD Audio	Quantize for CD Audio	CD-Prep Menu
Ranking Filter	Median Filter	Filter Menu
Real Time Feed through	Multi-Filter / Live Preview Button	Filter Menu
Rebuild the Peak File (Waveform)	Rebuild Peak File	View Menu
Recorder	Record File	Edit Menu or Toolbar
Rectifier	*Polynomial Filter*	*Forensics Menu*
Reduce File Size	Save as .mp3 or .wma or .ogg or .oga	File Menu
Remove Portion of a .wav file	Cut	Edit Menu
Remove Silence from a file Automatically	*Remove Silence Tool*	*Forensics Menu*
Reverberation Reduction	Dynamics Processor (Expander Mode – Presets)	Effects Menu – Expander checkbox - Presets
Reverberation Simulation	Reverb	Effects Menu
Reverse RIAA EQ Curve	Paragraphic EQ Presets	Filter Menu
Review Real-time Audio	Flashback Mode	Multi-Filter
Rewind to Beginning	Rewind Button	Rewind Button on Toolbar
RIAA EQ Curve	Phono Pre Amp	Filter Menu
Right Channel Process Only	R Icon	Toolbar
Rip a CD	Rip CD Tracks	File Menu
Rumble Reduction 1	Continuous Noise Filter	Filter Menu
Rumble Reduction 2	High Pass Filter	Filter Menu
Run Function	Run Button	All Filters and Effects
Save a Destination Display .wav file	Save Destination As	File Menu
Scalar Amplitude Measurement	VU Meter	View Menu
Scratch & Crackle Filter	EZ Impulse	Filter Menu
Scrub Feature	Variable Speed Playback using the Mouse – Forward and Reverse	Toolbar
Selective Filtering	Sync Files	View Menu
Shelving Filters (Low or High Frequency)	Paragraphic EQ or VPP	Filter Menu
Simulate Tubes and Transistors	VVA	Effects Menu
Sine Wave Synthesis	Make Waves	Edit Menu
Slot Filter (IIR Based)	Notch Filter	Filter Menu
Software Revision Number	About DCart	Help Menu
Sound Activated Recording	Record File / VOX Recording Checkbox	Edit Menu or Toolbar
Sound Card Set Up	Preferences	Edit Menu
Spectral Subtraction Filter	Continuous Noise Filter	Filter Menu

Spectral Inverse Filter	*Normalize a Signal to Constant Power Spectral Density*	*Forensics Menu under Spectral Filter "EQ Mode" Selector*
Spectrogram	*View Spectrogram*	*View Menu*
Spectrogram Options	*Preferences / Spectrogram*	*Edit Menu*
Spectrum Analyzer	Spectrum Analyzer	View Menu
Speech Filter (FIR based)	Brick Wall Filter (presets)	Forensics Menu
Speech Filter (IIR based)	Band Pass Filter (presets)	Filter Menu
Speed Correction	Change Speed Filter	Effects Menu
Splash Screen – On / Off	Preferences / Display	Edit Menu
Split Stereo File into 2 Mono Files	File Split and Re-Combine	Edit Menu
Square Wave Synthesis	Make Waves	Edit Menu
Static Remover	Expert Impulse Filter	Filter Menu
Status Bar Activation	Status Bar	View Menu
Stereo Simulation 1	File Converter (Time Offset)	Filter Menu
Stereo Simulation 2	Reverb	Effects Menu
Stereo Simulation 3	Paragraphic Equalizer, Presets	Filter Menu
Stroboscopes, Printable	Diamond Cut Install Folder	Strobe50Hz.wmf Strobe60Hz.wmf
Sub-Harmonic Synthesis (÷ 2)	Sub-harmonic Synthesizer	Effects Menu
Swept Waveform Synthesis	Make Waves	Edit Menu
Sync Files	Sync Files	View Menu
Synchronize Two .wav files	Sync Files	View Menu
Synthesize Round Bass	VVA	Effects Menu
Synthesize "Sweet" Treble	VVA	Effects Menu
Synthesize "Warm" Treble	VVA	Effects Menu
System Setup	Preferences	Edit Menu
System Status	Status Bar	View Menu (Bottom of Screen)
Tape Equalization Curves	Paragraphic EQ Presets	Filter Menu
THD Measurement	Spectrum Analyzer (Checkbox)	View Menu
Time at Cursor	Time Display	View Menu
Time Compensation Calculator and Corrector	Change Speed	Effects Menu
Time Compression	Stretch & Squish	Effects Menu
Time Delay	Echo Effect	Effects Menu
Time Expansion	Stretch & Squish	Effects Menu
Time Offset	File Conversions	Filter Menu
Time Span	Time Display	View Menu
Time Stamp the Segment	Multi-Filter / Checkbox	Filter Menu
Time Start	Time Display	View Menu
Time Stop	Time Display	View Menu
Timer Recording	Timer Record	Edit Menu

Timing Measurements	View Time Display	View Menu
Tip of the Day Activation	Tip of the Day / Checkbox	Help Menu
Tip of the Day De-Activation	Tip of the Day / Checkbox	Launch, and then after the Splash Screen
Tone Controls	Phono Pre Amp	Filter Menu
Top Octaves Missing	Overtone Synthesizer or Virtual Valve Amplifier	Effects Menu
Total Harmonic Distortion Measurement	Spectrum Analyzer (Checkbox)	View Menu
Track a Frequency	*Frequency Tracking feature used in conjunction with the Spectrogram*	*Forensics Menu*
Track Sub-sonic events	*Subsonic Explorer*	*Forensics Menu*
Treble Control	Phono Pre Amp	Filter Menu
Triangle Wave Synthesis	Make Waves	Edit Menu
Tube Simulator	Virtual Valve Amplifier	Effects Menu
Turnover Curves	Paragraphic EQ Presets	Filter Menu
Undo Edit	Undo	Edit Menu
Universal Impulse Filter	Expert Impulse Filter	Filter Menu
User Discussion Group (BBS / Forum)	User Discussion Group	Help Menu
User Preferences	Preferences	Edit Menu
Valve Simulator	Virtual Valve Amplifier	Effects Menu
Variable Frequency Response vs. Time	Filter Sweeper	Effects Menu
Variable Noise vs. Time	Adaptive Filter	Forensics Menu
Vector Measurement	XY Display (X vs Y)	View Menu
Vertical Cut Record to Monophonic Conversion	File Conversions / Presets	Filter Menu
Vinyl LP Click Filter	Expert Impulse Filter	Filter Menu
Voice Activated Recording	Record File / VOX Recording Checkbox	Edit Menu or Toolbar
Voice Print	*Voice ID*	*Forensics Menu*
Volume Control	Volume Control	View Menu
Volume Control	Virtual Phono Preamp	Filter Menu
Vorbis (Conversion to Ogg Vorbis File format)	Save as .ogg or.oga	File Menu\Save As
VOX Recording	*Record File / VOX Recording Checkbox*	*Edit Menu or Toolbar*
VU Meter	VU Meter	View Menu
VU Meter Scale, Log or Linear	Preferences / General	View Menu
Weighted Median Filter	Median Filter	Filter Menu
White Noise Synthesis	Make Waves	Edit Menu
Wideband Noise Reduction	Continuous Noise Filter	Filter Menu
Whisper Audibility	*Whisper Enhancer*	*Forensics Menu*
Wind Noise	Wind Noise Filter (speech)	Filter Menu
WMA Format File Saving	Save As .wma	File\Save As Menu

WMD Drivers Setup	Preferences / Soundcard	Edit Menu
XY Display	XY Display	View Menu
Zoom (Binary)	Zoom In or Out X2	View Menu or Toolbar
Zoom (Highlighted)	Zoom In or Out	View Menu or Toolbar

Measurement Tools Table

Measurement	Stimulus System	Response System
Acoustical Signature	Calibrated microphone driving sound card input	Spectrogram
Aliasing Products	Swept Sine Wave (Make Waves Generator)	Spectrum Analyzer
Amplifier Linearity	Triangle Wave Generator (Make Waves)	Waveform Display Window
Amplitude Measurements	Time Domain Signal (highlighted by user)	Waveform Statistics in Forensics Menu
Amplitude vs. Frequency	Any signal requiring analysis	Spectrum Analyzer
Amplitude vs. Time	Any signal requiring analysis	Waveform Display Window
Analog Tape Authentication	Look for line frequency or multiple line frequency spectral spikes or look for a higher order noise roll-off rate	Spectrum Analyzer operating in high resolution mode
Analog Tape Recorder Tape Head Azimuth Alignment	Azimuth Reference Tape	XY Display / Vector-scope
Ballistics Fingerprint	Audio Recording of ballistics events to be compared	Spectrogram
Average Phase Angle	Any Binaural Signal	Averaging Selection Box in the XY Display
Cross-talk	Sine Wave into One Channel from the Make Waves Generator; Terminate the opposite channels input with the proper impedance	Spectrum Analyzer or VU Meters
DC Offset	Terminate input with proper input impedance	Waveform Display Window
Dynamic Range	Any combination of signals requiring analysis	VU Meters with Peak Hold
Fall Time	Any signal requiring analysis	Waveform Display Window and Time Display with Markers
Frequency	Any signal requiring analysis	Spectrum Analyzer
Frequency & Amplitude vs. Time	Any signal requiring analysis	Spectrogram
Frequency Distribution	Any signal requiring analysis	Spectrum Analyzer
Frequency Ratio	Any signal requiring analysis	XY Display / Vector-scope
Frequency Response	Swept Sine Wave or Random	VU Meter (when using the

	Noise made by the Make Waves Generator	Swept Sine Wave) or the Spectrum Analyzer (when using the Random Noise Generator)
Hard Disc Recording Time Available	Any signal being recorded	Recording Function
Instantaneous Frequency	Any signal requiring analysis	Waveform Display Window and Time Display with Markers. Calculate: F= 1/t
Inter-modulation Distortion	Dual Sine Wave Tones made with the Make Waves Generator and summed together with "Paste Mix"	Spectrum Analyzer
Left Channel vs. Right Channel	Any signal(s) requiring analysis	XY Display / Vector scope
Linearity	Triangle Wave created using the Make Waves Generator	Waveform Display Window
Noise Floor	Properly Terminated Input	Spectrum Analyzer
Peak Amplitude	Any signal requiring analysis	VU Meter using the Peak Hold feature
Phase Angle	Any pair of signals having coherence	XY Display / Vector-scope
Phase Margin of equipment having a control loop system (1^{st}& 2^{nd} order)	Stereo Square Wave Generator (Make Waves)	Time display window (dampening factor of 'ring-out')
Phase vs Time	Two Channels with respect to one another	Forensics Menu – View Phase vs Time
Power Amplifier Frequency Response vs. Output	Swept Sine Wave (Make Waves)	Proper Loading resistor and True RMS reading Voltmeter
Real Time of an Event	Any signal requiring analysis	Timer Recording with Time and Date Stamping
Recording Position	Any signal being recorded	Recording Function
Relative Amplitude (Scalar)	Any combination of signals requiring analysis	VU Meter
Relative Loudness	Any combination of signals requiring analysis	VU Meter
Rise Time	Any signal requiring analysis	Waveform Display Window and Time Display with Markers
Room Acoustical Balance	Random Noise Generator (Make Waves)	Spectrum Analyzer
Room Acoustical Propagation Delay & Reflection	Calibrated microphone driving a sound card input & an Impulse source like a handclap or an impulse created with the "Pencil" feature	Waveform Display Window, Markers and Time Display Feature
Signal to Noise Ratio	Sine Wave @ 0 dB vs. Noise Floor	Spectrum Analyzer

Slew Rate	Square Wave Generator (Make Waves)	Waveform Display Window and Time Display with Markers
Sound Card Performance	Please Refer to Application Note 2 (AN-2)	Please Refer to Application Note 2 (AN-2)
Sound Card Recording Level / Clipping	Audio signal applied to sound Card	Recording Function with recording VU meter and peak indicator
Stereo Separation	Sine Wave into One Channel from the Make Waves Generator; Terminate the opposite channels input with the proper impedance	Spectrum Analyzer, VU Meters or X-Y Display
Tape Recording Speed	Any analog tape recorded signal	Spectrum Analyzer
Time Derivative (dV / dt)	Any signal requiring analysis	Waveform Display Window and Time Display with Markers
Time Interval between Events	Markers	Time Display Feature
Total Harmonic Distortion (% THD)	Any Audio Device requiring performance testing using a Sine Wave stimulus (Make Waves)	Spectrum Analyzer with “Show THD” function and “Show Peak” enabled
Transient Response	Any Active Audio Device with a control loop with a Square Wave applied (Make Waves Generator)	Waveform Display Window
Turntable RPM	Neon or Fluorescent Lamp connected to a known Frequency Source	Printable Strobe discs found in the your software.
Voice Print comparison	Audio Signal of Voice(s) to be compared	Spectrogram
Waveform Statistics	Forensics Recordings	Histogram or Histogram vs Time
Whisper Amplifier/Enhancer	Forensics Audio Recordings with very Weak “far” parties	Whisper Enhancer (Forensics Version Only)
Wow and Flutter	Sine Wave (Make Waves)	Spectrum Analyzer

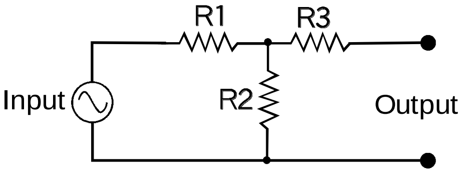
Attenuation Chart

Here is a table of resistance multipliers for a symmetrical (equal input and output impedance) "T" Pad attenuator:

R1 = Attenuator Input Resistor

R2 = Attenuator Shunt Resistor
R3 = Attenuator Output Resistor

(Note: R1 = R3 & the output impedance of the Input Source and the input impedance of the Output Load circuit must also be equal to R1 and R3)



Attenuation (dB)	**R1 & R3 (normalized Ohms)**	**R2 (normalized Ohms)**
0.0	0.000000	infinite
0.5	0.028775	17.362
1.0	0.057501	8.6668
2.0	0.114620	4.3048
3.0	0.171000	2.8385
4.0	0.226270	2.0966
5.0	0.280130	1.6448
6.0	0.332280	1.3386
7.0	0.382480	1.1160
8.0	0.430510	0.94617
9.0	0.476220	0.81183
10.0	0.519490	0.70273
20.0	0.818180	0.20202
40.0	0.980198	0.020002
60.0	0.998000	0.0020000
80.0	0.999800	0.00020000
100.0	1.000000	0.000020000

To use this table, multiply the input (or output) impedance of your circuit by the numbers associated with the attenuation that you desire. Remember, this table of values requires that the input terminating impedance and the output terminating impedance of the circuits on each side of the attenuator be present and of the same value. To obtain values of attenuation that are not in this table, merely cascade "T" sections adding up to the value (in dB) which you desire. For example, to achieve 23 dB, cascade a 20 dB section with a 3 dB section.

Decibels

The following table shows the relationship between Voltage, Current, and Power ratios and Decibels:

Numerical Ratio	**Voltage or Current Ratio in dB**	**Power Ratio in dB**
1 : 1	0	0
2 : 1	6.0	3.0
3 : 1	9.5	4.8
4 : 1	12.0	6.0
5 : 1	14.0	7.0
6 : 1	15.6	7.8
7 : 1	16.9	8.5
8 : 1	18.1	9.0
9 : 1	19.1	9.5
10 : 1	20	10
100 : 1	40	20
1,000 : 1	60	30
10,000 : 1	80	40
100,000 : 1	100	50
1,000,000 : 1	120	60
10,000,000 : 1	140	70
100,000,000 : 1	160	80
1,000,000,000 : 1	180	90
10,000,000,000 : 1	200	100

Resistor Color Code

Standard RMA (Radio Manufacturers Association) Color Code:

Color	**Significant Figure (Mantissa)**	**Decimal Multiplier (Exponent)**
Silver	-	0.01
Gold	-	0.1
Black	0	1.0

Brown	1	10
Red	2	100
Orange	3	1,000 (1K)
Yellow	4	10,000 (10K)
Green	5	100,000 (100K)
Blue	6	1,000,000 (1M)
Violet	7	10,000,000 (10M)
Gray	8	100,000,000 (100M)
White	9	1,000,000,000 (1G)

Note 1: K = Kilo, M = Mega, G = Giga
Note 2: This same color code scheme is sometimes used to identify the values of other electronic components and electrical wires in a system.
Note 3: The tolerance band (next to the Multiplier Band) has the following translation: No Band: +/- 20 %, Silver Band: +/- 10 %, Gold Band: +/- 5%, Red Band: +/- 2%, Brown Band: +/- 1%, Green Band: +/- 0.5 %, Blue Band: +/- 0.25%, Violet Band: +/- 0.10%
Note 4: Some resistors have an additional band denoting the Temperature Co-efficient of the component.

Sound Level

The following is a chart showing some sound sources and their Acoustic Power and Sound Power Levels measured at 10 meters from the source:

Sound Source (measured at 10 meters from source)	**Total Acoustic Power (dB ref. to 10^-12 Watts) (A Weighted)**	**Power Level in dB (A Weighted)**

Very soft Voice	1 NanoWatt	30
Conversational Voice	10 MicroWatts	70
Shouting Voice	1 MilliWatt	90
Auto on Highway	10 MilliWatts	100
Blaring Radio	100 MilliWatts	110
Piano	1 Watt	120
Small Aircraft Engine	3 Watts	125
Pipe Organ	100 Watts	140
75 Piece Orchestra	100 Watts	140
4 Propeller Airplane	1,000 Watts	150
Turbojet Engine	10,000 Watts	160
Ram-Jet Engine	100,000 Watts	170

Dynamic Range

Below is a table of common values of audio system resolution and their associated dynamic ranges:

Number of Bits of Resolution	Theoretical Maximum Dynamic Range
4 bits	24 dB
8 bits	48 dB
16 bits	96 dB
20 bits	120 dB
24 bits	144 dB*
32 bits	192 dB*

*Note: These values are not achievable in practice due to the physics associated with the thermal noise floor of electronic amplification systems at room temperatures.

