

Music Restoration Handbook

Early Disc Records to Today's LPs



Marc S. Hildebrant

6.0 Edition

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HCLLC

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Dedication

To My Wife, who continues to inspire me to achieve so much in my Life.

Marc Hildebrant



The Author and One of his Grandchildren Enjoying Restored Music!

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1.0 Introduction

1.1 About the Author and Handbook

Music has always been an important part of my life. At an early age I had a phonograph and records to play and enjoyed hearing music. I continued to add to my record collection using money earned from mowing lawns and babysitting and quickly developed an interest in all types of music from LP collections of early jazz music to the latest Rock and Roll music. If I liked the song, it was not important to me if the music was current or vintage.

During my teenage years, my father started buying very early record players and records. These old records had a cylinder shape with performers' names that I had never heard of. I helped my father sort and file the records as his collection grew and grew. At this time, 1960's, collecting early Phonographs and Records was just beginning to become historically important. Collecting early phonographs was still a hobby, not an investment, so that the price for these early phonographs and records were quite low. My father took advantage of this low cost to build up a large collection of this early technology. At his peak of collecting the number of cylinder records in his collection approached 2000 along with a couple hundred phonographs. In the afternoons after school, I would listen to and organize the latest batch of his old records. This went on for many years during which I developed a lasting interest in old music. While attending college I obtained BSEE and MSEE degrees and I started a career in Electrical Engineering.

As happens to many people, my life changed as I married my Wife and we started our new life together in the early 1970's. My interest in music continued but our money and time went towards providing for our growing family.

In the early 1980's I recorded some Edison four-minute cylinder records, using a tape recorder, located at my parent's home. A casual comment was made by my father stating that the sound recorded from a tape player with automatic level control seemed to sound better when played back than using the original phonograph. While his comment did not start me towards music restoration it did stay with me as I wondered if the music could be improved from the record with some modifications made to the original sound.

During the 1990's I had a modest collection of Edison cylinder records from my father's estate and at the same time located a software program from a company called Diamond Cut Productions; that provided me with abilities to create a digital version of the song, along with some basic clean-up tools for the music. At this time, the Diamond Cut Productions product, DCart 3, was provided on two floppy discs.

A DSP (Digital Signal Processing) class, taken for my Master's Degree in Electrical Engineering, gave me an understanding of the potential improvements possible for improving audio signals. Also, the steady increase in the computing power and decrease in cost for desktop computers provided a means to implement very complex DSP math on music files. My father's early observation about improving sound from the use of automatic level control provide me with some basic music improvements using Diamond Cut Productions software. My career continued as an electrical engineer with some occasional music restoration work.

Around 2010 my career as a full-time electrical engineer ended and my work changed to retirement activities along with Electrical Engineering consulting work. I returned to the Diamond Cut Productions software products and found that the company had made significant improvements to their earlier products. I purchased additional computer hardware and started a written journal of my music restoration work while using their software. I also joined the Diamond Cut Productions Forum and other related audio groups to discuss and learn about music restoration.

During this same period, I met an early record collector with a very large collection of both 78 RPM records and Edison Diamond Disc records, which I used for learning how to restore records for him from his collection; along with Edison Records and LP's that I owned.

While I was learning to use the software from the Diamond Cut Productions Company it became clear to me that there were limited resources available to guide you through the process of restoring music. The help files and owner's manual that comes with the software are useful, however, much of the needed knowledge is learned by trying a method and then listening to the results followed by another attempt and another. As I used the software tools to restore the music, I found that I would have to repeat my work many times as I was constantly learning new techniques.

Another problem I found in restoring this early music is that most of the technical information about recording and playback of 78 RPM and Edison records is only found in out-of-date books and magazines. Today, the technology of recording records is quite different as the sound is captured from the recording room and turned into a digital stream of data rather than a varying analog signal.

I decided to write this Handbook so that people with an interest in music from the early beginning to the latest LPs could enjoy their non-digital music again with the clear sound as recorded in the music studio and in turn store that music and play it back using today's technology. I have included specific information in this handbook that reflects the many hours of time that I spent learning how to restore music. Also, I wanted to introduce you to audio technology so that you would have confidence in your restoration of music by understanding the reasons for various software settings.