Audio Frequency Spectrum

The following is a listing of some common audio sources and the portion of the audio spectrum that they typically occupy, including their harmonics:

Audio Source	**Fundamentals + Harmonics**	**Fundamental Only**
Accordion	130 Hz to 15 kHz	130 Hz to 1.8 kHz

Bass Drum	50 Hz to 5 kHz	----
Bassoon	65 Hz to 9 kHz	65 to 650 Hz
Bass Tuba	40 Hz to 7 kHz	40 Hz to 375 Hz
Bass Viola	45 Hz to 8 kHz	45 to 300 Hz
Cello	70 Hz to 14 kHz	70 Hz to 900 Hz
Clarinet (Soprano)	150 Hz to 14 kHz	150 Hz to 1.7 kHz
Cymbals (14 inch)	300 Hz to 17 kHz	----
Female Speech*	165 Hz to 10 kHz	165 to 255 Hz
Flute	250 Hz to 14 kHz	250 Hz to 2.5 kHz
Foot Steps	80 Hz to 15 kHz	----
French Horn	70 Hz to 6 kHz	70 to 825 Hz
Hand Clapping	100 Hz to 15 kHz	----
Harmonica	450 Hz to 15 kHz	450 Hz to 1.3 kHz
Jingling Keys	1.5 kHz to 14 kHz	----
Male Speech*	85 Hz to 8 kHz	85 to 180 Hz
Oboe	250 Hz to 15 kHz	250 Hz to 1.7 kHz
Piano	30 Hz to 6 kHz	30 Hz to 4.2 kHz
Piccolo	500 Hz to 15 kHz	500 Hz to 3.8 kHz
Pipe Organ	16 – 32 Hz to 15 kHz	16 – 32 Hz to 8 kHz
Room Noise	30 Hz to 18 kHz	----
Snare Drum	80 Hz to 15 kHz	----
Timpani Drums	50 Hz to 4.5 kHz	----
Trombone	80 Hz to 8 kHz	80 Hz to 500 Hz
Trumpet	180 Hz to 9 kHz	180 Hz to 900 Hz
Violin	190 Hz to 15 kHz	190 Hz to 3 kHz

*Note: Frequency response is specified for mature & healthy adults

Human Hearing Frequency Response vs. Age

(0 dB is referenced to the 20 – 39 year old age group)
(Values are Averages for both Men and Women)

Age	400 Hz	1 kHz	2 kHz	4 kHz	10 kHz
<19	0 dB	0 dB	0 dB	+1 dB	+3 dB
20-29	0 dB	0 dB	0 dB	0 dB	0 dB
30-39	- 1 dB	- 2 dB	- 2 dB	- 3 dB	- 6 dB
40-49	-2 dB	- 3 dB	- 5 dB	-9 dB	-15 dB
50-59	- 4 dB	- 7 dB	- 13 dB	- 20 dB	- 30 dB
60-69	- 5 dB	- 12 dB	- 21 dB	- 32 dB	- 45 dB

Musical Scale

The following table provides the frequencies of four octaves of the tempered musical scale (1/2 step between notes) rounded in integers:

Note	Frequency	Note	Frequency
A	110	A (above middle C)	440
A# (B flat)	117	A# (B flat)	466
B	123	B	494
C (low C)	131	C (high C)	523
C# (D flat)	139	C# (D flat)	554
D	147	D	587
D# (E flat)	156	D# (E flat)	622
E	165	E	659
F	175	F	698
F# (G flat)	185	F# (G flat)	740
G	196	G	784
G# (A flat)	208	G# (A flat)	831
A (below middle C)	220	A (above high C)	880
A# (B flat)	233	A# (B flat)	932
B	247	B	988
C (middle C)	262	C	1,047
C# (D flat)	277	C# (D flat)	1,109
D	294	D	1,175
D# (E flat)	311	D# (E flat)	1,245
E	330	E	1,319
F	349	F	1,397

F# (G flat)	370	F# (G flat)	1,480
G	392	G	1,568
G# (A flat)	415	G# (A flat)	1,661
A (above middle C)	440	A	1,760

Note 1: Standard Pitch is based on the tone "A" of 440 Hz. With this standard, the frequency of Middle C should actually be 261.626 Hz.
Note 2: The entire Musical Scale from C_0 (16.35 Hz) to D9# (9,956.06 Hz) are available as presets in the "Make Waves" Generator (Edit Menu).

A above Middle C Frequencies vs Time

Designation	**Frequency in Hertz (Hz)**	**Time Period (Historical)**	**Standard Name**
Standard Pitch (Stuttgart Pitch)	440	1936 on Forward	ISO - 16
French Standard	435	1859	-
Concert Pitch	440 - 444	Varies	-
Baroque Pitch	415	1600 to 1750	Baroque
Chorton Pitch	466	1618 to 1795	Choir Pitch (Oxford)

Hard Drive Recording Space Consumption

How much hard drive space will you need for your next recording? This handy chart should get you in the ballpark.

Digitization Disc Space Consumption as a function of Recording Mode and Sample Rate @ 16 bit resolution	
Sample Rate & Recording Mode	**Mbytes per Minute**
192 kHz Monophonic	22.500
192 kHz Stereophonic	45.000

96 kHz Monophonic	11.250
96 kHz Stereophonic	22.500
48 kHz Monophonic (Pro-Audio)	5.760
48 kHz Stereophonic (Pro-Audio)	11.520
44.1 kHz Monophonic	5.292
44.1 kHz Stereophonic (Compact Disc)	10.584
22.05 kHz Monophonic	2.646
22.05 kHz Stereophonic	5.292
16.000 kHz Monophonic (Forensics)	1.920
16.000 kHz Stereophonic (Forensics)	3.840
11.025 kHz Monophonic	1.323
11.025 kHz Stereophonic	2.646

Note 1 - Values are given for one process only (such as recording).
Note 2 - Values are given for 16-bit resolution only.
Note 3 - Multiply the above storage rates X 1.5 for 24-bit recording.
Note 4 - Windows currently operates with a 2 Gigabyte limit on .wav file size.

Compact Discs

There are a number of standards for data contained on Compact Discs. They are as follows:

Type	Application	Comments
Red Book	CD Audio / Compact Disc	PCM, 44.1 kHz sampling, 16 bit x 2 channels, 588

		bits/frame, 192 bits/frame for the audio stream.
Yellow Book	Computer Data	Data structure based on ISO 9660
White Book	Video CD	MPEG audio/video track encoding
Blue Book	Enhanced Music CD (Audio + Data)	Structure similar to ISO 9660
Orange Book	CD-MO, CD-R & CD-RW	Magneto Optical / CD Write Once / CD Re-Writable
Photo CD Book	Photographs	Based on CD-I Bridge spec.
Multi-session CD	Multiple Session not recordable	Data structure based on ISO 9660

78 RPM Record Turnover Frequency Chart

Type, Brand, or Process	**Turnover Frequency**
Acoustical Recordings	0 Hz
Columbia (1925 - 1937)	200 Hz
Victor (1925 - 1937)	200 Hz
Westrex	200 Hz
Decca (1935 - 1949)	250 Hz
EMI	250 Hz
English Columbia	250 Hz
HMV (1931)	250 Hz
EMI (1931)	250 Hz
London	250 Hz
Blumlein	250 Hz
Columbia (1938 – End)	300 Hz
BSI	350 Hz

Capitol	400 Hz
Mercury	400 Hz
Brunswick	500 Hz
Decca (1925 – 1929)	500 Hz
Edison Laterals (1929)	500 Hz
MGM	500 Hz
Parlophone	500 Hz
Victor (1938 – 1952)	500 Hz
629	629 Hz

Note: Many of these Turnover Curves can be found as presets in the VPP or the Paragraphic EQ.

LP Equalization Chart for Records by Label (Phonographic)

(Prior to the EQ Standardization in 1955)

Label (Manufacturer)	Turnover Frequency in Hz	Roll-off at 10 kHz in dB
Angel	500	-13.7
Audio Fidelity	500	-16
Arizona	500	-13.7
Bach Guild	500	-16
Bartok	629	-16
Bethlehem	500	-13.7
Boston	500	-16
Caedmon	629	-16
Capitol	400	-12
Capitol-Cetra	400	-12
Cetra-Soria	500	-16
Classic Editions	500	-13.7
Clef	500	-13.7
Colosseum	400	-12
Columbia	300	-16
Concert Hall	400	-12
Decca	400	-12
Decca FFRR (1951)	300	-14
Decca FFRR (1953)	450	-11

Ducretet-Thompson	450	-11
EMS	375	-12
Epic	500	-16
Esoteric	400	-12
Folkways	500	-16
Haydn Society	500	-16
HMV	500	-16
Kapp	500	-13.7
London	450	-11
London International	450	-11
Lyrichord	500	-16
McIntosh	500	-13.7
Mercury	400	-12
MGM	500	-13.7
Montilla	500	-13.7
New Jazz	500	-13.7
Norgran	500	-13.7
Oceanic	500	-16
Oiseau-Lyre	500	-8.5
Overtone	500	-16
Polymusic	500	-16
Prestige	500	-13.7
RCA Victor (until 1953)	500	-13.7
Remington	500	-16
Riverside	500	-13.7
Romany	500	-13.7
Savoy	500	-13.7
Urania	500	-16
Vanguard	400	-12
Vox	400	-16
Westminster	400	-16

Note: Many of these curves can be found as presets within the Virtual Phono Preamp (VPP).

LP Equalization Curves by Curve Name (Phonographic)

The following is a listing of LP turnover frequencies and roll-off attenuation values for the various equalization curves that were used for playback by the phonographic industry (many of which can be found as presets in the Diamond Cut **V**irtual **P**hono **P**reamp {VPP}):

Equalization Curve	**Turnover Frequency**	**Roll-off dB @ 10 kHz**
AES	400 Hz	- 12 dB
Columbia LP	300 Hz	- 16 dB
EMI LP	500 Hz	- 10.5 dB
ffrr (1949)	250 Hz	- 5 dB
ffrr (1951)	300 Hz	- 14 dB
ffrr (1953)	450 Hz	- 11 dB
NAB	500 Hz	- 16 dB
NARTB	500 Hz	- 12 dB
RCA Early Orthophonic	500 Hz	- 11 dB
RCA New Orthophonic	500 Hz	- 13.7 dB
RIAA	500 Hz	-13.7 dB

RIAA Curve Table of Values

Frequency in Hz	**Level in dB referenced to 0 dB @ 1 kHz***	**Frequency in Hz**	**Level in dB referenced to 0 dB @ 1 kHz***
20	+ 19.3	800	+ 0.7
30	+ 18.6	1,000	0.0 *
40	+ 17.8	1,500	- 1.4
50	+ 17.0	2,000	- 2.6
60	+ 16.1	3,000	- 4.8
80	+ 14.5	4,000	- 6.6
100	+ 13.1	5,000	- 8.2
150	+ 10.3	6,000	- 9.6
200	+ 8.2	8,000	- 11.9
300	+ 5.5	10,000	- 13.7
400	+ 3.8	15,000	- 17.2
500	+ 2.6	20,000	- 19.6

Note: The RIAA EQ system operates in Constant Amplitude mode below the 500 Hz Turnover Frequency. It also operates in Constant Amplitude mode above the 2120 Hz Rolloff Frequency. The system operates in Constant Velocity mode between the Turnover and the Rolloff Frequencies.

Record Styli Sizes and Types

Record Type	Stylus Size (mil)	Stylus Type
16 inch Transcriptions	2.5	TE
16 inch Transcriptions (very late)	2.0	E
33.3 RPM Monophonic LP	1.0	E
33.3 RPM Stereophonic LP	0.7	E
33.3 RPM Badly worn LP	1.5	TE
33.3 RPM Early Mono LPs	1.5	TE
33.3 RPM CD-4 Quadraphonic LPs	0.2	Shibata
45 RPM Monophonic	1.0	E
45 RPM Stereophonic	0.7	E
1931 to 1935 RCA Pre-Grooved Home Recordings	4.0 – 5.0	S
1930's (late) Lateral 78 RPM Shellac Discs	2.8	TE
Acetate & Aluminum Instantaneous Discs	6.0	TE
Acoustical 78s (very early)	4.0	TE
Acoustical Wide Groove 78 RPM Lateral Discs	3.8	TE
Edison 80 RPM Diamond Discs	3.0 – 3.7	S or C
Edison Blue Amberol Cylinders	3.0 – 4.2	S
Edison Wax Amberol Cylinders	4.2	S
Edison White & Brown Wax, Concert, & Gold Molded Cylinders	7.4	S
Electrical Recordings (Shallow Groove)	2.0	TE
General Purpose 78 RPM	3.0	TE
Metal Stampers*	Depends	BR
Narrow Groove 78s such as Polydor	2.4	TE
Pathe' 78s	3.7	TC
Pathé Etched-label up to 14 inches in diameter	8.0	S
Pathe' Etched-label greater than 14 inches in diameter	16.0	S
Pre-1935 Lateral Cut Electrical 78s	3.3	TE
Standard Groove 78 RPM Discs	3.0	TE

Transcription Recordings	2.3	TE
Transcription. 1930's and 1940's 16 inch Acetates	2.6	TE
Wagner-Nichols Records	0.5	TE

Stylus Type Key: BR = Bi-Radial, C = Conical, TC = Truncated Conical, E = Elliptical, TE = Truncated Elliptical, S = Spherical

* Note: When "stampers" are played on a conventional turntable equipped with a bi-radial stylus, you will need to use the Diamond Cut File Reversal feature so that it can be converted to forward play.

Record Speed Chart (RPM)

Record Type	**Speed (RPM)**
45s (Victor and others)	45
Audiobooks for the Blind (10 inch)	8.33
Berliner 7 inch Records	70
Berliner (pre-1900)	57 to 72
Berliner, Victor, Zonophone (early)	71.3
Early Microgroove (some)	16.66
Edison Black (wax) Amberols (2 min)	160
Edison Blue Amberols (4 min)	160
Edison Brown Wax Cylinders (early)	125 to 144
Edison Brown Wax Cylinders (1892 – 1899)	125
Edison White Wax Cylinder (1888 – 1892)	100
Edison Brown Wax Cylinders ("New Process" – 1900)	144
Edison Concert Cylinders	100
Edison Gold Molded Cylinders	160
Edison Diamond Discs (Vertical Cut)	80
Electrical Era 78s (Europe)	77.92
Electrical Era 78s (US)	78.26
Electrical Era (Edison Lateral "Thin-Cut" – Needle Type)	78.8
Pathe' (some)	90
Spoken Word	16.66
Vertical Cut (Pathe´, Brunswick, Okey, Columbia)	80
Victor Acoustical (1908 to 1925)	76.6
Victor Acoustic (early), Berliner	71.3
Victor Acoustical (most)	76.59
Vinyl LPs (Columbia and others)	33.33
Zonophone	71.29

Fractional Speed Record Transfers

You can use the Diamond Cut Change Speed filter to provide Fractional Speed transfer capability from a 45 RPM turntable. Some important Change Speed ratios are as follows:

1. 45 RPM to 78.26* RPM - Use +73.7 % speed change
2. 45 RPM to 78.8 RPM - Use +75.1 % speed change
3. 45 RPM to 80 RPM - Use +77.1 % speed change
4. Other values can be simply calculated by applying ratio-proportions.

*Note: (Actually it is 78.26086957 which is a 46:1 standard gear reduction from a 60 Hz Line operated Synchronous Motor turning at 3600 RPM.)

Stroboscope Chart (Phonograph)

The following is a chart that you can use to create your own phonograph strobe disc using common line frequencies and RPM values:

RPM	# of Divisions for 50 Hz	# of Divisions for 60 Hz
16	375	450
33.33	180	216
45	133	160
78.26	77	92
80	75	90

Note 1: Actually, two pulses of light are produced per cycle of the line by the power line. But, for improved visibility, it is better to use every other pulse to light up the strobe as is reflected by the chart above.

Note 2: Printable Stroboscope Disc Metafiles can be found you're your Diamond Cut software documentation. The 50 Hz strobe disc is called "**Strobe50Hz.wmf**" and the 60 Hz version is called "**Strobe60Hz.wmf**".

<Users Documents>/DCForensics10/Strobe50Hz.wmf

or

<Users Documents>/DCForensics10/Strobe60Hz.wmf

Note 3: The most effective illumination for a phonograph strobe disc is a power line operated fluorescent or neon lamp.

Tape Speeds in Inches Per Second (ips)

The following is a listing of common speeds used by tape recorders:

Speed (IPS)	Pro Reel to Reel	Home Reel to Reel	Compact Cassette	Micro Cassette
30	**X**	-	-	-
15	**X**	-	-	-
7 1/2	-	**X**	-	-
3 3/4	-	**X**	-	-
1 7/8	-	-	**X**	**X**
15/16	-	-	-	**X**
15/32*	-	-	-	**X**

* This speed is also used by reel-to-reel analog data recorders.

Rotary Head Tape Recorder Speeds

- **DAT:**
  0.321 ips (8.15 mm / sec)
- **VHS:**
  1.31 ips - SP (Standard Play)
  0.66 ips - LP (Long Play)
  0.44 ips - EP (Extended Play)
- **Beta:**
  1.58 ips (4.0 cm / sec) - Beta I
  0.797 ips (2.0 cm / sec) - Beta II
  0.524 ips (1.33 cm / sec) - Beta III

- **U-Matic:**
  3.75 ips

<u>Audio Connection Standards (Connectors)</u>

(Connectors and Cables)

1. <u>Balanced "XLR" (Cannon) Standard</u>
   A. Pin # 1 = Shield
   B. Pin # 2 = Signal + (Signal Hot)
   C. Pin # 3 = Signal - (Signal Cold)

2. <u>1/4 inch Stereo Phone Plug (TRS) for Balanced Audio Circuits</u>
   A. Tip = Signal + (Signal Hot)
   B. Ring = Signal - (Signal Cold)
   C. Sleeve = Shield

3. <u>1/4 inch Mono Phone Plug (TR) for Unbalanced Audio Circuits</u>
   A. Tip = Signal + (Hot)
   B. Sleeve = Signal - (Shield & Signal Return Path)

4. <u>RCA / Phono Plug</u>
   A. Tip = Signal + (Hot)
   B. Sleeve = Signal - (Shield & Signal Return Path)

5. <u>RCA Phono Plug Color Codes</u>
   A. Red = Audio, Right Channel
   B. White = Audio, Left Channel
   C. Yellow = Composite Video
   D. Orange = Digital Audio (S/PDIF {Sony/Philips Digital Interface Format})
   E. Green = Y (Component Video - YPbPr)
   F. Blue = Cb or Pb (Component Video - YPbPr)

G. Red = Cr or Pr (Component Video - YPbPr)

6. <u>Amphenol 3 Pin Balanced Microphone Connector</u>

A. Pin #1 = Shield
B. Pin #2 = Signal + (Signal Hot)
C. Pin #3 = Signal - (Signal Cold)

7. <u>Amphenol 4 Pin Microphone Connector (Balanced and Unbalanced)</u>

A. Pin #1 = Shield
B. Pin #2 = + (Hot) Unbalanced (Note: Unbalanced output is with respect to the Shield)
C. Pin #3 = Signal + (Hot) Balanced
D. Pin #4 = Signal - (Cold) Balanced

8. <u>DIN 5 Pin Connector (Tape Deck I / O Connector</u>
(13.2 mm diameter)

A. Pin #1 = Right Channel Record Input
B. Pin #2 = Shield & Signal Common)
C. Pin #3 = Right Channel Playback Output
D. Pin #4 = Left Channel Record Input
E. Pin #5 = Left Channel Playback Output

9. <u>1 / 8 inch (3.5 mm) Mono Phone Plug (TR)</u>

A. Tip = + (Hot)
B. Sleeve = - (Shield & Signal Return Path)

10. <u>1 / 8 inch (3.5 mm) Stereo Phone Plug {The type used on most Sound Cards}</u>

A. Tip = Left Channel + (Hot)
B. Ring = Right Channel - (Hot)
C. Sleeve = Shield & Signal Common

11. <u>1 / 8 inch (3.5 mm) Microphone Input Plug (TRS) {The type used on most Sound Cards}</u>

A. Tip = Signal + (Hot)
B. Ring = Phantom Power (~3 to 4 Volts @ ~ 0.75 to 1.5 mA)
C. Sleeve = Shield & Circuit Common

12. <u>Modular Telephone Jack Wiring (- 48 Volt, 4 terminal, 2 line system / United States)</u>

A. Red* or Blue or Blue with White Stripe = Line #1 (negative) (Hot)
B. Green* or White or White with Blue Stripe = Line #1 (positive)
(Common)
C. Yellow* or Orange or Orange with White Stripe = Line #2 - (Hot)
D. Black* or White or White with Orange Stripe = Line #2 + (Common)
* Denotes the most standard color code

13. Standard USB (Universal Serial Bus) Pinout and Color Code
A. Pin #1 = Vbus (5 Volts @ 500 mA) (Red Wire)
B. Pin #4 = Vbus Ground Return Conductor (Black Wire)
C. Pin #3 = Data + (Green Wire)
D. Pin #2 = Data – (White Wire)
Note: Data + and Data – Conductors are a Twisted Pair

13. 3.5 mm Phone style Jack Pinout (same as 1/8 inch phone jacks)
(also referred to as TRRS plugs)
A. Pin #1 (Tip) = Left Channel Output
B. Pin #2 (1st Ring) = Right Channel Output
C. Pin #3 (2^{nd} Ring) = Ground/Common
D. Pin #4 (Sleeve) = Microphone Input

15. Stereo Phono Cartridge Wiring Color Standard
A. Red = Right Channel Signal (Hot)
B. Green = Right Channel Signal Return
C. White = Left Channel Signal (Hot)
D. Black = Left Channel Signal Return

16. 2.5 mm *3/32”) Stereo Phone Plug* {used on some mobile phones}
A. Tip = Left Channel + (Hot)
B. Ring = Right Channel - (Hot)
C. Sleeve = Shield & Signal Common
*Note: This type of connector is sometimes used to deliver dc power to an electronic device

Wire Table

The Wire Table below is useful for calculating losses in power amplifier to speaker system cable connections.