I assume that you have or will be using Diamond Cut Productions Software and would like to improve your techniques in the audio restoration process.

I am not an employee of the Diamond Cut Productions Software Company nor was I paid to write this Handbook by them; rather I found their products to be a useful tool for restoration and enhancement of music.

1.2 *The Layout of this Handbook*

This handbook covers a subject that is somewhat difficult to understand. The reason is that knowing how to restore music does not follow in a linear learning flow. What is meant is that the learning process does not follow learning one subject after another till you arrive at a final stage of understanding. The restoration process involves many interrelated subjects that are applied at the same time. As you are learning how to transcribe music you also need to understand the history of music, why the various equalization methods were developed, the correct number of digital bits to represent the music, etc. You really need to learn quite a variety of interrelated subjects as you start to producing a restored song. To aid in this Learning Process the Handbook has been divided into Three Major Sections.

The first section labeled *Basic Music Restoration Techniques* covers the start of the restoration where the music is transcribed from the original source and then converted to a digital music file. This new digital file will have distortion in the form of noise removed and a clean version is available for listening or advanced work.

The next section labeled *Advance Enhancement Methods* covers additional processing on this clean version of the music. These advanced topics include frequency modification to both the values present and the generation of new frequencies. A special section devoted to the improvement of Acoustically Recorded Edison Diamond Disc Records with techniques I have pioneered is present in this section.

The third section is a *Reference* for technical and historic articles about record creation and playback. This section provides a deeper understanding of the subjects covered in the previous parts.

1.3 Record Types Addressed in this Handbook

The records that are discussed and restored in this handbook are all types of *flat* records; early Acoustic 78 RPM records, Edison Diamond Disc records, 78 Electric Recorded Records, 45 RPM and LP records.

While early cylinder records will not be covered, the concepts used for these early acoustic recorded records could be successfully used to restore these cylinder recordings with little additional modification.

Tape recordings are a unique form of storing music that differs from records in significant ways and the help files in the Diamond Cut Manual covers most of the needed information to restore this type of music storage.

1.4 Diamond Cut Productions Software Used in this Handbook

The *Basic Music Restoration Techniques section* and the *Advance Enhancement Methods* use the DCart11 Software. Some sections in the *Reference section* use the DC Forensics11 software. The Forensics Version contains some additional software functions that are useful for measurements of equipment performance, described in the reference section.

All the restoration work can be performed with the DCart11 Software. If desired, the DC Forensics11 Software could also be used.

Basic Music Restoration Techniques

2.0 *Definitions of Restoration & Enhancement*

2.1 *Restoration*

Restoration of music is the effort to return the music from today's playback of the record to the same sound that occurred during the original recording session. If you listen to the final restored music, it should sound as if you were hearing the desired result the artists *would have intended* when they first created the music at the recording studio. The restoration process should remove the unwanted modifications to the music that have occurred through the entire process after the recording of the music in the studio to the manufacture of the records, and then playback of the record. All the distortion and noise from the limited recording technology used; the scratches on the record, the dirt and damage to the grooves, and the defects in the playback should be removed and *replaced* with an exact version of what was desired at the recording session.

This is a rather high bar for the restoration work but it is possible to come very close to this with the high-powered hardware and software tools available today. For the restoration process the sound is not modified with new echo or extra bass and treble; rather the goal is to return the music *To the Way it was Recorded*.

2.2 *Enhancement*

Enhancement of music is the effort to *add to* or *subtract from* the restored music frequency content and musical tones to achieve the restorer's desired sound. Some examples are adding echo or reverberation to the music, creation of a stereo sounding song from a monaural recording, or other changes that *you* may want to obtain the desired sound. There are many reasons to enhance the restored music. Some examples are:

1. A poor-quality recording was originally made from the master recording. All musical recordings must record the music using the limitations of the technology at the time of recording. Even today with the current state of the art in recording technology, poor microphone placement and studio acoustics can still produce a poor master recording.
2. The record producer's master recording did not capture the music as wanted by the musicians. The history of recording music details many times that the artist's desire in the sound of the final product was not what the producer and record companies'

management wanted. While new mixes of the music cannot be created, some modifications to the restored version may be closer to the artist's desired result.

3. The record producer modified the frequencies in the music to compensate for the playback equipment that used the current technology. The equipment for playback of records has improved from the original tin-foil recordings to today's high fidelity sound systems. The powerful stereo amplifiers and massive speakers available today can fill a listening room with a wonderful and magnificent sound that can satisfy the most critical listener. This potential *sound stage* is a far cry from the music playback systems that were used by many 78 RPM record users in the past. These 78 RPM record producers understood that if their recording sounded good on an average system *at the time the record was made*, then sales of the record could be strong. Therefore, the same record that sounded good in 1935 on a home system could sound poor today on a hi-fi system. The next example demonstrates that modification to the music for the customer's playback equipment may still occur today.
4. The current practice of using an algorithm to *compress* the music file (for example MP3) will introduce losses in the range and frequency content of the restored music. This common practice to save memory space seems *strange and not natural* given the current low cost of digital storage, however the fact remains that it does occur and can be somewhat overcome through various enhancement methods.

These are just some of the reasons that an enhancement to the restored music may be used. The Advanced Enhancement Methods section will provide details.

3.0 Motivation for Restoration of Music

3.1 Records Wear Out

Time is hard on all of us. A clean LP record when removed from its jacket becomes a magnet for dirt. Accidents like bumping the pickup arm while the record is playing usually creates a scratch on the record during the best part of a song (Murphy's Law).

The older 78 RPM records were often played with a steel needle or at best a very heavy sapphire electric pickup. This type of playback for these records would wear down the record grooves even though these records were made from a tough material (Ground up Rock with a Shellac Binder).

You will play your favorite records many times and they will wear out. Restoration can restore much of the original sound from these worn-out records.

3.2 Why Restoration of Music is Important and Worthwhile

The type of music recorded by the record companies has been constantly changing as the music tastes of the public have in turn changed. For Jazz music much of the early original music was recorded using acoustical technology and early electrical recording technologies. The ability to hear this music as it was originally recorded in the studio is just wonderful.

During the years that LPs (Long Playing Vinyl Records) were popular many people purchased them to hear the new and exciting Rock and Roll music. These LPs were played and enjoyed many times and developed numerous scratches. Later, as CDs (Compact Disc) and other Digital Storage techniques became available, the re-release of original artist's songs that were first heard on LPs had high musical expectation for a new noiseless version of the same songs once heard on these LPs. Instead, the new Digital versions of the original music seemed to have a harsh sound and did not contain all the songs that were on your original issued LPs. The common Greatest Hits CD collections seemed to omit your favorite song! Now you can create a Digital version from your LP's that will sound as good or even better than when you first heard the song.

The music that was recorded on records from the very beginning of recorded sound up to the end of the 78 RPM record era represents many unique and original opportunities for the listener of the restored music. One of the most important reasons is the ability to *hear* the original versions of various music styles played by the original artist. If you enjoy jug band

music and have an old recording of *Newport News Blues* by the Memphis Jug Band, you can hear the song, after restoration, in its original glory sung with the enthusiasm and phrasing that only the original recording artists could provide.

For Bing Crosby fan's you can hear him sing as a young man with the Paul Whiteman Rhythm Boys' knowing that he was destined for a long music career. The Dorsey Brothers, Glenn Miller, and many of the Swing band leaders got their start in early jazz bands in the early 1920s. When this music is brought back to life, the musical performances sound just magical.

When early music was recorded in the music studio a single microphone was often used to capture the sound from the performers. An initial, temporary recording was made using a very soft material from which the actual records would be later manufactured. Multiple takes could be done but were often limited to a small number as each new recording required a unique temporary record. This meant that when the musicians came into the studio, they had to be ready to record and were expected to make a high-quality balance sound. Multiple takes, remixing, echo added, and voice overs that are common in today's music were just not done. The music you heard on the record was what you would hear when the musicians played during a live concert. In the opinion of many people this type of music has a natural sound and the artists sounded authentic and realistic. Some would say that this is quite a contrast to the over produced, artificial sounding music produced today!

Another benefit with a live recording and the use of one recording microphone is that all the artists playing in the song would hear each other and the total blended sound for the recording. This gave the artists additional *feedback* to their own playing which in turn allowed them to balance the intensity and tone of their music as it was being blended into the recording.

Many of these types of recordings can only be heard on these early records. While today's music has wonderful digital recording with sound delivered with 16 bits and 44.1 thousand samples per second, these original recordings have an important and unique place for those that enjoy music in all forms.