(Standard Annealed Copper)

Wire Gauge in AWG	**Wire Diameter(in Mils)**	**Resistance per Foot in Ohms (@ 20 degrees C)**
0	324.9	0.00009827
1	289.3	0.0001239
2	257.6	0.0001563
3	229.4	0.0001970
4	204.3	0.0002485
5	181.9	0.0003133
6	162.0	0.0003951
7	144.3	0.0004982
8	128.5	0.0006282

9	114.4	0.0007921
10	101.9	0.0009989
11	90.74	0.001260
12	80.81	0.001588
13	71.96	0.002003
14	64.08	0.002525
15	57.07	0.003184
16	50.82	0.004016
17	45.26	0.005064
18	40.30	0.006385
19	35.89	0.008051
20	31.96	0.01015
21	28.46	0.01280
22	25.35	0.016140
23	22.57	0.02036
24	20.10	0.02567

Wire Gauge Rule of Thumb:

Conductor resistivity roughly doubles for every 3 AWG increase.
(Resistivity = 1 / Conductivity)

Note 1: The temperature coefficient of resistance for copper wire $\simeq$ + 0.41% / degree C (4,100 ppm / degree C) referenced to 20 degrees C.
Note 2: The resistance of a 2 conductor cable will be need to be doubled to account for the round trip.
Note 3: Generally, audio power signals are carried with 16 AWG or lower conductors except in some low power 25 or 70 Volt Constant Voltage audio distribution systems (from power amplifiers under 100 Watts).
Note 4: Except in some very unusual circumstances, no wire sizes higher than 16 AWG should be used to carry the output of an audio power amplifier to an audio system's loudspeakers, especially in permanent building installations.
Note 5: C10100 Oxygen-Free Electronic (OFE) copper wire is 99.99% pure copper (Cu) with $\leq$ 0.0005% O_2. It exhibits 1% lower resistance per foot compared to standard Cu wire (which is of little benefit in audio applications).

Telephone Touch Tone Frequency Chart

The Spectrum Analyzer can be used to detect and identify a telephone number dialed when used in conjunction with this chart. Also included are the letters A, B, C, and D, which was used in the US military's Autovon phone system.

1	**2**	**3**	***A***	**697** (687–708)
4	**5**	**6**	***B***	**770** (759–782)
7	**8**	**9**	***C***	**852** (839–865)
*****	**0**	#	***D***	**941** (927–955)
1209 (1191–1227)	**1336** (1316-1356)	**1477** (1455-1499)	**1633** (1609-1658)	**Frequency (Hz)**

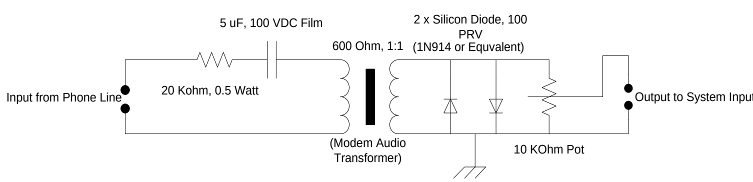
Note 1: The tolerance for these frequencies is +/- 1.5 %, and the range of which is shown in parenthesis. The highest frequency must also be as loud as the lowest frequency, or as much as 4 dB louder (this difference in level is referred to as "twist."
Note 2: Call waiting tone in the US is 440 Hz.
Note 3: Caller ID on call waiting in the US is 2130 + 2750 Hz.

Worldwide Dial Tone Frequencies
(Telephone Signals)

Country	**Freq.1**	**Freq. 2**	**Country**	**Freq.1**	**Freq.2**
Belgium	425 Hz	Not Used	Singapore	270 Hz	320 Hz
France	440 Hz	Not Used	South Korea	350 Hz	440 Hz
Germany	425 Hz	Not Used	Sweden	425 Hz	Not Used
Israel	400 Hz	Not Used	Switzerland	425 Hz	Not Used
Italy	425 Hz	Not Used	Taiwan	350 Hz	440 Hz
Japan	400 Hz	Not Used	U.S.	350 Hz	440 Hz
Netherlands	425 Hz	Not Used	U.K.	350 Hz	440 Hz
Norway	425 Hz	Not Used	Venezuela	425 Hz	Not Used

*Note1: These can be useful as a reference frequency when measuring DTMF signals
Note 2: Further information can be found in an ITU document located at:
http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf

Land Line Telephone Interface Circuit (USA System)



Circuit Diagram

This land line analog interface circuit provides a higher quality signal compared to inductive or acoustic coupling methods. Its frequency response and low distortion are important as well as its ground-loop isolation between the phone line and your recording equipment. Hum levels will be quite low using this interface circuit.
It is very important to note that it is universally illegal to tap the phone line of someone else without a court order. However, in some states it is

permissible to record your own phone line without second party approval. But, some states require two-party consent (both parties) before recording phone conversations. Please research your specific situation based on local and state regulations pertaining to telephone conversation recording.

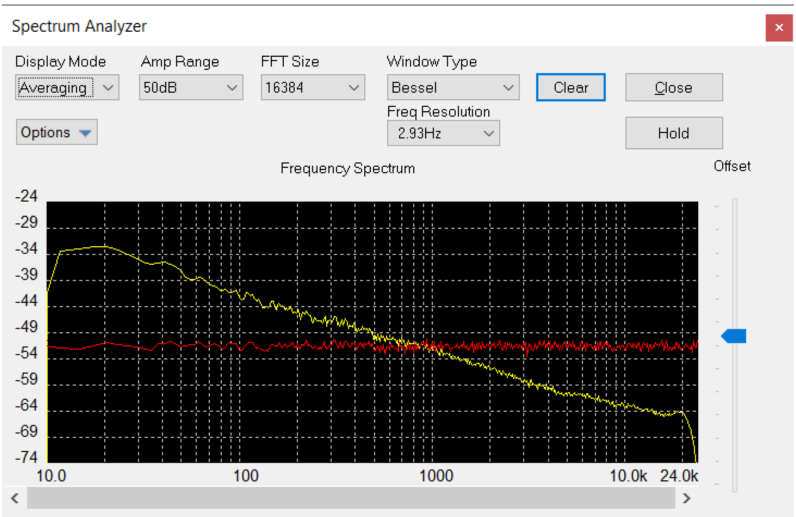
Interface Circuit Notes:

1. Phone polarity input connection does not matter with this system.
2. The 5 uF Capacitor must be a Non-Polarized Film Type, 100 VDC or higher.
3. The two protection diodes can be small glass signal types; power diodes are not necessary. Use type 1N914 or equivalent (100 PRV, 200 mA, axial lead)
4. The 600 Ohm, 1:1 ratio audio transformer can be scavenged from an old computer modem board or purchased on the internet from Digikey, Newark Electronics or other similar outlets. You can use Tamura TTC-294 or equivalent.
5. The phase relationship between the primary and secondary sides of the transformer are not important.
6. The 20 KOhm fixed resistor should be +/- 10 % or better (it can be carbon composition, metal film or metal oxide – the type doesn't matter).
7. The 10 KOhm Potentiometer should also be +/- 10 % and can be of the linear or audio taper variety, but audio taper pots usually give the user easier adjustability. You can use carbon or cermet; both types work well in this application.
8. Adjust the 10 KOhm Potentiometer for the proper input level to the systems sound-card or audio amplifier. Adjust for minimum noise without distortion or overloading.
9. On some systems, you can use the Line Level Input and on others you may have to use the Mic Level Input. It depends on the input Voltage sensitivity of the system to which the Interface Circuit is connected. Experiment to find the best input to use.
10. The input connector preferably should be a modular telephone jack for convenience.
11. The output connector can be a 1/8 inch mini-phone jack.
12. A convenient way to construct this is on a vector style board with 0.1 inch spacing between holes.

13. The circuit can be housed in a metal Altoid, Sucrets, or an old Band-Aid box (or something similar). An insulating spacer should be used to prevent short-circuits between the circuit board and the metal housing.
14. The metal case of the circuit housing can be connected to the point in the diagram designated as the "chassis ground". This will improve the noise rejection ability of the interface circuit to external influences.
15. Care should be taken to assure that none of the electrical components come in contact with the metal housing.

16. Do not handle the un-enclosed circuit board when it is plugged into the telephone line. The telephone ring Voltage can give you a potentially dangerous shock.

White & Pink Noise Spectral Display



Averaging time period = 30 seconds on the Diamond Cut DCArt10.8x Spectrum Analyzer. Red Line = White Noise; Yellow Line = Pink Noise (White (Random) Noise created by the Make Waves Generator. Pink Noise is a conversion made by the White to Pink Noise Converter in the Multifilter.

Random Noise by Color (Various Noise Distributions)

Color (or type)	**Characteristics**	**Slope**	**Density**
White	Equal Energy per unit Hertz	Flat	Uniform
Pink	Equal Energy per unit Octave	-3 dB/Octave	1/F
Brown	Analogous to Brownian Motion	-6 dB/Octave	1/F^2
Red	Another name for Brown Noise	-6 dB/Octave	1/F^2
Blue	Azure Noise (proportional to F)	+3 dB/Octave	F
Green	Pink Noise with Emphasis at 500 Hz	Non-Linear	Somewhat Subjective
Violet	Differentiated White Noise	+6 dB/Octave	F^2
Grey	Psychoacoustic equal loudness Curves	Proportional to SPL	Varies with loudness
Seismic	Random Noise below 20 Hz	-3 dB/Octave	In-audible
Black	Analogous to black body radiation	Silence	Edit/Mute

These various Random Noise distributions by color can be created directly via the Make Waves Generator. They can also be found as white to X color Multifilter Presets portion of the software program.

Common Audio Electron Tubes / Valves

Note: Some of these tube transfer functions are implemented in the Diamond Cut Virtual Valve Amplifier (VVA).

Audio Power Amplification Output Tubes

Tube Type	**Tube Configuration**	**Output Power (Class AB1, 2 tubes used in a Push-Pull topology – typical levels)**	**Comments**
2A3	Power Triode	8 Watts	Directly Heated

			Cathode (Vintage Tube)
6BQ5 / EL-84	Power Pentode	15 Watts	9 pin miniature base
6CA7 / EL-34	Power Pentode	35 Watts	English Design
6K6	Power Pentode	8 Watts	Early Octal Base Power Pentode
6L6GC / 5881	Beam Power Pentode	25 Watts	Very Common Tube
6V6	Beam Power Pentode	12 Watts	Low Power Hi-Fi
6550	Beam Power Pentode	75 Watts	Hi-Pwr Hi-Fi
7027	Beam Power Pentode	30 Watts	Often used in P-P Parallel Configuration
8417	Beam Power Pentode	75 Watts	Often used in P-P Parallel in high-power PA applications
EL-37	Power Pentode	25 Watts	Mid 40s English Hi-Fi Pwr Amplifiers
KT-66	Beam Power Tetrode	30 Watts	English Design Hi-Fi Power Amplifiers
KT-88	Beam Power Tetrode	70 Watts	English Design Hi-Fi Power Amplifiers

Audio Amplification Signal Tubes

Tube Type	**Tube Configuration**	**Key Characteristic**	**Comments**
6C4 / EC90	Single Medium Mu Triode	Cathode Follower	Buffer / Low gain Preamp

6CG7 / 6FQ7	Dual Medium Mu Triode	Phase Inverter / Driver	Used in Power Amplifiers
6EJ7 / EF164	Single Hi Gain Pentode	Wide Bandwidth	Used in Tuners & Low Level Audio Stages
6SL7 / ECC35	Dual Hi Mu Triode	Low Noise / Low Level	Used in Preamplifiers & Mixers
6SN7 / 5692	Dual Medium Mu Triode	Phase Inverter / Driver	Used in Power Amplifiers
12AT7 / ECC81	Dual Hi Mu Triode	RF Mixer / Hi Levels of Harmonic Distortion	Used to produce “tube warmth” via high level of Harmonic Distortion
12AU7 / ECC82	Dual Medium Mu Triode	Phase Inverter / Driver	Used in Power Amplifiers
12AX7 / ECC83	Dual Hi Mu Triode	Low Noise / Low Level	Used in Preamplifiers & Mixers
6267 / EF86	Single Hi Gain Pentode	High Gain / Low Noise	Used in Microphone Preamps

Audio Rectifier Tubes

Tube Type	**Tube Configuration**	**DC Current Capability**	**Comments**
5AR4	Octal base, Indirectly Heated Cathode	270 mA	Hi Power Output Amplifiers
5U4	Octal base, Directly heated Cathode	200 mA	Medium Power Output Amplifiers
5Y3	Octal base, Directly Heated Cathode	140 mA	Low Power Amplifiers
6V4 / EZ-80	9 Pin Miniature Indirectly Heated Cathode	90 mA	Preamps / Tuners / Mixers
6X4	7 Pin Miniature Indirectly Heated	70 mA	Preamps / Tuners / Mixers

	Cathode		
6X5	Octal Indirectly Heated Cathode	80 mA	Preamplifiers / Tuners / Mixers
80	4 Pin, Directly Heated Cathode	140 mA	Vintage Medium Power Amplifiers

Language Translation – German/Spanish (Deutsch/Español)

DCArt10/DC Forensics10 program terminology translations from English to German and Spanish

English	Deutsch	Español
About ***Diamond Cut***	Über ***DC-Art***	Acerca del ***DC-Art***
Accuracy	Genauigkeit	Acuracia
Add Marker	Marker hinzufügen	Añadir marcador
Advanced Controls		Controles avanzados
Amplitude (dB)	Amplitude (dB)	Amplitud (dB)
Arrange Icons	Symbole anordnen	Arreglo de Icones
Attack (mSec)	Anstiegsflanke (mS)	Atacar (mSec)
Attenuation	Dämpfung	Atenuacion
Average	Mittelwert	Promedio
Averaging Filter	Mittelwertfilter	Determinacion del Promedio del Filtro
Band Pass	Bandpass	Banda de frecuencias alimentadas
Bandpass Filter	Bandpaßfilter	Filtro de Paso de Banda
Bandwidth	Bandbreite	Anchura de Banda
Bright	Hell	Brillante
Bypass	Bypass	Sobrepasar
Cancel	Abbrechen	Cancelar
Cascade	Fenster kaskadieren	Cascada
CD Preparation	CD Vorbereitung	Preparacion del CD
Center Frequency	Mittenfrequenz	Centro de Frecuencia
Change Speed	Geschwindigkeit ändern	Cambio de velocidad
Chop File into pieces	Datei aufteilen	Recortar el fichero en

		pedazos
Clear all Markers	Alle Marker löschen	Despejar todos los marcadores
Click Filter	Klickfilter	Golpecito seco en el Filtro
Clicks / Second	Klicks / Sekunde	Golpecitos en seco / Segundo
Close	Schließen	Fin
Close Destination	Zieldatei schließen	Cerrar Destinacion
Close Source	Quelldatei schließen	Cerrar las fuente de datos
Contents	Inhalt	Contenidos
Continuous Noise	Dauerrauschen	Sonido Continuo
Continuous Noise Filter	Rauschfilter	Filtrador Continuo de Sonido
Copy	Kopieren	Copia
Crossfade	Crossfade	Interseccion del aumento / disminucion del volumen alto
Curved	Kurve	Curvado
Cut	Ausschneiden	Cortar
Dark	Dunkel	Oscuro
DC-Art Progress	***DC-Art*** Prozeß	Progreso del ***DC-Art***
Decay	Ausklingzeit	Decaimiento
DeEsser	DeEsser	Normalizador de seseos
Delete	Löschen	Suprimir
Delete Files	Dateien löschen	Suprimir ficheros / archivos
Demo	Demo	Demostracion
Detail	Detail	Detalle
Device I/O Selection	Device I/O Auswahl	Seleccion del Dispositivo I / O
Drive	Drive	Impulsar
Dry	Trockenes Signal	Seco
Dynamic Noise	Dynamisches Rauschen	Sonido Dinamico
Dynamic Noise Filter	Dynamisches Rauschfilter	Filtro Dinamico de Sonido
Dynamics Processor	Dynamikprozessor	Procesor Dinamico
Early Level	Vorreflexion	Primera reverberacion de Amplificacion
Edit	Bearbeiten	Editar
Effects	Effekte	Efectos
Ending Pitch	Endtonhöhe	Terminando el Grado de Inclinacion
Even Only	Nur geradzahlige	Paridad Par
Exciter	Exciter	Excitador
Exit	Beenden	Salida
Expander/Gate	Expander/Gate	Ensanche / Puerta
Fade In	Einblenden	Aumento de Volumen
Fade Out	Ausblenden	Disminucion de Volumen
Fast Forward	Schneller Vorlauf	Advanzar / adelantar
File	Datei	Fichero / Archivo
File Conversions	Dateikonvertierungen	Traducciones del fichero
File Information	Dateiinformation	Informacion de Archivo

File Name	Dateiname	Nombre del fichero / archivo
File Toolbar	Datei Werkzeugleiste	Archivo de paneles
Files of Type	Dateityp	Modelos de ficheros
Filter	Filter	Filtro
Filter Freq.	Filterfrequenz	Filtro de frecuencia
Filter Harmonics	Oberwellen filtern	Filtro Armonicos
Filter Slope	Filterkurve	Filtro Inclinado
Filter Toolbar	Filter Werkzeugleiste	Filtro de instrumentos
Find and Mark Silent Passages	Stille Passagen finden und markieren	Encontrar y marcar los pasages silenciosos
Frequency	Frequenz	Frecuencia
Frequency Response	Frequenzgang	Frecuencia de respuesta
Frequency Spectrum	Frequenzspektrum	Frecuencia de espectros
From Mono to	Von Mono nach	Desde el Mono al
From Stereo to	Von Stereo nach	Desde el Stereo al
Full Range	Volle Bandbreite	Margen Entero
Gain	Verstärkung	Amplificacion
Gain Change	Verstärkungsänderung	Cambio de Volumen
Gain Normalize	Pegel normalisieren	Regularizar el sonido
Graphic Equalizer	Graphischer Equalizer	Equalizador Grafico
Harmonic Reject	Harmonische ausfiltern	Reyeccion Armonica
Harmonic Reject Filter	Oberwellen Unterdrückungsfilter	Filtro Armonico de Reyeccion
Help	Hilfe	Ayuda
High Frequency	hohe Frequenz	Frecuencia Alta
High Pass	Hochpass	Paso Alto
Highlight Marked Area	Markierten Bereich anwählen	Destacar las Areas Marcadas
Impulse Filter Parameters	Impulsfilter	Filtrar Parametros de Impulso
Impulse Noise	Impulsrauschen	Sonido de Impulso
Keep Residue	Rest behalten	Guardar residuo (resto)
Large	Groß	Grande
Left	Links	Mando izquierdo
Look In	Hineinsehen	Mirar En
Low Frequency	niedrige Frequenz	Frecuencia baja
Low Pass	Tiefpass	Avance de alimentacion de frecuencia baja
Lowpass Filter	Tiefpaßfilter	Filtro de pasaje bajo
Make Destination the Source	Ziel zur Quelle machen	Convertir los datos salidos del Fichero en datos nuevos de entrada de Fichero
Make Waves	Wellenformdarstellung	Hacer ondas
Marker	Marker	Marcador
Maximum Harmonic	Maximale Oberwelle	Armonia Maxima
Median	Zentralwert	Media
Median Filter	Mittelungsfilter	Filtro Medio
Medium	Mittel	Mediano