3.2.1 Advantage of Using the Correct Playback Equipment for Records

Many people playing their music from CD's or other digital sources of music and are not aware that older electrically recorded music had major signal modifications made to the music before it was placed into the record grooves. These music modifications were later *removed* in the preamplifiers used to playback these disc records. The modifications were standardized for LPs and 45s but were not consistent for the 78 RPM or Edison Electrical recorded records. When an electrically recorded 78 RPM record is played on a current audio system meant for LP records, the music will sound dull and lack the upper frequencies. When the correct modifications are applied to the older records the muted high frequency parts of the music can now be heard and the excessive bass is removed.

Often the listener of an old record will be surprised to hear how good they can sound when played back with the correct equipment.

3.3 Early Music Releases from CD Restorations & Digital Downloads

The major music companies, while having access to many studio master recordings, have not produced many re-issues of older material. There are several smaller companies that have issued restored recordings and while some are very good many of the restorations consist of simply removing the high frequency tones in the recordings to reduce the noise. The removal of these tones reduces the scratchy sound of the record but also makes the recording sound dull and muted.

Since you have the advantage of restoring the music using *your* time and *effort*, the results that you can produce are as good as *you* want the restoration to be and can often exceed commercial music products.

4.0 *Tools Needed for Audio Restoration*

A workstation for your audio restoration can be as basic or as elaborate as your budget allows. A guide to the needed equipment will be shown, but you can increase the level of complexity and features to any desired level. A simple setup can provide very good results, whereas each incremental improvement to your audio system will cost greater amounts of money.

Basement space in my home was used for my current Music Restoration Studio. Refer to figure 4-1.