Mix	Mischung	Mixto
Mono	Mono	Mono
Mute	Stummschalten	Mudo
No	Nein	No
Noise Threshold	Rauschschwelle	Umbral de Sonido
Notch	Notchfilter	Hindumieno
Octave	Oktave	Octavo
Odd Only	Nur ungeradzahlige	Paridad impar
Open	Öffnen	Abrir
Open as Read Only	Schreibgeschützt öffnen	Abrir tal como se lee - Solo
Open Destination	Zieldatei öffnen	Salida abierta de archivo
Open Playlist	Spielliste öffnen	Lista de titulos
Open Source	Quelldatei öffnen	Abrir la fuente de datos
Operating Point (Quiescent Point)	Arbeitspunkt	Punto de Ininiazion
Optimization	Optimierung	Optimizacion
Output	Ausgang	Salida
Output Mix	Ausgangsmix	Mezcla de salidas
Overload	Übersteuerung	Sobrecarga
Paragraphic Equalizer	Parametrischer Equalizier	Equalizador Paragrafico
Paste	Einfügen	Pegar
Pause	Pause	Pausa
Pause File	Datei anhalten	Pausa del fichero / archivo
Pentode	Pentode	5 Elementos
Percent Done	Prozent erledigt	Porcentage acabado
Pitch Range	Tonhöhenbereich	Margen del Grado de Inclinacion
Play Controls	Transportkontrolle	Uso de los Controles
Play File	Datei spielen	Activar el fichero
Play Looped	Endlosschleife	Ejecutar el registro y repetir
Preferences	Voreinstellungen	Preferencias
Preview	Vorschau	Vostazo/Chequeo
Quantize for CD Audio	Für CD Audio quantisieren	Cuantificar para el audio CD
Ratio	Verhältnis	Proporcion
Record	Aufnahme	Registro
Record File	Datei aufnehmen	Registro del fichero / archivo
Reflections	Reflexionen	Reflejos
Release (mSec)	Abfallzeit (mS)	Librar (mSec)
Reset	Zurücksetzen (Reset)	Reajustar
Reset Levels	Pegel zurücksetzen	Reajustar los Niveles
Restoring a Recording	Aufnahme restaurieren	Restaurando el Registro
Restoring the Demo	Demo restaurieren	Restauracion del muestreo
Reverb	Hall	Reverberacion / Eco
Reverse	Rückwärts	Inverso
Reverse File	Datei rückwärts	Invertir fichero / archivo

	abspielen	
Rewind	Zurückspulen	Rebobinar
Rewind File	Datei zurückspulen	Rebobinar el fichero / archivo
Right	Rechts	Mando derecho
Room Size	Raumgröße	Tamaño de espacio
Run Effect	Effekt abspielen	Comenzar el funcionamiento de Efecto
Run Filter	Filter anwenden	Comenzamiento del filtro
Save Source as	Gleiche Quelldatei wie	El mismo dato que el
Sample Noise	Quantisierungsrauschen	Ejemplo de Sonido
Samples	Samples	Ejemplos
Save	Speichern	Guardar
Save as Type	Speichern als	Guardar tal como se ha mecanografiado
Save Destination as	Zieldatei spiechern als	Guardar destinacion como
Save In	Speichern in	Guardar En
Search for Help on	Suche Hilfe über	Busqueda de ayuda en
Select all	Alles auswählen	Seleccionar todo
Shape	Form	Form / Forma
Shift Threshold	Schwellwert verschieben	Cambio del Margen
Small	Klein	Pequeño
Source	Quelle	Fuente de Datos
Spectrum	Spektrum	Spektrum / Espectros
Spectrum Analyzer	Spektrumanalysator	Analizador de espectros
Speed	Geschwindigkeit	Velocidad
Starting Pitch	Anfangstonhöhe	Comienzo de Frecuencia
Status Bar	Zustandsleiste	Condicion de los paneles
Status Toolbar	Zustandsleiste	Compartimiento / Tablero de los paneles
Stereo	Stereo	Stereo
Stereo Reverse		Invertir el Stereo / retroceder atras Stereo
Stop File	Datei stoppen	Parar el fichero / archivo
Straight Line	Gerade	Linea Recta
Sweet	Süß	Agradable
Sync Files	Dateien synchronisieren	Sincronizacion de archivos
Threshold	Schwellwert	Margen
Tile	Fenster anordnen	Diseño de Azulejos
Time Offset	Zeitversatz	Cambio de tiempo
Tip of the Day	Tip des Tages	Consejo del dia
Total Clicks Processed	Anzahl der bearbeiteten Klicks	Total de golpecitos secos (pinchazos/clicks) procesados
Total Samples to Process	Anzahl der zu bearbeiteten Samples	Total de ejemplos para procesar
Tracking	Gleichlauf	Rastreando
Transformer Coupled	Transformator	Transformador Acoplado de

Class AB	gekopplete Class AB	Clase AB
Triode	Triode	Triodo
Tube Type/Configuration	Röhrentyp/Konfiguratio n	Clase de tubo / configuracion
Undo	Rückgängig	Deshacer
Very Large	Sehr groß	Muy Grande
View	Anzeigen	Vista
Vinyl LP	Vinyl LP	Vinilo LP
Virtual Valve Amplifier	Virtueller Röhrenverstärker	Amplificador de valvula virtual
Warm	Warm	Calido
Wet	Effektsignal	Mojado
Window	Fenster	Ventana
Yes	Ja	Si
Zoom to Markers	Zoom zwischen den Markern	Enforcar rapidamente
Zoom In	Zoom In	Enforcar
Zoom Out	Zoom Out	Desenfocar
# of Filters	Filteranzahl	Numero de filtro
% Change	% Veränderung	Porcentage de Cambio
% Selected Time	% Zeit	Porcentage del Tiempo seleccionado
2 Stage Class A	Zweistufige Class A	2 Procesos - Clase A
2 Stage Class AB	Zweistufige Class AB	2 Procesos - Clase AB

German Translation: Courtesy of Konstantin Themelidis
Spanish Translation: Courtesy of Monica Sanz Aznar (Monica Hash)

Preset Listings

10 Band Graphic EQ (see Graphic Equalizer – 10 Band)

20 Band Graphic EQ

1. Alto Vocal Boost
2. Alto Vocal Cut
3. Baritone Vocal Boost
4. Baritone Vocal Cut
5. Bass Vocal Boost
6. Bass Vocal Cut
7. Bright Brass
8. De-Boom
9. Deep Bass
10. Fat Bass
11. Flat Response
12. HF Boost with Shelf
13. HF Cut with Shelf
14. Less "Ess"
15. Low Mid Range Bump-Up
16. Low Mid Range Cut-Down
17. Mid-band Bump
18. Mid-band Dip
19. Midrange Bump-Up
20. Midrange Cut-Down
21. Sad Face
22. Smiley Face
23. Soft Brass
24. Soprano Vocal Boost
25. Soprano Vocal Cut
26. Sparkling Brilliance
27. Tenor Vocal Boost
28. Tenor Vocal Cut
29. Ultrasonic Pierce
30. Set All Sliders to Maximum Position
31. Set All Sliders to Middle Position
32. Set All Sliders to Minimum Position

30 Band Graphic EQ

1. 1 kHz Slot Filter
2. 1.3 kHz Slot Filter
3. 1.6 kHz Slot Filter
4. 10 kHz Slot Filter
5. 100 Hz Slot Filter
6. 125 Hz Slot Filter
7. 13 kHz Slot Filter
8. 16 kHz Slot Filter
9. 160 Hz Slot Filter
10. 2 kHz Slot Filter
11. 2.5 kHz Slot Filter
12. 20 kHz Boost Filter
13. 200 Hz Slot Filter
14. 25 Hz Slot Filter
15. 250 Hz Slot Filter
16. 31 Hz Slot Filter
17. 3.1 kHz Slot Filter
18. 320 Hz Slot Filter
19. 4 kHz Slot Filter
20. 40 Hz Slot Filter
21. 400 Hz Slot Filter
22. 5 kHz Slot Filter
23. 50 Hz Slot Filter
24. 500 Hz Slot Filter
25. 6.2 kHz Slot Filter
26. 62 Hz Slot Filter
27. 640 Hz Slot Filter
28. 8 kHz Slot Filter
29. 80 Hz Slot Filter
30. 800 Hz Slot Filter
31. 1 kHz Notch Filter
32. 1.3 kHz Notch Filter
33. 1.6 kHz Notch Filter
34. 10 kHz Notch Filter
35. 100 Hz Notch Filter
36. 125 Hz Notch Filter
37. 13 kHz Notch Filter
38. 16 kHz Notch Filter
39. 160 Hz Notch Filter
40. 2 kHz Notch Filter
41. 2.5 kHz Notch Filter
42. 20 kHz Boost Filter
43. 200 Hz Notch Filter
44. 25 Hz Notch Filter
45. 250 Hz Notch Filter
46. 31 Hz Notch Filter
47. 3.1 kHz Notch Filter
48. 320 Hz Notch Filter
49. 4 kHz Notch Filter
50. 40 Hz Notch Filter
51. 400 Hz Notch Filter
52. 5 kHz Notch Filter
53. 50 Hz Notch Filter
54. 500 Hz Notch Filter
55. 6.2 kHz Notch Filter
56. 62 Hz Notch Filter
57. 640 Hz Notch Filter
58. 8 kHz Notch Filter
59. 80 Hz Notch Filter
60. 800 Hz Notch Filter
61. Alternating Phase 1
62. Alternating Phase 2
63. Alternating Phase Great 1
64. Alternating Phase Great 2
65. Alternating Phase Small 1
66. Alternating Phase Small 2
67. Anti Speech Filter 1 (Steep)
68. Anti Speech Filter 2
69. Bass Boost
70. Bass Boost with Shelf
71. Bass Boost 2

72. Bass Cut
73. Bass Cut 2
74. Bass Cut with Shelf
75. Brilliance (Treble Boost)
76. Flat Response
77. Low Bass Boost
78. Low Bass Cut
79. Midrange Bump
80. Midrange Dip
81. Negative Slope
82. Negative Slope with 2 Shelves
83. Randomized Pass Band
84. Rumble & Hiss Filter
85. Rumble Filter
86. Sad Face
87. Sad Face with 2 Shelves
88. Smiley Face
89. Smiley Face with 2 Shelves
90. Speech Filter 1
91. Speech Filter 2
92. Speech Filter 3
93. Speech Filter 4
94. Speech Filter 5
95. Speech Filter 6
96. Speech Filter 6 (Bottom & Top End Emphasis)
97. Treble Boost with Shelf
98. Treble Cut
99. Treble Cut with Shelf
100. Set All Sliders to Maximum Position
101. Set All Sliders to Middle Position
102. Set All Sliders to Minimum Position

Adaptive Filter
1. Basic Forensics Adaptive Filter
2. Basic Forensics Adaptive Filter with Unity Convergence
3. Basic Normalized Adaptive Filter
4. Binaural Filter with Left Channel Reference
5. Binaural Filter with Right Channel Reference
6. Sharp Adaptive Filter with delayed Monophonic Reference
7. Warbling Tone Rejection Filter

Adaptive Filter (Legacy Adaptive Filter)
1. Basic Forensics Adaptive Filter & - 5 dB Threshold
2. Basic Forensics Adaptive Filter & - 10 dB Threshold
3. Basic Forensics Adaptive Filter & - 15 dB Threshold
4. Basic Forensics Adaptive Filter & - 20 dB Threshold
5. Basic Forensics Adaptive Filter & - 20 dB Threshold
6. Basic Forensics Adaptive Filter & - 25 dB Threshold
7. Basic Forensics Adaptive Filter & - 30 dB Threshold
8. Basic Forensics Adaptive Filter & - 40 dB Threshold
9. Basic Forensics Adaptive Filter & - 50 dB Threshold
10. Basic Forensics Adaptive Filter & - 100 dB Threshold

Auto Voice Filter
1. Default
2. Heavy Attenuation
3. Maximum Attenuation
4. Minimum Attenuation
5. Moderate Attenuation
6. Nominal Attenuation

Averaging Filter
1. Heavy Handed Averaging Filter
2. Half Maximum Value Averaging Filter
3. Maximum Value Averaging Filter
4. Simple Cylinder Recording De-Hisser
5. 1950's Jukebox Sound
6. Light Surface Noise Remover

Band-pass Filter
1. 1930 Vintage Table Top Radio
2. 1950's Vintage Juke Box
3. AM Pocket Transistor Radio
4. Cardiac Sonograph Filter
5. Loud Walkman heard by person nearby
6. Megaphone
7. Metal Horn based Acoustical Phonograph
8. Modern "High-End" Audio System
9. Modern Cheap Table Top Radio
10. Night Club as heard in Parking Lot
11. Olde Acoustic Phonograph
12. Public Address System at Outdoor Event
13. Speech Filter
14. Stereo System heard in adjacent room
15. Telephone "off-the-hook" sound

Big Click Filter
1. Insensitive
2. Insensitive with De-Thumper
3. Most Insensitive
4. Most Insensitive with De-Thumper
5. Most Sensitive
6. Most Sensitive with De-Thumper
7. Nominal Sensitivity
8. Nominal Sensitivity with De-Thumper
9. Sensitive
10. Sensitive with De-Thumper

Brick Wall Filter
1. 1000 Hz Band-Stop Filter
2. 200 Hz High-pass Brick wall filter
3. 5000 Hz Low-pass Brick Wall Filter
4. 6000 Hz Low-pass Brick Wall Filter
5. 9000 Hz Low Pass Brick Wall Filter
6. Alpha Brainwave Bandpass Filter
7. Beta Brainwave Bandpass Filter

8. Theta Brainwave Bandpass Filter
9. Theta to Beta Brainwave Bandpass Filter
10. Basic Forensics Speech Filter
11. 60 Hz Band-stop Filter
12. Gentle Slope Forensics Speech Filter
13. Steep Slope Forensics Speech Filter
14. DTMF Band-pass Filter
15. Sub-Sonic Filter (Steep)

Cell Phone Noise Filter

1. Default
2. GSM – Aggressive Setting
3. GSM – Light Setting
4. GSM – Nominal Setting
5. GSM – Typical Starting Point
6. GSM – Very Aggressive Setting
7. GSM – Very Light Setting

8., Nextel – Aggressive

9. Nextel – Least Aggressive
10. Nextel – Typical Starting Point
11. Nextel – Very Aggressive

Change Speed Filter

1. Fractional Speed Mastering: 33.3 to 45 RPM
2. Fractional Speed Mastering: 45 to 78.2 RPM
3. Fractional Speed Mastering: 45 to 78.8 RPM
4. Fractional Speed Mastering: 45 to 80 RPM
5. Fractional Speed Mastering: 78.2 to 80 RPM
6. 0.5% Speed Decrease
7. 0.5% Speed Increase
8. 1.00 % Speed Decrease
9. 1.00 % Speed Increase
10. 1.50 % Speed Decrease
11. 1.50 % Speed Increase
12. 3.00 % Speed Decrease
13. 3.00 % Speed Increase
14. 6.00 % Speed Decrease
15. 6.00 % Speed Increase
16. 1.00 % Pitch Shift Upwards
17. 1.00 % Pitch Shift Downwards
18. Rise and Fall
19. Sin Wave Function
20. Sin Wave Function – Small change
21. Tape Recorder VOX Start Compensation
22. Tape Recorder VOX Stop Compensation

Channel Blender

1. Dreamy Veil
2. Early Stereo Anti “Ping-Pong” Effect
3. FM Stereo Multi-path Distortion Filter
4. FM Stereo Noise Filter
5. Lead Vocal Attenuator 1
6. Lead Vocal Attenuator 2
7. Lead Vocal Attenuator 3
8. Lead Vocal Attenuator 4
9. Severe FM Stereo Noise Filter
10. Spatializer 1
11. Spatializer 2
12. Spatializer 3
13. Spatializer 4
14. Stereo Ambience Signal Only
15. Vinyl LP Bass Clarifier
16. Vinyl LP Rumble Cancellation Filter

Continuous Noise Filter

1. Flat-line Response
2. CrO2 (High Bias) Cassette Tape Fingerprint (no NR)
3. CRO2 (High Bias) Cassette Tape Fingerprint (with NR)
4. Fe (Normal Bias) Cassette Tape Fingerprint (no NR)
5. FE (Normal Bias) Cassette Tape Fingerprint (with NR)
6. FeCRO2 (High Bias) Cassette Tape Fingerprint (no NR)
7. FeCRO2 (High Bias) Cassette Tape Fingerprint (with NR)
8. Metal (Metal Bias) Cassette Tape Fingerprint (no NR)
9. Metal (Metal Bias) Cassette Tape Fingerprint (with NR)
10. Demo Audio Wave file De-Noise
11. Dynamic Rumble (Only) Filter
12. Edison Diamond Disc Fingerprint
13. Lazy Man's Noise Sample
14. Pathé 80 RPM Acoustic Fingerprint
15. Typical 1940's Acetate Noise Fingerprint
16. Typical 1940's Shellac 78 Fingerprint
17. Typical 33.3 RPM Vinyl Noise Fingerprint
18. Typical 45 RPM Vinyl Noise Fingerprint
19. 1 7/8 ips reel-to-reel tape fingerprint
20. 3 ¾ ips reel-to-reel tape fingerprint
21. 3 ¾ ips reel-to-reel tape fingerprint (w/ enhancement)
22. 7 ½ ips reel-to-reel tape fingerprint
23. 15 ips reel-to-reel tape fingerprint
24. 33 RPM LP Starting Point
25. 45 RPM Starting Point
26. 78 RPM, 8 inch Audiodisc Acetate Transcription
27. 78 RPM, 10 inch Audiodisc Acetate Transcription
28. 78 RPM, 10 inch Presto Acetate Transcription
29. 160 RPM Vertical Cut Cylinder Starting Point
30. 1890 Edison 2 Minute White Wax Cylinder
31. 1902 Columbia 78 RPM Fingerprint
32. 1903 Acoustical Lateral Fingerprint
33. 1904 Columbia Phonograph Record Fingerprint
34. 1904 Standard Disc Record 78 RPM Fingerprint
35. 1905 Victor 78 RPM Fingerprint
36. 1906 Edison 2 Minute Wax Cylinder (Standard)
37. 1912 Edison 2 Minute Celluloid Cylinder
38. 1919 Edison Blue Amberol 4 Minute Cylinder
39. 1928 Edison Lateral Cut 78 RPM Fingerprint
40. 1930’s (Early) Victor 33 RPM Program Transcription
41. 1932 Victor Pre-Grooved Home Recording
42. 1933 HMV (His Masters Voice E) 78 RPM Fingerprint
43. 1939 English Columbia 78 RPM Fingerprint
44. 1940 Bluebird 78 RPM Fingerprint
45. 1940’s Victor 78 RPM Fingerprint
46. 1948 Capitol 78 RPM Fingerprint
47. 1952 Vinyl 78 RPM Fingerprint
48. 1954 Monophonic 10 inch diameter LP Fingerprint
49. 1952 Mercury 78 RPM Fingerprint
50. 1973 Columbia Vinyl LP Fingerprint (RIAA Equalization)
51. Forensics AFDF – Fast & Heavy

52. Forensics AFDF – Fast & Light
53. Forensics AFDF – Fast & Very Light
54. Forensics AFDF – Slow & Heavy
55. Forensics AFDF – Slow & Light
56. Forensics AFDF – Slow & Very Heavy
57. Forensics AFDF – Slow and Very Light
58. Forensics AFDF – Very Slow & Very Heavy
59. Forensics AFDF – Very Fast & Very Heavy
60. Forensics AFDF – Very Slow & Heavy
61. Forensics AFDF – Very Slow & Very Heavy
62. Micro-cassette 1 7/8 ips Ferris Oxide Fingerprint
63. Micro-cassette 1 7/8 ips Metal Fingerprint
64. Micro-cassette 15/16 ips Ferris Oxide Fingerprint
65. Micro-cassette 15/16 ips Metal Fingerprint
66. Spectral Subtraction Sweet Spot
67. SW Radio Fingerprint – 16 Meter Band
68. Lazy Man's Noise Sample #2
69. 1938 Victor Scroll Label 78 Fingerprint
70. 1952 Vinyl 78 Fingerprint
71. 1954 Monophonic LP (10 Inch)
72. 1966 Roulette 45 RPM Fingerprint
73. 1971 RCA 45-RPM Fingerprint
74. 1973 Columbia LP (RIAA)
75. 1974 Columbia 45-RPM Fingerprint
76. Auto Spectrum CNF Aggressive Setting
77. Auto Spectrum CNF Maximum Setting
78. Auto Spectrum CNF Nominal Setting
79. Auto Spectrum CNF Very Aggressive Setting
80. Auto Spectrum CNF Very Light Setting
81. 78 RPM Acoustical Starting Point
82. 78 RPM Electrical Starting Point
83. 80 RPM Vertical Cut Disc Starting Point
84. General Purpose (flat-line response) Starting Point

De-Clipper
1. Analog Tape Recorder De-Clipper
2. Digital Wave file De-Clipper
3. Operational Amplifier De-Clipper
4. Rate Occurrence De-Clipper
5. Severe Overdrive De-Clipper
6. Transistor Amplifier De-Clipper
7. Vacuum Tube Amplifier De-Clipper
8. Very Strong Interpolation Curvature

Dynamic Bass Processor

1. C3 and lower Bass enhancement (only)
2. CD and lower bass Stout Enhancement (only)
3. Classical Chamber Music with Rumble Reduction Stout
4. Classical Chamber Music with Rumble Reduction
5. Deep Bass Enhancement (only) Stout
6. Deep Bass Enhancement (only)
7. Deep Bass heavy Enhancement (only)
8. Deep Bass light Enhancement (only)
9. Deep Bass light Enhancement (only) Stout
10. Deep Bass Punch – Stout
11. Deep Bass Punch
12. Default
13. Default Backup
14. De-Thumper (Strong Effect)
15. Dyna Bass Demo File – Stout
16. Dyna Bass Demo File
17. Dyna Bass Demo File (Bass enhancement only) stout
18. Dyna Bass Demo File (Bass enhancement only)
19. Dyna Bass Demo File (rumble reduction & bass boost)
20. Dynamic Bass Punch Mode 1 – Stout
21. Dynamic Bass Punch Mode 2
22. Dynamic Bass Punch Stout
23. Large Venue HVAC Noise Reduction (strong)
24. Large Venue HVAC Noise Reduction
25. Maximum Green Zone Settings (Compander Mode)
26. Minimum Green Zone Settings
27. Nominal Green Zone Settings

Dynamic Noise Filter
1. 3 ¾ ips Tape Hiss Attenuator
2. Brass Instrument Enhancer
3. Cassette Tape De-Hisser
4. Demo Audio Wave file De-Hiss
5. Forensics Wavefile Severe Noise Reduction Filter
6. Late 1930's 78-RPM Record Noise Reduction Filter
7. Spectral Exciter Enhancer 1
8. Spectral Exciter Enhancer 2
9. Spectral Exciter Enhancer 3
10. Spectral Exciter/Enhancer 1
11. Spectral Exciter/Enhancer 2

Dynamics Processor
1. 1 kHz De-Esser
2. 2 kHz De-Esser
3. 3 kHz De-Esser
4. 4 kHz De-Esser
5. 5 kHz De-Esser
6. 6 kHz De-Esser
7. 7 kHz De-Esser
8. 8 kHz De-Esser
9. 9 kHz De-Esser
10. 10 kHz De-Esser
11. ALC with Very Long Time Constants
12. Automatic Level Control
13. Automatic Level Control with Expander
14. Basic 2500 Hz De-Esser
15. ALC with Long Time Constants
16. ALC with Very Long Time Constants
17. Automatic Level Control
18. Automatic Level Control with Expander
19. Basic 2,500 Hz De-Esser

20. Echo Reduction 1 (Spoken Word)
21. Echo Reduction 2 (Spoken Word)
22. Echo Reduction 3 (Spoken Word)
23. Elevator Music Compressor
24. Heavy Compression
25. Heavy Downward Expansion
26. Light Compression
27. Light Noise Reduction
28. Limiter
29. Noise Gate – Fast
30. Noise Gate – Slow
31. The Politicians Friend
32. ALC with Short Time Constants
33. Background Sound Enhancement
34. Basic 2,500 Hz De-Esser
35. Basic 2,500 Hz De-Esser
36. De-Ess #1
37. De-Ess #2
38. De-Ess #3
39. Downward Expander
40. General Purpose Compressor
41. Moderate Speed Ballad Compressor
42. Noise Gate
43. Noise Gate with 100 mSec Time Constants
44. Reverb Reduction 1 (Spoken Word)
45. Reverb Reduction 2 (Spoken Word)
46. Rock Compressor
47. Slow Ballad Compressor
48. Telephone "near-far" party compensation
49. The Sportscaster
50. The Wedding DJ

Echo Effect Presets

1. Asbury Park
2. Basketball Bounce
3. Esser 1
4. Esser 2
5. Forensics Speech Articulator 1
6. Forensics Speech Articulator 2
7. Forensics Speech Articulator 2
8. Grand Canyon
9. Harmony in 2 Parts
10. Harmony in 3 Parts
11. Little Canyon
12. Long and Narrow Club
13. New Orleans
14. Boise Buildup Effect
15. Phaser
16. Pole-Zero Pair
17. Reverse Echo
18. San Francisco Dramatic
19. San Francisco Very Light
20. Slapback
21. Small and Intimate Club
22. Small Club
23. Space Wars
24. Spring Reverb Light
25. Spring Reverb Loosely Wound
26. Spring Reverb Tightly Wound
27. St. Patrick’s Cathedral
28. Stereo Heavy Plate Reverb
29. Stereo Light Plate Reverb
30. Stereo Ping-Pong Light
31. Stereo Ping-Pong Very Light
32. Stereo Reverse Light Plate Reverb
33. Stereo Reverse Ping Pong Light
34. Stereo Reverse Ping-Pong Very Light
35. Stereo Reverse Plate Reverb
36. Stereo Simulator 1
37. Stereo Simulator 2
38. Stereo Simulator 3
39. Stereo Simulator 4
40. Vocal Emphasize
41. Xenon

EZ-Clean Filter

1. EZ Clean (Aggressive)
2. EZ Clean (Average)
3. EZ Clean (Light Touch)
4. EZ Clean (Maximum)
5. EZ Clean (Minimum)
6. EZ Clean (Moderate – Light Hiss)
7. EZ Clean (Nominal Setting)
8. EZ Clean (Very Aggressive)
9. EZ Clean (Gentle Touch)
10. EZ Clean (Very Light Touch)
11. EZ Clean (Zero Effect)
12. 50 Hz Hum Filter Only
13. 60 Hz Hum Filter Only
14. Crackle Filter Only (Light)
15. Crackle Filter Only (Aggressive)
16. Crackle Filter Only (Nominal)
17. Crackle Filter Only (Very Aggressive)
18. Hiss Filter Only (Extremely Aggressive)
19. Hiss Filter Only (Gentle Touch)
20. Hiss Filter Only (Heavy Hand)
21. Hiss Filter Only (Light Touch)
22. Hiss Filter Only (Moderate Touch)
23. Hiss Filter Only (Very Heavy Hand)
24. Hiss Filter Only (Maximum)
25. Impulse Filter (Light)
26. Impulse Filter (Nominal)
27. Impulse Filter (Very Aggressive)
28. Impulse Filter (Aggressive)
29. Scratch Filter Only (Aggressive)
30. Scratch Filter Only (Light)
31. Scratch Filter Only (Nominal)
32. Scratch Filter Only (Very Aggressive)

EZ-Enhancer

1. Audiophile Quality
2. Automatic Level Control 1
3. Automatic Level Control 2
4. Automatic Level Control 3
5. Automatic Level Control 4
6. Default

7. Effects Bypass
8. Exciter – Evens Compressor
9. Exciter – Evens Expander
10. Exciter – Odds & Evens Compressor
11. Exciter – Odds & Evens Expander
12. Fat Bottom Evens, Balanced Compression
13. Fat Bottom Evens Balanced Expansion
14. Fat Bottom Evens, Compression
15. Fat Bottom Evens Expansion
16. Fat Bottom Evens Tubes
17. Fat Bottom Odds, Balanced Compression
18. Fat Bottom Odds Compression
19. Fat Bottom Odds Expansion
20. Fat Bottom Odds Tubes
21. Final Cleanup 33 RPM
22. Final Cleanup 45 RPM
23. Final Cleanup 78 RPM
24. Final Cleanup Tape
25. Full Spectrum Evens Tubes
26. Full Spectrum Odds Tubes
27. Multiband Limiter
28. Multiband Noise Gate
29. Slow Response Compressor
30. Slow Response Expander
31. Spoken Word Filter
32. Sweet Balanced Exciter
33. Sweet Evens Compression
34. Sweet Evens Exciter
35. Sweet Evens Expansion
36. Sweet Evens Tubes
37. Sweet Odds & Evens Exciter
38. Sweet Odds, Balanced Compression
39. Sweet Odds, Balanced Expansion
40. Sweet Odds Compression
41. Sweet Odds Expansion
42. Sweet Odds Tubes
43. Warm Balanced Exciter
44. Warm Evens, Balanced Compression
45. Warm Evens Balanced Expansion
46. Warm Evens Compression
47. Warm Evens Exciter
48. Warm Evens Expansion
49. Warm Evens Tubes
50. Warm Odds & Evens Exciter
51. Warm Odds, Balanced Compression
52. Warm Odds, Balanced Expansion
53. Warm Odds Compression
54. Warm Odds Expansion
55. Warm Odds Tubes

EZ-Impulse Noise Filter

1. 45-RPM Record Starting Point
2. 7 Inch, 33-RPM Starting Point
3. Aggressive Scratch & Crackle Remover
4. Aggressive Scratch and Crackle Remover
5. Crackle Only Eliminator
6. Gentle Scratch and Crackle Remover
7. Heavy Scratch, Light Crackle
8. Late 1940's 78
9. Light Scratch, Moderate Crackle Remover
10. Moderately Gentle Impulse Filter
11. Scratch and Crackle Eliminator
12. Scratch Only Eliminator
13. Very Aggressive Impulse Filter
14. Very Large 78 RPM Clicks
15. Very Large LP Clicks

EZ-Forensics Filter

1. (170 Filter Factory Presets which are available via a Wizard Driv

File Conversions

1. 1000 Hz, – 90 Degree Phase Shift Converter
2. 1000 Hz, + 90 Degree Phase Shift Converter
3. 400 Hz, +120 Degree Phase Shift Converter
4. 400 Hz, -120 Degree Phase Shift Converter
5. 50 Hz, + 90 Degree Phase Shift Converter
6. 50 Hz, +120 Degree Phase Shift Converter
7. 50 Hz, -120 Degree Phase Shift Converter
8. 50 Hz, -90 Degree Phase Shift Converter
9. 60 Hz, - 90 Degree Phase Shift Converter
10. 60 Hz, + 90 Degree Phase Shift Converter
11. 60 Hz, +120 Degree Phase Shift Converter
12. 60 Hz, -120 Degree Phase Shift Converter
13. Adaptive Demo
14. Mono to Stereo Simulator
15. Mono to Stereo Simulator 2
16. Mono to Stereo Simulator 3
17. Monophonic to Stereo
18. Monophonic Wave file Clone
19. Monophonic Wave file Clone w/ 3 dB Attenuation
20. Monophonic Wave file Clone with 3 dB Gain
21. Monophonic Wave file Clone with 6 dB Attenuation
22. Monophonic Wave file Clone with 6 dB Gain
23. Stereo Lateral Cut to Monophonic
24. Stereo Phase Inversion
25. Stereo Solo Vocal Reduction
26. Stereo to Monophonic
27. Stereo to Single Track Mono (left)
28. Stereo to Single Track Mono (right)
29. Stereo to Stereo Reverse
30. Stereo Vertical Cut to Monophonic
31. Stereo Wave file Clone
32. Stereo Wave file Clone w/ 3 dB Attenuation
33. Stereo Wave file Clone w/ 3 dB Gain
34. Stereo Wave file Clone w/ 3 dB Gain
35. Stereo Wave file Clone w/ 6 dB Attenuation
36. Stereo Wave file Clone with 6 dB Gain
37. Stereo Wave file Clone with 9.9 dB Gain
38. Strange Brew
39. Vertical Cut To Monophonic Converter

Filter Sweeper

1. 33.3 RPM Low Pass Masked Fade-In
2. 33.3 RPM Low Pass Masked Fade-Out
3. 78 RPM Low Pass Masked Fade-In
4. 78 RPM Low Pass Masked Fade-Out

5. Exponential High Pass Masked Fade-In
6. Exponential High Pass Masked Fade-Out
7. Exponential Low Pass Masked Fade-In
8. Exponential Low Pass Masked Fade-Out
9. Exponentially Swept Notch Filter
10. Linear High Pass Masked Fade-In
11. Linear High Pass Masked Fade-Out
12. Linear Low Pass Masked Fade-In
13. Linear Low Pass Masked Fade-Out
14. Linearly Swept Notch Filter
15. Log Low Pass Masked Fade-In
16. Log Low Pass Masked Fade –Out
17. Logarithmically Swept Notch Filter

Forensics AFDF

1. Fast & Heavy
2. Fast & Light
3. Fast & Medium
4. Fast & Very Light
5. Slow & Heavy
6. Slow & Light
7. Slow & Very Heavy
8. Slow & Very Light
9. Very Fast & Very Heavy
10. Very Slow & Heavy
11. Very Slow & Very Heavy

Gain Change

1. 0.5 dB Gain Decrease
2. 0.5 dB Gain Increase
3. 1.0 dB Gain Decrease
4. 1.0 dB Gain Increase
5. 1.5 dB Gain Decrease
6. 1.5 dB Gain Increase
7. 10 dB Gain Decrease
8. 10 dB Gain Increase
9. 11 dB Gain Decrease
10. 11 dB Gain Increase
11. 12 dB Gain Decrease
12. 12 dB Gain Increase
13. 13 dB Gain Decrease
14. 13 dB Gain Increase
15. 14 dB Gain Decrease
16. 14 dB Gain Increase
17. 16 dB Gain Decrease
18. 16 dB Gain Increase
19. 17 dB Gain Decrease
20. 17 dB Gain Increase
21. 18 dB Gain Decrease
22. 18 dB Gain Increase
23. 19 dB Gain Decrease
24. 19 dB Gain Increase
25. 2 dB Gain Decrease
26. 2 dB Gain Increase
27. 20 dB Gain Decrease
28. 20 dB Gain Increase
29. 25 dB Gain Decrease
30. 3 dB Gain Decrease
31. 3 dB Gain Increase
32. 30 dB Gain Decrease
33. 4 dB Gain Decrease
34. 4 dB Gain Increase
35. 40 dB Gain Decrease
36. 5 dB Gain Decrease
37. 5 dB Gain Increase
38. 6 dB Gain Decrease
39. 6 dB Gain Increase
40. 60 dB Gain Decrease
41. 7 dB Gain Decrease
42. 7 dB Gain Increase
43. 78 RPM Laterals Gain Correction
44. 8 dB Gain Decrease
45. 8 dB Gain Increase
46. 80 dB Gain Decrease
47. 9 dB Gain Decrease
48. 9 dB Gain Increase
49. Curvilinear Fade-In
50. Curvilinear Fade-Out
51. De-Clipper Pre-Processing Gain Change
52. Fade Down / 20 dB
53. Fade Up / 20 dB
54. Logarithmic Fade In
55. Logarithmic Fade Out
56. Slew Gain Down from 0 dB to -10 dB
57. Slew Gain Down from 0 dB to -15 dB
58. Slew Gain Down from 0 dB to -20 dB
59. Slew Gain Down from 0 dB to -3 dB
60. Slew Gain Down from 0 dB to -4 dB
61. Slew Gain Down from 0 dB to -5 dB
62. Slew Gain Down from 0 dB to -6 dB
63. Slew Gain Down from 0 dB to -7 dB
64. Slew Gain Down from 0 dB to -8 dB
65. Slew Gain Down from 0 dB to -9 dB
66. Slew Gain Down from +10 dB to 0 dB
67. Slew Gain Down from +15 dB to 0 dB
68. Slew Gain Down from +20 dB to 0 dB
69. Slew Gain Down from +3 dB to 0 dB
70. Slew Gain Down from +4 dB to 0 dB
71. Slew Gain Down from +5 dB to 0 dB
72. Slew Gain Down from +6 dB to 0 dB
73. Slew Gain Down from +7 dB to 0 dB
74. Slew Gain Down from +8 dB to 0 dB
75. Slew Gain Down from +9 dB to 0 dB
76. Slew Gain Up from – 10 dB to 0 dB
77. Slew Gain Up from – 15 dB to 0 dB
78. Slew Gain Up from – 20 dB to 0 dB
79. Slew Gain Up from – 3 dB to 0 dB
80. Slew Gain Up from – 4 dB to 0 dB
81. Slew Gain Up from – 5 dB to 0 dB
82. Slew Gain Up from – 6 dB to 0 dB
83. Slew Gain Up from – 7 dB to 0 dB
84. Slew Gain Up from – 8 dB to 0 dB
85. Slew Gain Up from – 9 dB to 0 dB
86. Slew Gain Up from 0 dB to + 10 dB
87. Slew Gain Up from 0 dB to + 15 dB

88. Slew Gain Up from 0 dB to + 20 dB
89. Slew Gain Up from 0 dB to + 3 dB
90. Slew Gain Up from 0 dB to + 4 dB
91. Slew Gain Up from 0 dB to + 5 dB
92. Slew Gain Up from 0 dB to + 6 dB
93. Slew Gain Up from 0 dB to + 7 dB
94. Slew Gain Up from 0 dB to + 8 dB
95. Slew Gain Up from 0 dB to + 9 dB

Graphic Equalizer (10 Band)

1. 1000 Hertz Bump
2. 1000 Hertz Dip
3. 1000 Hertz Small Bump
4. 1000 Hertz Small Dip
5. 1000 Hertz Very Small Bump
6. 1000 Hertz Very Small Dip
7. 125 Hertz Bump
8. 125 Hertz Dip
9. 125 Hertz Small Bump
10. 125 Hertz Small Dip
11. 125 Hertz Very Small Bump
12. 125 Hertz Very Small Dip
13. 16000 Hertz Bump
14. 16000 Hertz Dip
15. 16000 Hertz Small Bump
16. 16000 Hertz Small Dip
17. 16000 Hertz Very Small Bump
18. 16000 Hertz Very Small Dip
19. 2000 Hertz Bump
20. 2000 Hertz Dip
21. 2000 Hertz Small Bump
22. 2000 Hertz Small Dip
23. 2000 Hertz Very Small Bump
24. 2000 Hertz Very Small Dip
25. 250 Hertz Bump
26. 250 Hertz Dip
27. 250 Hertz Small Bump
28. 250 Hertz Small Dip
29. 250 Hertz Very Small Bump
30. 250 Hertz Very Small Dip
31. 31 Hertz Bump
32. 31 Hertz Dip
33. 31 Hertz Small Bump
34. 31 Hertz Small Dip
35. 31 Hertz Very Small Bump
36. 31 Hertz Very Small Dip
37. 4000 Hertz Bump
38. 4000 Hertz Dip
39. 4000 Hertz Small Bump
40. 4000 Hertz Small Dip
41. 4000 Hertz Very Small Bump
42. 4000 Hertz Very Small Dip
43. 500 Hertz Bump
44. 500 Hertz Dip
45. 500 Hertz Small Bump
46. 500 Hertz Small Dip
47. 500 Hertz Very Small Bump
48. 500 Hertz Very Small Dip
49. 62 Hertz Bump
50. 62 Hertz Dip
51. 62 Hertz Small Bump
52. 62 Hertz Small Dip
53. 62 Hertz Very Small Bump
54. 62 Hertz Very Small Dip
55. 8000 Hertz Bump
56. 8000 Hertz Dip
57. 8000 Hertz Small Bump
58. 8000 Hertz Small Dip
59. 8000 Hertz Very Small Bump
60. 8000 Hertz Very Small Dip
61. Acoustical 78 EQ
62. Acoustical 78 EQ
63. Alternating Ripple
64. Alternating Ripple Reverse
65. Bass Boost
66. Bass Boost with Slight High End Cut
67. Bass Cut
68. Brilliance
69. Deep Bass
70. Flat
71. Front Row Center
72. High Definition
73. Loudness Contour
74. Mids Boost
75. Mids Cut
76. Noble
77. Presence
78. Robust Male Speaking Voice
79. Sad Face
80. Sharp Edge Remover
81. Smiley Face
82. Sub-Woofer Accentuation
83. Treble Boost
84. Treble Cut
85. Ultra Bass
86. Set All Sliders to Maximum Position
87. Set All Sliders to Middle Position
88. Set All Sliders to Minimum Position

Harmonic Reject Filter

1. 35 mm Cine Frame Audio Flicker Filter
2. American 120 Hz Buzz Filter
3. American 120 Hz Buzz Filter - Gentle touch
4. American 60 Hz Buzz Filter
5. American 60 Hz Buzz Filter - Gentle touch
6. European 100 Hz Buzz Filter
7. European 100 Hz Buzz Filter - Gentle touch
8. European 50 Hz Buzz Filter
9. European 50 Hz Buzz Filter - Gentle touch
10. Flange-1
11. Flange-2
12. Flange-3
13. Flange-4
14. Flange-5
15. NTSC TV Vertical Bleed-through Filter
16. PAL TV Vertical Bleed-through Filter

17. SECAM TV Vertical Bleed-through Filter

High-pass Filter
1. Demo Audio Wavefile De-Rumble
2. DC Offset Remover
3. De-Thumper (for selective filtering)
4. De-Thumper 2 (for selective filtering)
5. Differentiator
6. Impulse Viewer
7. Rumble Filter
8. Steep Rumble Filter
9. Steeper Rumble Filter
10. Sub-Sonic Filter
11. Sub-Sonic Filter 2
12. Sub-Sub Sonic Filter
13. Sub-Sub Sonic Filter 2

Impulse Noise Filter
1. 1940's Acetate Starting Point
2. 1940's Shellac 78 Starting Point
3. 45 RPM Starting Point
4. AM Radio Static Filter
5. Badly Gouged 78 De-Scratcher
6. Cylinder (2 minute) Record Starting Point
7. Cylinder (4 minute) Record Starting Point
8. Demo Audio Wave file De-Click
9. Diamond Disc 80 RPM Starting Point
10. Early Shellac 78 RPM Lateral Starting Point
11. High Fidelity 78s using HQ Mode
12. LP De-Click Starting Point
13. LP Static Discharge Noise Suppressor
14. Pathé 80 RPM Starting Point
15. Sharp Rise-time AM Radio Static Suppressor
16. Universal Impulse Filter
17. Vertical Cut Starting Point
18. Very Large 78-RPM Clicks
19. Very Light Touch 78 RPM De-Clicker
20. Vinyl 78 RPM Lateral Starting Point
21. Vinyl Extremely Small Click Only filter
22. Vinyl first pass using HQ mode
23. Vinyl Large and Dense Click Attenuator
24. Vinyl Large Click Only Filter
25. Vinyl Mode with Threshold Limit
26. Vinyl second pass using HQ mode
27. Vinyl Small Click Only Filter
28. Vinyl third pass using HQ mode
29. Vinyl Very Large Click Only filter
30. Vinyl Very Small Click Only Filter

Low-pass Filter
1. FM Multiplex Stereo Multi-path Filter
2. Integrator
3. Light LP Surface Noise Filter
4. Passive Analog Scratch Filter
5. Steep 78 RPM Surface Noise Attenuator
6. Steep LP Surface Noise Attenuator
7. Steep LP Surface Noise Attenuator
8. Ultrasonic Filter

Make Waves Generator
1. All Musical Notes running from C0 to D9#
2. 15 General Purpose Test Signals

Median Filter
1. 78 RPM Record De-Crackler
2. Extremely Muffled Recording Enhancer
3. Muffled Communications Enhancer
4. Timbre
5. Weighted Median Filter

Multi-Filter
1. 35 mm Cine Odd & Even Audio Flicker Filter
2. 45-RPM Record Cleanup Filter
3. A Room within a Room within a Room
4. Adaptive Filter with Noise Gate
5. Audio Mixer
6. Brick Wall Speech Enhancer with Gain Leveler
7. Brick Wall Speech Filter with Gain Leveler
8. Bright Tubes and Round Bass
9. Bright Tubes and Round Bass with Ambience
10. Butterworth Speech Filter w/Noise Gate
11. Butterworth Speech Filter With Gain Leveler
12. Butterworth Speech Filter with Gain Leveler
13. Cell Phone Noise Interference Attenuator (1 – 6)
14. Classical Music Compressor
15. De-Clipper (Mono)
16. De-Clipper (Stereo)
17. Dual Impulse Filter
18. Dual Impulse Filter 2
19. Dual Impulse Filter 3
20. Edison Acoustical Diamond Disc De-Thumper
21. Extreme 50 Hz Harmonic Reject Filter
22. Extreme 60 Hz Harmonic Reject Filter
23. EZ Forensics_Proto1
24. EZ Forensics_Proto2
25. Forensics Demo Cleanup Filter
26. Forensics Filter Lineup-Nominal
27. General Purpose Record Cleaner
28. General Purpose Vinyl De-Clicker
29. Inter-modulation Distortion Test Filter, 262 Hz & 2 kHz
30. Inter-modulation Distortion Test Filter, 60 Hz & 7 kHz
31. Live 50 Hz Buzz Filter
32. Live 50 Hz Buzz Filter (Odd and Even)
33. Live 60 Hz Buzz Filter
34. Live 60 Hz Buzz Filter (Odds & Evens)
35. Live Adaptive Filter
36. Live Average Filter
37. Live Band-pass Filter
38. Live Band-pass Filter
39. Live Brick Wall Filter
40. Live Channel Blender
41. Live Channel Blender
42. Live Channel Delay Line
43. Live Continuous Noise Filter
44. Live De-Clipper

45. Live Dynamic Noise Filter
46. Live Dynamics Processor
47. Live Echo
48. Live Ez-Impulse Noise Filter
49. Live File Converter
50. Live Graphic Equalizer
51. Live Hi-Pass Filter
52. Live Impulse Noise Filter
53. Live Low-Pass Filter
54. Live Median Filter
55. Live Mode Demo Clean-up SW Radio
56. Live Notch Filter
57. Live Paragraphic Equalizer
58. Live Polynomial Filter
59. Live Reverb
60. Live Spectral Filter
61. Live Tube Amplifier
62. Live Tube Amplifier
63. Loudness Maximizer
64. Low Cost Mono Cassette Tape Enhancer
65. Low Cost Mono Cassette Tape Enhancer
66. Magnetic Tape Emulator
67. McGhee Filter
68. MP3 Enhancer / Full Tilt Boogie
69. MP3 Enhancer Four
70. MP3 Enhancer One
71. MP3 Enhancer Three
72. MP3 Enhancer Two
73. Multiple Notch Filter
74. Multiple Notch Filter
75. Olde Overloaded Radio Playing Far Away
76. Pink to White Noise Converter, 20 kHz (+/-1.5 dB)
77. Poor Reception FM Stereo Enhancer
78. Record Clean-up Filter
79. Record Clean-Up Filter
80. Shellac American 78 RPM Starting Point (Flat Preamp)
81. Shellac American 78 RPM Starting Point (RIAA Preamp)
82. Shellac American 78 RPM Starting Point (Flat Preamp)
83. Shellac American 78 RPM Starting Point (RIAA Preamp)
84. Short Wave Radio Anti-Fader & Cleanup Filter
85. Speech Clarifier
86. Speech Enhancer (Basic)
87. Speech Filter with Noise Gate
88. Speech Filter with Noise Gate and Enhancer
89. Speech Filter with Noise Gate and Ultra-Enhancer
90. Stanton 500 RIAA Compensation Curve
91. Three Stage 50 Hz Buzz Filter
92. Three Stage 60 Hz Buzz Filter
93. Triple Impulse Filter
94. Ultra-Flange
95. Universal Pop & Click Filter
96. Universal Pop and Click Filter
97. Vertical Cut Acoustical Record De-Clicker Starting Point (Flat F
98. Vertical Cut Acoustical Record De-Clicker Starting Point (Flat F
99. Vertical Cut Electrical Record De-Clicker Starting Point (Flat Pr
100. Vertical Cut Electrical Record De-Clicker Starting Point (RIAA
101. Very Bright Tubes
102. Very Bright Tubes
103. Very Scratch Vinyl De-Clicker
104. Victor Program Transcription Record (cleaner)
105. Vinyl LP (Early Columbia) De-Clicker Starting Point – (Flat Pre
106. Vinyl LP (Early Columbia) De-Clicker Starting Point – (RIAA F
107. Vinyl LP or 45 RPM De-Clicker Starting Point – (Flat Preamp)
108. Vinyl LP or 45 RPM De-Clicker Starting Point – (Flat Preamp)
109. Vinyl LP or 45 RPM De-Clicker Starting Point – (RIAA Preamp
110. Vinyl LP or 45 RPM De-Clicker Starting Point – (RIAA Preamp
111. Voice-Over Filter & Signal Conditioner1 (deep)
112. Voice-Over Filter & Signal Conditioner2 (male)
113. Voice-Over Filter & Signal Conditioner3 (female)
114. Voice-Over Filter & Signal Conditioner4 (child)
115. Voice-Over Filter & Signal Conditioner 5 (shallow)
116. White (Random) to Brown Noise Converter
117. White (Random) to Brown Noise Converter 2
118. White (Random) to Grey Noise Converter
119. White Random To Seismic Noise Converter
120. White to 1/3 Octave Bucket Converter
121. White to Pink Noise Converter, 20 kHz
122. White to Seismic Noise Converter
123. White to Seismic Noise Converter, 100 Hz
124. White To Seismic Noise Converter, 20 Hz
125. White To Seismic Noise Converter, 50 Hz

Narrow Crackle Filter

1. Default
2. Gentle Touch Narrow Crackle
3. Large Sized Narrow Crackle
4. Medium Sized Narrow Crackle
5. Nominal Setting
6. Small Sized but Tall Narrow Crackle
7. Small Sized Narrow Crackle
8. Very Aggressive Narrow Crackle
9. Very Large Sized Narrow Crackle
10. Very Small Sized and Tall Narrow Crackle
11. Very Small Sized Narrow Crackle

Notch Filter

1. 1/3 Octave Bandpass Filter
2. 100 Hertz Slot Filter
3. 1000 Hz Slot Filter - One Tenth Octave
4. 1000 Hz Slot Filter- Half Octave
5. 1000 Hz Slot Filter- One Octave
6. 1000 Hz Slot Filter- Quarter Octave
7. 120 Hertz Slot Filter
8. 400 Hertz Notch Filter (Aircraft Electrical Noise Filter)
9. 400 Hertz Slot Filter
10. 50 Hz Slot Filter
11. 60 Hertz Slot Filter
12. American 120 Hertz Hum Filter
13. American 120 Hertz Hum Filter (sharp)

14. American 60 Hertz Hum Filter
15. American 60 Hertz Hum Filter (sharp)
16. American AM Heterodyne Filter
17. American AM Heterodyne Filter (sharp)
18. Endocardiograph Response Simulator
19. European 100 Hertz Hum Filter
20. European 100 Hertz Hum Filter (sharp)
21. European 50 Hertz Hum Filter
22. European 50 Hertz Hum Filter (sharp)
23. European AM Heterodyne Filter
24. European AM Heterodyne Filter (sharp)
25. FM Stereo Pilot Frequency Attenuator
26. Middle C Finder (262 Hertz on the Musical Scale)
27. NTSC Horizontal Scan Signal Attenuator (15,750 Hz)
28. PAL Horizontal Scan Signal Attenuator
29. SECAM Horizontal Scan Signal Attenuator
30. Simple 78 RPM de-crackle filter
31. Simple 78 RPM Record Hiss Filter

Overtone Synthesizer

1. Amplified Orchestral Effect
2. Balanced Orchestral Effect
3. Brilliant
4. Brilliant (very)
5. Default
6. Exaggerated Second Harmonic Overtones
7. Forensics Pseudo Sibilant (Clarifier)
8. Gentle Tough of Overtones
9. Near – Ultrasonic
10. Sparkling
11. Strong Second Harmonic Overtones
12. Subtle Second Harmonic Overtones
13. Top Octave Overtones Only
14. Top Octave Overtones Only (Strong)
15. Touch of Brilliance
16. Very Strong Second Harmonic Overtones
17. Wide Spectrum Second Harmonic Overtones
18. Wide Spectrum Second Harmonic Overtones (Subtle)
19. Wide Spectrum Second Harmonic Overtones (Very subtle)

Paragraphic Equalizer

1. 100 Hz Boost with 12 dB Shelf
2. 100 Hz Boost with 16 dB Shelf
3. 100 Hz Cut with -12 dB Shelf
4. 100 Hz Cut with -16 dB Shelf
5. 2 kHz Boost with 12 dB Shelf
6. 2 kHz Boost with 16 dB Shelf
7. 2 kHz Cut with -12 dB Shelf
8. 2 kHz Cut with -16 dB Shelf
9. 25 Hz Brick Wall High-pass Filter
10. 25 mSec De-Emphasis
11. 25 mSec Pre-Emphasis
12. 3 kHz Boost with 12 dB Shelf
13. 3 kHz Cut with -12 dB Shelf
14. 6 kHz Brick Wall Low-pass Filter
15. 75 mSec De-Emphasis
16. 75 mSec Pre-Emphasis
17. 78 RPM, 125 Hz Turnover Curve
18. 78 RPM, 200 Hz Turnover Curve
19. 78 RPM, 250 Hz Turnover Curve
20. 78 RPM, 500 Hz Turnover Curve
21. 78 RPM, 800 Hz Turnover Curve
22. 78-RPM, 300 Hz Turnover Curve
23. 78-RPM, 350 Hz Turnover Curve
24. 78-RPM, 400 Hz Turnover Curve
25. 78-RPM, 629 Hz Turnover Curve
26. American 120 Hz Hum Filter
27. American 120 Hz Hum Filter (sharp)
28. American 60 Hz Hum Filter
29. American 60 Hz Hum Filter (sharp)
30. Ampex Master EQ (AME) Playback Curve
31. Bass Boost, 12 dB, 200 Hz Turnover
32. Bass Boost, 12 dB, 250 Hz Turnover
33. Bass Boost, 12 dB, 400 Hz Turnover
34. Bass Boost, 12 dB, 500 Hz Turnover
35. Bass Boost, 9 dB, 200 Hz Turnover
36. Bass Boost, 9 dB, 250 Hz Turnover
37. Bass Boost, 9 dB, 400 Hz Turnover
38. Bass Boost, 9 dB, 500 Hz Turnover
39. Bass Boost, 6 dB, 200 Hz Turnover
40. Bass Boost, 6 dB, 250 Hz Turnover
41. Bass Boost, 6 dB, 400 Hz Turnover
42. Bass Boost, 6 dB, 500 Hz Turnover
43. Bass Boost, 3 dB, 200 Hz Turnover
44. Bass Boost, 3 dB, 250 Hz Turnover
45. Bass Boost, 3 dB, 400 Hz Turnover
46. Bass Boost, 3 dB, 500 Hz Turnover
47. Bass Cut, 12 dB, 200 Hz Turnover
48. Bass Cut, 12 dB, 250 Hz Turnover
49. Bass Cut, 12 dB, 400 Hz Turnover
50. Bass Cut, 12 dB, 500 Hz Turnover
51. Bass Cut, 9 dB, 200 Hz Turnover
52. Bass Cut, 9 dB, 250 Hz Turnover
53. Bass Cut, 9 dB, 400 Hz Turnover
54. Bass Cut, 9 dB, 500 Hz Turnover
55. Bass Cut, 6 dB, 200 Hz Turnover
56. Bass Cut, 6 dB, 250 Hz Turnover
57. Bass Cut, 6 dB, 400 Hz Turnover
58. Bass Cut, 6 dB, 500 Hz Turnover
59. Bass Cut, 3 dB, 200 Hz Turnover
60. Bass Cut, 3 dB, 250 Hz Turnover
61. Bass Cut, 3 dB, 400 Hz Turnover
62. Bass Cut, 3 dB, 500 Hz Turnover
63. CCIR 19cm / sec Home Playback Curve
64. CCIR 19cm / sec Home Recording Curve
65. CCIR 19cm/ sec Studio Playback Curve
66. CCIR 19cm/ sec Studio Recording Curve
67. CCIR 38cm/ sec Studio Playback Curve
68. CCIR 38cm/ sec Studio Recording Curve
69. DTMF Comb Filter (Narrow)
70. DTMF Comb Filter (Normal)
71. DTMF Comb Filter (Ultra-Wide)

72. European 100 Hz Hum Filter
73. European 100 Hz Hum Filter (sharp)
74. European 50 Hz Hum Filter
75. European 50 Hz Hum Filter (sharp)
76. Flat Response
77. Fletcher-Munson Aural Hot Spot Accentuation
78. Fletcher-Munson Aural Hot Spot Attenuator
79. Fletcher-Munson Contour @ 100 dB SPL
80. Fletcher-Munson Contour @ 120 dB SPL
81. Fletcher-Munson Contour @ 20 dB SPL
82. Fletcher-Munson Contour @ 40 dB SPL
83. Fletcher-Munson Contour @ 60 dB SPL
84. Fletcher-Munson Contour @ 80 dB SPL
85. Forensics Bar Room Compensation Filter
86. NAB Tape Playback Curve
87. Ortophon3 (Grado Phono Cartridge Compensator)
88. Random IIR Based Phase Shifter 1
89. Random IIR Based Phase Shifter 2
90. Random IIR-Based Phase Shifter 1
91. Random IIR-Based Phase Shifter 2
92. Reverse NAB Tape Playback Curve
93. Reverse RIAA Phono Equalization Curve
94. Reverse RIAA w/ 125 Hz 78 Turnover
95. Reverse RIAA w/ 200 Hz 78 Turnover
96. Reverse RIAA w/ 250 Hz 78 Turnover
97. Reverse RIAA w/ 500 Hz 78 Turnover
98. Reverse RIAA w/ 800 Hz 78 Turnover
99. RIAA Phono Equalization Curve
100. Rolloff of 11 dB @ 10 kHz
101. Rolloff of 12 dB @ 10 kHz
102. Rolloff of 14 dB @ 10 kHz
103. Rolloff of 16 dB @ 10 kHz
104. Rolloff of 5 dB @ 10 kHz
105. Rolloff of 8.5 dB @ 10 kHz
106. Rumble Filter
107. Soft Slope (negative 1 dB / Octave)
108. Soft Slope (positive 1 dB / Octave)
109. Sound Level A-Weighting Curve
110. Sound Level C Weighting Curve
111. Stereo Simulator Left Channel Comb Filter
112. Stereo Simulator Left Channel Comb Filter (Wide)
113. Stereo Simulator Right Channel Comb Filter
114. Stereo Simulator Right Channel Comb Filter (Wide)

Reverb
1. Large Auditorium
2. Large Cathedral
3. Large Cavern
4. Large Submarine
5. Large Train Station
6. Light Reverb
7. Lighthouse
8. Lots of Reverb
9. Medium Size Auditorium
10. Medium Size Wood Room 1
11. Medium Size Wood Room 2
12. Phasing Effect 1
13. Phasing Effect 2
14. Slap-back 1
15. Slap-back 2
16. Slap-back 3
17. Slap-back 4
18. Small Intimate Night Club 1
19. Small Intimate Night Club 2
20. Small Intimate Night Club 3
21. Small Masonry Room 1
22. Small Masonry Room 2
23. Small Stone Church
24. Small Theater 1
25. Small Theater 2
26. Small Wood Church
27. Small Wood Room
28. Ultimate Reverb
29. Big Brass Boomy Bass
30. Bright Boomy Bass
31. Grand Canyon
32. Saint Peters Basilica
33. Boomy Old Wood Room
34. Delay Line, Very Short
35. Delay Line, Short
36. Delay Line, Medium
37. Delay Line, Long
38. Echo 1
39. Echo 2
40. Echo 3
41. Echo 4
42. Ambience 1
43. Ambience 2
44. Ambience 3
45. Ambience 4
46. Gentle Touch 1
47. Gentle Touch 2
48. Metal Plate Reverb
49. Classical Concert Hall
50. Smokey Club

Polynomial Filter
1. 5th Order Asymmetrical Compressing Non-Linearity
2. 5th Order Asymmetrical Expander Non-Linearity
3. Asymmetrical Expanding Non-Linearity with Phase Inv.
4. Basic Asymmetrical Non-Linearity
5. Class A Amplifier
6. Comb Generator
7. Compress +, Expand –
8. Double Inflection Point Non-Linearity
9. Double Inflection Point Non-Linearity #2
10. Even + Odd Order Harmonic Generator
11. Even Order Frequency Multiplier
12. Even Order Frequency Multiplier 2
13. Four Inflection Points #1
14. Four Inflection Points #2
15. Four Inflection Points #2

16. Full Wave Rectifier
17. Gross Non-Linearity
18. Instantaneous Compressor
19. Instantaneous Expander
20. Inverse Class A Amplifier
21. Inverse Push-Pull Power Amplifier
22. Inverse Push-Pull Power Amplifier with Phase Inv.
23. Inverting Real Time Expander
24. Level Sensitive Fuzz Box
25. Negative DC Offset
26. Phase Inverter
27. Positive DC Offset
28. Real Time Compressor 2
29. Real Time Expander 2
30. Saturated Amplifier with Local Feedback
31. Sensitive Full Wave Rectifier
32. Sensitive Half Wave Rectifier
33. Slightly Expansive Non-Linearity
34. Very Expansive Non-Linearity
35. Very Kinky
36. Weak Bulbs in the Olde Radio Set
37. X 0.25 Gain Increase
38. X 0.25 Gain Increase With Expansion
39. X 4 Gain Increase

Punch and Crunch

1. 1950's Jukebox
2. AM Radio De-Compressor
3. Background Music Compressor
4. Big Bad Bass
5. Big Brash Bass
6. Big Brass Bass
7. Cheap Cassette Tape Recorder Dynamic Expander
8. Classical Music Compressor
9. Exciter
10. Fat Round Bottom
11. Female Lead Vocal Enhancer
12. Fission Reaction
13. FM Radio Dial Presence
14. Forensics (Noise Reduction)
15. Front Row, Center
16. Fusion Reaction
17. Heavy Compression / Balanced Spectrum
18. Heavy Expansion / Balanced Spectrum
19. Heavy Expansion / Bright
20. Heavy Expansion / Brilliant
21. Heavy Expansion / Round Bottom
22. Heavy Expansion Big Bass
23. Hip Hop
24. Light Compression / Balanced Spectrum
25. Light Expansion / Balanced Spectrum
26. Light Expansion / Bright
27. Light Expansion / Brilliant
28. Light Expansion / Round Bottom
29. Male Lead Vocal Enhancer
30. Moderate Compression / Balanced Spectrum
31. Moderate Compression / Muted
32. Moderate Expansion / Balanced Spectrum
33. Moderate Expansion / Bright
34. Moderate Expansion / Brilliant
35. Moderate Expansion / Round Bottom
36. Multi-band ALC
37. Very Heavy Compression / Balanced Spectrum
38. Very Heavy Compression / Muted
39. Very Heavy Expansion / Bright
40. Very Heavy Expansion / Brilliant
41. Very Light Compression / Balanced Spectrum
42. Very Light Compression / Muted
43. Very Light Expansion / Balanced Spectrum
44. Very Light Expansion / Brilliant

Remove Silence Tool

1. Type 20-10-5
2. Type 20-25-5
3. Type 20-5-5
4. Type 25-25-5
5. Type 25-5-5
6. Type 25-10-5
7. Type 30-10-5
8. Type 30-25-5
9. Type 30-5-5
10. Type 35-10-5
11. Type 35-25-5
12. Type 35-5-5
13. Type 40-10-5
13. Type 40-25-5
15. Type 40-5-5
16. Type 45-10-5
17. Type 45-25-5
19. Type 45-5-5

Spectral Filter

1. 100 Hz Hum Filter (for Defective European Filter Capacitors)
2. 120 Hz Hum Filter (for Defective US Filter Capacitors)
3. 400 Hz Aircraft Electrical Noise Filter
4. 50 Hz Hum Filter (European)
5. 60 Hz Hum Filter (US)
6. 60 Hz Buzz Filter
7. 5 kHz Brick Wall Low Pass Filter
8. Speech Filter
9. Speech and 400 Hz Aircraft Electrical Buzz Filter
10. Speech and 50 Hz Buzz Filter
11. Speech and 60 Hz Buzz Filter
12. Phase Shifter- Randomized
13. Phase Shifter
14. 400 Hz Aircraft Electrical Buzz Filter
15. Human Voice Filter
16. Male Voice Contour
17. Female Voice Contour
18. Auto EQ Forensic Normalize to Brown Noise
19. Auto EQ Forensic Normalize to Inverse Brown Noise
20. Auto EQ Forensic Normalize to Human Voice
21. Auto EQ Forensic Normalize to Inverse Human Voice
22. Auto EQ Forensic Normalize to Pink Noise

23. Auto EQ Forensic Normalize to Inverse Pink Noise
24. Auto EQ Forensic Normalize to Pink Noise, Full
25. Auto EQ Forensic Normalize to Inverse Pink Noise
26. Auto EQ Forensic Normalize to White Noise
27. Auto EQ Forensic Normalize to White Noise, Brick Wall
28. Auto EQ Forensic Normalize to Brown Noise, Full
29. Auto EQ Forensic Normalize to White Noise, Narrow

Stretch & Squish Filter

1. 10 % Faster
2. 10 % Slower
3. 15% Faster
4. 15% Slower
5. 2% Faster
6. 2% Slower
7. 20 % Faster
8. 20 % Slower
9. 30 % Faster
10. 30 % Slower
11. 5 % Faster
12. 5 % Slower
13. Constant Pitch
14. Disguised Voice Effect 1
15. Disguised Voice Effect 2
16. Disguised Voice Effect 3
17. Disguised Voice Effect 4
18. Disguised Voice Effect 5
19. Disguised Voice Effect 6
20. Higher Pitch and Speed
21. Lower Pitch and Speed
22. Much Higher Pitch and Speed
23. Much Lower Pitch and Speed
24. Spoken Word Transcription Rate
25. Spoken Word Transcription Rate #3
26. Spoken Word Transcription Rate 1
27. Spoken Word Transcription Rate 2

Sub-harmonic Synthesizer

1. Balanced Spectrum, Bump and Thump
2. Balanced Spectrum, Maximus Bump and Thump
3. Balanced Spectrum, Minimus
4. Balanced Spectrum, Nominal Level
5. Balanced Spectrum, Slight Bump and Thump
6. Deep Bass Balanced
7. Deep Bass Light
8. Deep Bass Maximus Thumpus
9. Deep Bass Minimus
10. Deep Bass Moderato
11. Default
12. Disco Bass
13. Terra Bass Balanced Level
14. Terra Bass Dance Club Atmosphere
15. Terra Bass Maximus Thumpus
16. Terra Bass Minimus
17. Sharp Cutoff – 70 Hz
18. Sharp Cutoff – 80 Hz
19. Sharp Cutoff – 100 Hz
20. Sharp Cutoff – 120 Hz
21. Sharp Cutoff – 140 Hz
22. Sharp Cutoff – 140 Hz (Strong Effect)

Virtual Phono Preamp (VPP)

1. Approximate AES LP EQ Curve
2. Approximate FFRR LP EQ Curve
3. Approximate NAB LP EQ Curve
4. Approximate Atlantic LP Label
5. Approximate Bartok LP Label
6. Approximate Blue Note Jazz LP Label
7. Approximate Canyon LP Label
8. Approximate Capitol – Cetra LP Label
9. Approximate Capitol LP Label
10. Approximate Concert Hall LP Label
11. Approximate Contemporary LP Label
12. Approximate Cook LP Label
13. Approximate Elektra LP Label
14. Approximate EMS LP Label
15. Approximate Folkways LP Label
16. Approximate Good – Time Jazz LP Label
17. Approximate L'Olseau-Lyre LP Label
18. Approximate London LP Label
19. Approximate Lyricord LP Label
20. Approximate Mercury LP Label
21. Approximate Oceanic LP Label
22. Approximate Philharmonic LP Label
23. Approximate Polymusic LP Label
24. Approximate RCA Victor LP Label (Early)
25. Approximate Remington LP Label
26. Approximate Urania LP Label
27. Approximate Westminster LP Label
28. Exact Allied LP Label
29. Exact American Record Society LP Label
30. Exact Angel LP Label
31. Exact Bach Guild LP Label
32. Exact Bethlehem Label
33. Exact Boston LP Label
34. Exact Caedmon LP Label
35. Exact Camden LP Label
36. Exact Cetra – Soria LP Label
37. Exact Classic Editions Label
38. Exact Clef Label
39. Exact Collosseum LP Label
40. Exact Columbia LP Label
41. Exact Decca LP Label
42. Exact Epic LP Label
43. Exact Esoteric LP Label
44. Exact Haydn Society LP Label
45. Exact Kapp Label
46. Exact McIntosh Label

47. Exact MGM Label
48. Exact Montilla Label
49. Exact New Jazz Label
50. Exact Pacific Jazz LP Label
51. Exact Prestige Label
52. Exact RCA Victor Label
53. Exact Riverside LP Label
54. Exact Romany LP Label
55. Exact Savoy LP Label
56. Exact Tempo LP Label
57. Exact Vanguard LP Label
58. Exact Vox LP Label
59. Exact Walden LP Label
60. Default
61. Flat Preamp Hardware playing Acoustical Records
62. Flat Preamp Hardware playing American 78s
63. Flat Preamp Hardware playing Columbia Vinyl LPs
64. Flat Preamp Hardware playing European 78s
65. Flat Preamp Hardware playing RIAA Vinyl LPs
66. Line Input – Phono Preamp Bypass
67. RIAA Preamp Hardware Playing Acoustical Records
85. RIAA Preamp Hardware Playing American 78s
86. RIAA Preamp Hardware Playing Columbia Vinyl LPs
70. RIAA Preamp Hardware Playing European 78s
87. RIAA Preamp Hardware Playing RIAA Vinyl LPs
72. Approximate RIAA EQ with DIN Comp (Flat Preamp)
73. Approximate RIAA EQ with DIN Comp (RIAA Preamp)

Virtual Valve Amplifier (VVA)

1. 12AT7 based Class A Amplifier, Full Range
2. 12AU7 Based Class A Amplifier (Full Range)
3. 12AU7 Based Class A Amplifier (Sweet)
4. 12AU7 Based Class A Amplifier (Warm)
5. 12AU7 based Class A Amplifier, Full Range
6. 12AX7 Based Class A Amplifier (Full Range)
7. 12AX7 Based Class A Amplifier (Sweet)
8. 12AX7 Based Class A Amplifier (Warm)
9. 1935 Retro 15 Watt Push-Pull Amp
10. 1938 Retro 4 Watt Class A Power Amp
11. 25 Watt / Channel Power-Amp 1 (6L6-GC)
12. 25 Watt / Channel Power-Amp 2 (6L6-GC)
13. 25 Watt / Channel Power-Amp 3 (6L6-GC)
14. 25 Watt / Channel Power-Amp 4 (6L6-GC)
15. 2A3 Based Class A Amplifier (Full Range)
16. 2A3 Based Class A Amplifier (Sweet)
17. 2A3 Based Class A Amplifier (Warm)
18. 2A3 based Class A Amplifier Full Range
19. 6 EJ7B based Class A Amplifier, Full range
20. 6EJ7 Based Class A Amplifier (Full Range)
21. 6EJ7 Based Class A Amplifier (Sweet)
22. 6EJ7 Based Class A Amplifier (Warm)
23. 8 Watt, 6L6 Single-Ended Amplifier 1
24. 8 Watt, 6L6 Single-Ended Amplifier 2
25. 8 Watt, 6L6 Single-Ended Amplifier 3
26. Bright & Brassy Brass
27. Grunge (Odds Only)
28. Harmonic Enhancer 1 (Evens + Odds)
29. Harmonic Enhancer 2 (Evens + Odds)
30. Harmonic Enhancer 3 (Evens Only)
31. Harmonic Enhancer 4 (Evens Only)
32. Harmonic Enhancer 5 (Evens Only)
33. Harmonic Enhancer 6 (Odds Only)
34. Harmonic Enhancer Sweet Spot
35. High-End (Triode) Audio Pre-Amplifier
36. Overloaded Guitar Power Amplifier
37. Overloaded Guitar Pre-Amplifier
38. Pentode Microphone Pre-Amplifier 1
39. Pentode Microphone Pre-Amplifier 2
40. Purist 1
41. Purist 2
42. Purist 3
43. Red Hot Jazz
44. Silk – Medium Sibilance
45. Silk – Minimal Effect
46. Silk – Silky and Brash
47. Silk – Silky Brilliance
48. Silk – Silky Exciter
49. Silk – Silky Odd Harmonics
50. Silk – Silky Stridence
51. Silk – Silky Strings, Medium Effect
52. Silk – Silky Strings, Strong Effect
53. Silk – Slight Effect
54. Silk – Soft Sibilance
55. Silk – Subtle Effect
56. Silk – Very Subtle Effect
57. Triode Tube Pre-Amplifier 1
58. Triode Tube Pre-Amplifier 2
59. Triode Tube Pre-Amplifier 3
60. Triode Tube Pre-Amplifier 4
61. Triode Tube Pre-Amplifier 5
62. Triode Tube Pre-Amplifier 6
63. Vacuum Tube Fuzz Box 1
64. Vacuum Tube Fuzz Box 2
65. Vacuum Tube Fuzz Box 3
66. Vacuum Tube Fuzz Box 4

Voice Garbler

1. Default
2. Female Child Downshift 1
3. Female Child Downshift 2
4. Female Voice Downshift 1
5. Female Voice Downshift 2

6. Female Voice Downshift 3
7. Female Voice Upshift 1
8. Female Voice Upshift 2
9. Male Child Downshift 1
10 Male Child Downshift 2
11. Male Child Upshift 1
12. Male Child Upshift 2
13. Male Voice Downshift 1
14. Male Voice Downshift 2
15. Male Voice Upshift 1
16. Male Voice Upshift 2
17. Silly Girl

VVA Presets (Relating to the Fat Bass Feature)
1. Fat Bass – Audiophile Quality
2. Fat Bass – B15 (Ampeg)
3. Fat Bass – Balanced Harmonics
4. Fat Bass – Boomy
5. Fat Bass – Boomy 2
6. Fat Bass – Compressed & Grungy
7. Fat Bass – Even & Odd Harmonics
8. Fat Bass – Grimey
9. Fat Bass – Grungy
10. Fat Bass – Odd Harmonics
11. Fat Bass – Round Bottom End
12. Fat Bass – Subtle Effect

Whisper Enhancer (Forensics)
1. ~Starting Point
2. Default
3. Low Sensitivity – Moderate Response 1
4. Low Sensitivity – Moderate Response 2
5. Max Sensitivity – Fast Response
6. Max Sensitivity – Moderate Response
7. Max Sensitivity – Slowest Response
8. Max Sensitivity – Very Fast Response 1
9. Max Sensitivity – Very Fast Response 2
10. Moderate Sensitivity – Moderate Response 1
11. Moderate Sensitivity – Moderate Response 2
12. Moderate Sensitivity – Very Slow Response
13. Very Low Sensitivity – Moderate Response 1
14. Very Low Sensitivity – Moderate Response 2
15. Xtreme Settings

Wind Noise Filter
1. Category 1 Hurricane
2. Category 3 Hurricane
3. Category 4 Hurricane
4. Default – DCArt10
5. Default
6. Light Breeze
7. Strong Breeze
8. Tropical Storm
9. Very Light Breeze

A Brief History of Diamond Cut Productions

In the spring of 1986, an R&D engineer/scientist by the name of Craig Maier read an article in The Star Ledger, a local newspaper, entitled "Budget Cuts Cast Shadow on Edison National Historic Site." The article, written by science editor Kitta McPherson, described the deteriorating condition of the Edison National Historic Site and its archives located in West Orange, New Jersey. Among the many artifacts which were not receiving the proper curatorial attention due to poor funding was a collection of test-press recordings which were made by the Edison Company between the years of 1927 through 1929, which was their last few years in the record business. Craig told a friend and fellow engineer named Rick Carlson about the article in hopes that it might stir up in him some interest in the Edison site as well. Craig and Rick, after some considerable discussion, decided to offer to volunteer some of their spare time and technical expertise in the area of audio hardware and software engineering in order that the Edison Lateral collection of test pressing recordings could be transferred to digital tape so that the "sound artifacts" would be eternally preserved and archived in the digital domain at the site.

Contact was made with then Supervisor Museum Curator, Dr. Edward Pershey, Ph.D. During their first meeting at the site, Dr. Pershey showed the two engineers thousands of one-of-a-kind test pressing recordings that were piled in stacks on a long row of tables on the second floor of the Edison main laboratory building. This initial introduction to the collection was an earnest attempt to sober up these two individuals as to the magnitude of the undertaking for which they were volunteering. The total number of songs which were recorded numbered over 1200 in anywhere from two to five takes each. This only further increased their interest in the project since the possibility of finding some truly important music that had previously been unheard since the late 1920's would be quite high in such a large collection of test pressings. After several additional meetings with Dr. Pershey, an informal agreement was made such that the two engineers could proceed to seek out funding from private sources to set up an audio restoration laboratory in one of their own homes for the project. They contacted around 30 companies in the New Jersey area seeking funds to help build their laboratories. After about seven months of effort, they succeeded in raising enough money to fund their project. In addition to fund raising, they also designed and constructed several pieces of custom equipment that was needed for the project (equipment which was not readily available on the market at the time).

The next step was to become educated in the proper technique of archival audio transferring. To that end, they hired Mr. Tom Owens of the Rogers and Hammerstein musical library in New York City as an engineering consultant. Tom spent time with the two engineers at the New York City Public Library sound lab (Rogers and Hammerstein) teaching them some of the "tricks of the trade." Tom also visited the first sound lab which the two engineers set up for the restoration project located at Craig's home in Verona, NJ. He provided constructive criticism regarding the sound lab which the two engineers had set up, allowing them to improve upon their initial system. One significant problem which Tom highlighted for the two engineers was that of establishing the correct turnover frequency for the transfer of these lateral test pressings. Documentation could not be found at the Edison site regarding the specifics of this important parameter. So Rick and Craig devised some experiments which were conducted on a "high-end" vacuum tube based Edison phonograph designed around the same time period as the test pressings in order to deduce the correct turnover frequency. After their experiments, modifications were made to their magnetic phonograph pre-amplifier to provide the most likely proper turnover frequency for the transfers.

A seven year pro-bono contract was drawn up between the Edison National Historic Site / U.S. Department of the Interior, and Rick Carlson and Craig Maier for the purposes of executing the project outlined above. Finally, the two engineers were ready to begin the project. Nearly one full year had lapsed before the first record was transferred to digital tape at Craig's home in Verona, N.J. Shortly thereafter, the sound lab was rebuilt in the Maier's new home in Rockaway Township, NJ. That is the location in which the lion's share of the transfer project took place over the next seven years.

After transferring around 900 of the songs (times 2 - 5 takes per song, about 2,200 transfers in total) Craig and Rick decided that the music was not doing much good sitting in the underground vault of a museum. Since they were the only two people alive who had heard almost the entire collection, they decided that it would be a good idea to try to release some of this previously unreleased material (only around 200 of the songs had ever been released in the Edison lateral format). So they approached the Edison site in order to try to accomplish this. After about one year of frustration in dealing with the bureaucracy, they decided it would be a lot easier to form their own company and release these songs under their own record label. Thus was formed Diamond Cut Productions in 1992 with Craig and Rick providing their own seed capital for the venture. Their first release entitled "Unreleased Edison Laterals 1 - - - an anthology of Edison Needle type records" was such a success in the market that they were able to start another project in 1994 entitled "The California Ramblers, Edison Laterals 2." For this project, they decided to improve on the audio restoration process, which they had used on their previous release. Instead of analog signal processing, they migrated to digital signal processing utilizing their own algorithms to remove crackle, ticks, pops and hiss from the original material. They named their process (which ran on an inexpensive pc) "Diamond Cut Audio restoration tools" or DC-Art for short. Their technique proved successful to the extent that the Smithsonian Institution Press employed Diamond Cut Productions to perform audio restoration for some of their American Songwriter Series of CD releases using this process. Diamond Cut's third CD release entitled "Hot Dance of the Roaring 20's, Edison Laterals 3" was processed utilizing exclusively their own audio restoration program; all analog processing equipment had been abandoned by this point in time. In the meantime and in parallel with the efforts to bring "Hot Dance . . . " to the market, Craig worked with County records to produce and release an Edison olde time group on CD called "Ernest Stoneman and his Dixie Mountaineers" using their audio restoration process. In the spring of 1996, their program was first formally introduced into the commercial marketplace at a meeting of "Record Research" which was held at the Maier residence in Rockaway Township, NJ. Since then it has been sold throughout the world for not only musical audio restoration applications, but for others such as 911 call restoration, clarification of police surveillance recordings, cleanup of radio broadcasts for release on CD, restoration of historic spoken word recordings, cockpit voice recording restoration, plus many others.

Diamond Cut Productions, Inc. has now become one of the predominant international players in the audio restoration and enhancement software market. Their Forensics Audio Software product line has gained wide acceptance on an international scale. New features and improved performance will be added into their legacy audio restoration software products on a continuous process basis.

In the future, Diamond Cut Productions expects to continue releasing more CDs and digital downloads in their Edison Lateral Cut series. However, they have also branched out into other musical venues from the 1920s and 1930s time period.

Diamond Cut Audio Restoration Tools Development Timeline

(Product Development Timeline)

Mid 1993 -
While frustratingly removing clicks from an Edison record using just a computer and a mouse whilst drinking too much wine and beer, Rick Carlson and Craig Maier wrote the first few lines of code in an attempt to perform this process automatically. Although very crude at the beginning, this particular algorithm (the Impulse Filter) eventually developed into its present level of sophistication occupying about 500 K of compiled code. Several of those original lines and thinking are still in the present day code. Rick had stored those original lines of code under the filename of dcart.exe ultimately giving rise to the name "Diamond Cut Audio restoration tools" (DCArt).

Late 1993 -
Having heard about Diamond Cut through the grapevine, Bruce Talbot, executive producer for the Smithsonian Collection of Recordings contacted us and inquired if they would be willing to apply their "de-clicking" process to a particularly bad recording of "Darn That Dream." This restoration was released on the American Songbook Series on a CD entitled "James Van Heusen." which was copyrighted in 1994 as release number RD 048-18 / A 23955.

Early 1994 -
Having been satisfied with their work on the "James Van Heusen" CD, Bruce Talbot again contacted Diamond Cut to do some work on a release in the American Songbook Series, which was to be called "Richard Whiting." On it, they restored a "basket case" copy of a Rudy Vallée rendition of "Honey." In the liner notes for the CD it says "The only source available for "Honey" was a 78-rpm disc in very poor condition. The sound quality has been greatly improved by Craig Maier and Rick Carlson of Diamond Cut Productions, using their DC-Art system of sound restoration." This CD is also copyrighted in 1994 but under the release number of RD048-22 / A 24571.

Mid 1994 -
While recuperating from surgery and being extremely bored, Craig decided to restore an entire artist's output of Edison Lateral Cut records which Rick and Craig had earlier transferred to digital audiotape as part of the Edison Lateral Cut Disc archival project. The artist was Ernest V. Stoneman (and his Dixie Mountaineers). He used their DC-Art program to remove all extraneous clicks, pops, and hiss from the originals. Ultimately, Diamond Cut Productions sold the Master digital tape to County Records which released and copyrighted the CD in 1996 on County release number CD-3510.

Late 1994 -
Diamond Cut Productions decided that it was time to release another in its series of mostly Unreleased Edison Lateral Cut recordings. They choose "The California Ramblers" as the subject of the release. They also decided to use no additional analog processing equipment other than their DC-Art program to restore the recording. De-clicking, de-popping, de-hissing, de-rumbling and minor equalization were all performed via their computer algorithms. Thus was their first release which included only two analog steps, that of the mastering back in the late 1920's and that of the transfer to digital tape in the late 1980's and early 1990's. This CD is entitled "The California Ramblers - - - Edison Laterals 2" and was released under number DCP-301D and copyrighted in late 1994. This CD is still available from Diamond Cut Productions, Inc. and other venues throughout the world.

Early 1995 -
Diamond Cut Productions decided to expand their business from the production of CD's to the manufacture of audio restoration software products. They decided that since the program had been quite useful to in their CD business, it might also be useful to many other people with audio collections in need of restoration. The first public release took about 7 months to smooth out the bumps in the program that we had learned to work around. Making a commercial software product is much different than writing one for yourself as they were soon to find out!

April 21st 1995 -
Diamond Cut Productions sent out the initial Beta version of DC-Art to some potential customers identified as QA 1.1. They worked on de-bugging this software for the next few months. If you have an original copy of QA 1.1, it may be an antique! (Antique software - - - what's this world coming to??)

July 1996 -
Version 1.0 was officially released at a meeting of "Record Research" held at the Maier Rockaway Township residence. The first copies were sold at the end of the meeting during which the program had been demonstrated to a small group of about 15 people.

September 4th, 1997 -
A distributorship agreement is signed between Diamond Cut Productions, Inc. and Tracer Technologies, Inc. in order to help facilitate the marketing and distribution of the DC-Art product line.

December 1997 -
Version 2.0 of DC-Art featuring, among other things, real time Preview was introduced into the market.

September 1998 -
Version 3.0 of DC-Art, otherwise known as Diamond Cut 32 was introduced into the market with novel features such as the Virtual Valve Amplifier.

August 1999 -
Version 4.0 otherwise known as Live and Millennium was introduced. This brought with it a new level of performance and features in the audio restoration and enhancement software market. Features like Live feed through mode, and 24-bit/96 kHz support have changed the landscape in this area of endeavor.

December 1999 -
Established German distributorship for the Live and Millennium programs through Digital Broadcast Systems GmbH (dBS). The products can be seen and purchased in the German language at www.diamondcut.de

August 2001 -
Updated Live and Millennium to version 4.8 with bug fixes and the addition of a high resolution VU Meter. Also, the frequency resolution of the spectrum analyzer was greatly enhanced.

February 2001 -
Released the code for Enhance/MP3, a low cost product used for improving the sound quality of MP3 audio files. Diamond Cut Productions Inc. and Tracer Technologies form a new Partnership called Enhanced Audio Inc., whose sole purpose is to market, and distribute the world's best audio enhancement, restoration, and analysis tools.

June 2002 -
Released Live/Five to Beta

September 2002 -
Released Live/Five to Production

May 2004 -
Released Live/Six to Beta

August 2004 -
Released Live/Six to Production

July 2007 -
Released Version 7 to Beta

October 2007 -
Released Version 7 to Production

November 2008 -
Released Live/Forensics Audio Laboratory, Version 7.5 to Beta

January 2009 -
Released Live/Forensics Version Audio Laboratory, Version 7.5 to Production

November 2009 -
Released DC8, Version 8.0 to Beta

March 2010 -
Released DC8, Version 8.0 to Production

November 2010 –
Released DC Forensics8 Audio Laboratory, Version 8.0 to Beta Test

April 2011 –
Released DC Forensics8 Audio Laboratory, Version 8.0 to Production

December 2011 –
Released DC Forensics8 Audio Laboratory, Version 8.1 to Production

April 2012 –
Released DC8, Version 8.1 to Production

October 2015 –
Released DCForensics10 to Beta Test

March 2016 -
Released DCForensics10.0 to Production

March 2017 –
Released DCForensics10.03 to Production

March 2017 –
Released DCArt10 to Beta Test

June 2017 –
Released DCArt10 to Production –
July 2018
Released DCForensics10.5 to Production

November 2018 –
Released DCArt10.5 to Production

December 2019 –
Released DCArt10.6x with Dyna-Bass to Production

April 2020 –
Released DCForensics10.6x to Production

November 2020 –
Released DCArt10.70 to Production (with the Record Restoration Wizard added)

December 2020 –
Released DCForensics10.70 to Production (improved Voice ID, etc.)

June 2021 –
Released DC Forensics 10.75.5 to Production (improved sub-harmonic synthesizer with sharp-cutoff feature checkbox; added various colors of noise generation to Make Waves (7 added) and made them real-time Preview-able). Also, made some improvements to the registration process.

July 2021 –
Released DCFArt10.75 to production (various improvements in the algorithms - added make 7 noise generators to the Make Waves function having real time preview - improved the registration process.

October 2021 –
Released DCForensics10.8 and DCArt10.8 to beta testing.

November 2021 –
Released DCForensics10.8 and DCArt10.8 to production.

Diamond Cut Productions Edison Lateral Series CD and Cassette Music Releases

Diamond Cut Productions is proud to offer releases of Edison Lateral Cut Test Pressing recordings available in the CD format, with one available in both CD and Cassette. Also, we have an assortment of non-Edison recordings available. All of these recordings were restored using Diamond Cut Audio Restoration Software. These albums are also available in the .mp3 downloadable format located at the following link:

http://www.diamondcut.com/st3/product-category/music/

The following is our current product offering:

1. Unreleased Edison Laterals 1 CD Version (DCP-201D) - - - $17.95
   Unreleased Edison Laterals 1 Cassette Version (DCP-201S) - - - $9.95
2. The California Ramblers, Edison Laterals 2 (DCP-301D) - - - $17.95
3. Hot Dance of the Roaring 20's, Edison Laterals 3 (DCP-202D) - - - $17.95
4. Ernest V. Stoneman and his Dixie Mountaineers (Edison) - - - $13.95
5. Eva Taylor with Clarence Williams, Edison Laterals 4 (DCP-303D) - - - $17.95
6. Vaughn De Leath - The Original Radio Girl, Edison Laterals 5 (DCP-304D) - - - $17.95
7. Hot and Rare - Hot Dance tunes from Rare Jazz Recordings (DCP-203D) - - - $17.95
8. B.A. Rolfe and his Lucky Strike Orchestra (DCP-305D) - - - $17.95
9. The Marvelous Melodies of Peter Mendoza (DCP-306D) - - - $17.95
10. Edison Diamond Disc Fox Trots: 1920 - 1923 (DCP-307D) - - - $17.95
11. Rudy Vallée and His Connecticut Yankees: 1928 - 1930 (DCP-308D) - - - $17.95
12. Eddie Duchin and his Central Park Casino Orchestra 1932 - 1937 (DCP-309D) - - - $17.95
13. Jazz: It’s a Wonderful Sound! (DCP-500D) - - - $17.95
14. Ray Noble Plays Ray Noble (and others) 1935 – 1950 (DCP-310D) - - - $17.95 each
15. Novelty Tunes of the Roaring 20s, Volume 1 (DCP-308E) - - - $9.95 each (mp3 download only)*
16. Novelty Tunes of the Roaring 20s, Volume 2 (DCP-309E) - - - $9.95 each (mp3 download only)*

17. Handel's Messiah (DCP-398D) - (mp3 download only) - - - Free $0.00
18. Hootenanny '65, Cranford, NJ (DCP-397D-2) – (mp3 download only) - - - Free $0.00
19. Charlie Stiles – Gentle Touch – 1966 @ The Cranford Hotel (mp3 download only - - - Free $0.00
20. An Old Phonograph Christmas (compilation) – Circa 1930 – (mp3 download only) - - - Free $0.00
21. Pete Seeger Live Concert at Cornell University, 1957 (mp3 download only - - - Free $0.00)

*Visit Our Online music store at www.diamondcut.com to audition and/or purchase these tunes (via download).

For all other items, please include $2.30 per item to cover Shipping and Handling in the U.S. Europe, please include $4.20 per item.
N.J residents add 7% Sales Tax.

All of the Diamond Cut Music Albums can now be downloaded from our online store in .mp3 format at http://www.diamondcut.com/st3/product-category/music/ where they can be auditioned and/or purchased (via download).

To order, please visit our on-line store located at www.diamondcut.com, or please send your order including your check or money order payable in U.S funds to:

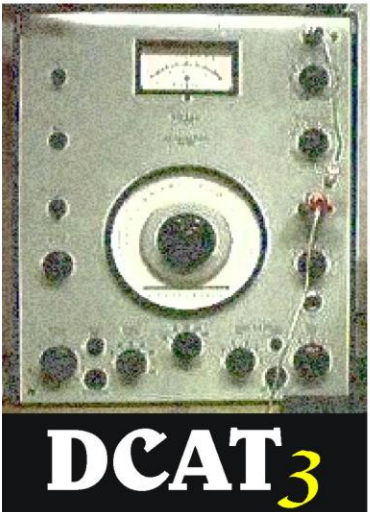
Diamond Cut Productions, Inc.
P.O. Box 305
Hibernia, NJ 07842-0305
Alternatively, you can fax your order to 973-316-5098
Visa, MasterCard, or Discover Accepted
(Wire Transfers or Checks are possible)
Or, call in your order at 973-316-9111
(Please call ahead if you are going to fax an order)

Contact us via our website or make your purchase at our online store located at:

http://www.diamondcut.com/st3/shop/

Note: Dealers, Retailers, and Record Distributors are invited to request a copy of our Product Discount Schedule.

DCAT-3 Audio Test CD Set



DCAT-3 is a comprehensive set of 147 audio signals contained on a 3 CD set, which is very useful for room acoustical response balancing and audio equipment testing and evaluation. Room acoustical response balancing can be performed with nothing more than a calibrated low cost sound pressure meter in conjunction with the DCAT-3's set of 31 narrowband random 1/3rd octave weighted noise signals.

All DCAT-3 signals are announced on a separate track allowing any signal to be repeated using the loop-play feature on your CD player. DCAT-3's signals are all tightly calibrated in frequency, amplitude and spectral distribution since they are all mathematically synthesized without the use of A/D converters. Sine waves are very low in harmonic distortion content while square waves exhibit excellent rise times while triangle waves display exceptional linearity. Also, none of the signals contain any annoying transients since each signal is smoothly faded-in and faded-out. DCAT-3 offers the user a great value at a very reasonable price. DCAT-3 contains the following signal types:

- Discrete Sine, Square, Triangle Waveforms
- Swept Sine, Square and Triangle Waveforms

- Random Noise signals including White, Pink, Brown, Subsonic and Seismic
- Weighted Narrowband Random noise in 1/3rd Octave increments (31 signals)
- Phase inverted and Quadrature signals
- Left & Right Channel Only signals
- Dual Tone signal for Inter-modulation Distortion Testing
- Calibrated silence (DAC counts = 0)

DCAT-3 Performance Specifications

Signal Type	Frequency Accuracy	Amplitude Accuracy	Distortion (% THD)
Sine	+ / - 0.005 %	+ / - 0.5 dB	0.0033 % @ 1 kHz
Square	+ / - 0.005 %	+ / - 0.5 dB	44 % @ 1 kHz
Triangle	+ / - 0.008 %	+ / - 0.5 dB	12 % @ 1 kHz
White	N/A	+ / - 0.7 dB	100 %
Pink	N/A	+ / - 1.5 dB	100 %
Narrowband Random (1/3rd Octave Steps)	+ / - 0.2 %	+ / - 1.5 dB	100 %
Swept Sine	+ / - 0.005 %	+ / - 0.5 dB	0.0033 % @ 1 kHz
Swept Square	+ / - 0.005 %	+ / - 0.5 dB	44 % @ 1 kHz
Swept Triangle	+ / - 0.5 %	+ / - 1.5 dB	12 % @ 1 kHz
Dual Tone	+ / - 0.005 %	+ / - 0.5 dB	0.008 % (high tone)

*The quality of your CD player's performance may degrade these numbers to varying degrees

To order, just go to www.diamondcut.com and follow the links to the software section of the online store.

Diamond Cut Software Product Model Number Nomenclature (English Versions)

DCArt10, v. 10.x – Full Version: **DCP-1010F**
DCArt10, v. 10.x – Upgraded Version: **DCP-1010UP**

DCForensics10 Audio Laboratory, v. 10.x – Full Version: **DCP-2110F**
DCForensics10 Audio Laboratory, v. 10.x – Upgrade Version: **DCP-2110UP**

DCForensics10.5 Audio Laboratory, v. 10.5x – Full Version: **DCP-3110F**
DCForensics10.5 Audio Laboratory, v. 10.5x – Upgrade Version: **DCP-3111UP**

DCForensics10.8 Audio Laboratory, v. 10.7x – Full Version: **DCP-3117F**
DCForensics10.8Audio Laboratory, v. 10.7x – Upgrade Version: **DCP-3117UP**

DC Forensics Adaptive Frequency Domain Filter (AFDF) VST Plugin: DCP-703

Diamond Cut Productions Virtual Valve Amplifier (VVA) VST Plugin: DCP-700

These products can be found at this link:

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