Figure 4-1 Music Restoration Studio



Figure 4-2 Record Turntable



Figure 4-3 Preamplifier and Analog Mixer



Figure 4-4 Analog to Digital and Digital to Analog Converter

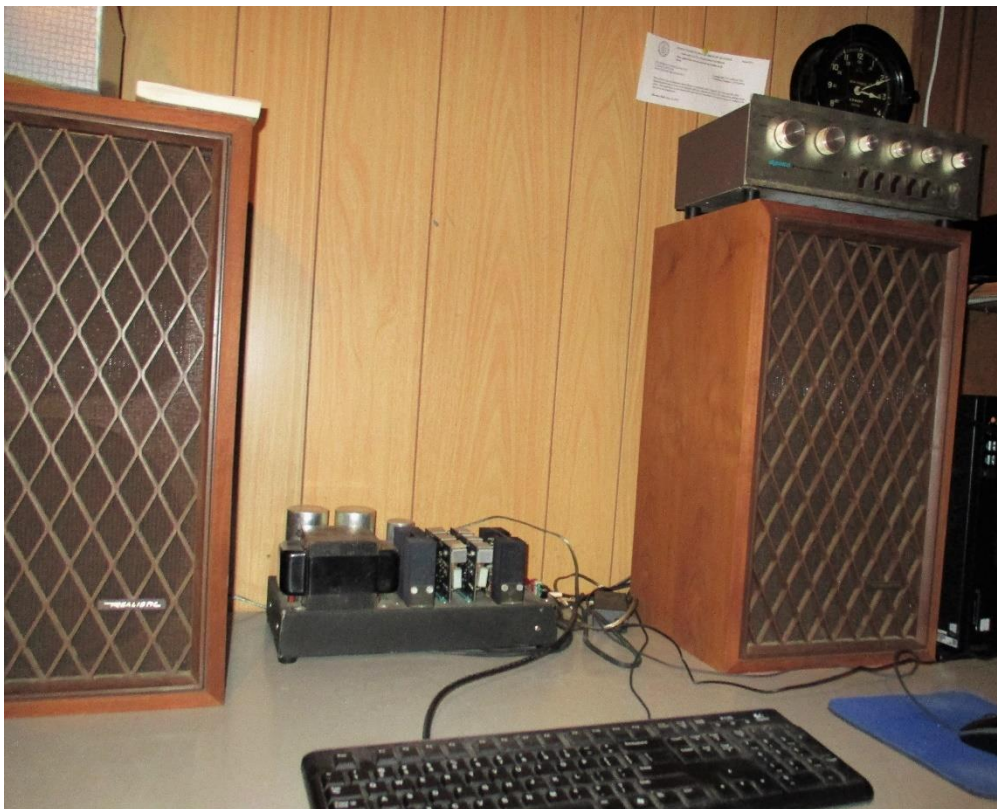


Figure 4-5 Preamp on Speaker with Playback Power Amplifier behind Monitor

4.1 *Restoration Software*

The software used in this Handbook is from Diamond Cut Productions, Inc. While the company sells several products, the specific one used is DCart11. Since the Diamond Cut Productions Company continuously up-dates and improves their products you can use future software updates from version 11 of their software with the examples shown in this Handbook. If you have an older version of the Diamond Cut software, you will find some differences to the software features shown.

DCart11 software works very well for audio restoration of music due to the large amount of control over its operation. Because the software offers many adjustments for the use of the tools this implies that *you* will need to understand their operation to achieve the best results. Once their operation has been learned, a large amount of the noise from the recordings can be removed with the resulting pleasure of clearly hearing the music again.

After the software is installed, you are urged to refer to the owner's manual and user's guide that comes with the software for help with the tools. This Handbook expects that you have some limited knowledge of the operation of the Diamond Cut Productions Software.

The engineers who developed these software products provided many adjustments and features in the implementation of their work. The intent of this handbook is to show what has worked in my journey to restoring music. You are encouraged to experiment and try to improve on the methods described within. The methods shown do work very well and are the result of restoring 1000's songs.

Diamond Cut Productions also offers a product directed toward Forensics Audio Work. This product, Forensics11, contains more software functions than DCart11 and can be used with no modifications to the methods shown.

4.2 *Computer Operating Systems Software*

Windows Professional Version 11 is the operating system for the computer. The Diamond Cut Productions web site should be consulted to see if a different operating system software can work for you.

4.3 *Audio Restoration Hardware*

4.3.1 *Computer*

The heart of the audio workstation is the computer. Many desktop systems can perform the needed software calculations with no problems. Where you can help with processing speed is to have a large RAM in the computer for system memory. The system currently has 8 Gigabytes of installed RAM that enables software programs to minimize disk drive activity which in turn, speeds up the program execution time. Disc drives are good for storing large music files but are slow for performing the many operations used in the software.

If possible, the computer should be dedicated to music restoration. In this manner the need for an Internet connection is reduced along with any potential problems from a software virus.

A large screen size for the computer monitor helps to display the details in the song and can show all the various software options that are available to use. The screen size shown in figure 4-1 is 24 inches, measured diagonally, which works very well for displaying the music waveform.

The cost of disk drive storage is low. The best set-up for workstation storage management is to store the completed audio files in all their forms (Original, Restoration Work, and Final Version), duplicated on two separate external disk drives. These disc drives can be attached to the computer via USB connections. The time spent on your music restoration work can be lengthy and the original recordings are not always replaceable. Having two duplicate copies of the audio files is an easy and low-cost method to safely store the restoration material. The organization of the music files will be explained in a later chapter.

The specific computer used in my music studio came for the ASUS company with an Intel i5-9400 CPU. The computer uses a solid-state disc drive for all software program executions, while an internal magnetic disc drive is used for all *working audio* files. The external storage uses USB magnetic disc drives.

4.3.2 *Analog to Digital & Digital to Analog Converters*

Since the music from the records is analog or continuous, a device is needed to convert from the analog to the digital domain for the Diamond Cut software algorithms. Your hearing

is analog so you must convert back to the analog world to hear the results of your music work. Many of today's computers contain a set of integrated circuits on the main board to perform these conversions. Another method is to have a converter installed in a PCI computer expansion slot for the conversions. These separate PCI plug-in modules are called *Sound Cards*. Still another method uses an external converter powered by a USB connection which sends the data conversions in real time to the computer over the same USB connection used to power the device.

These converters are critical to the success of your music restoration work. Refer to the Reference Section for detailed information regarding the testing and use of these converters. For a computer that contain a PCI converter or integrated circuits on the main board the audio input and output signals should use the *line-in* connectors. The standard computer colors are blue for *line-in* and green for *line-out*. The definitions of the signal levels for *line-in and out* are described in the Reference Section of the Handbook.

If the sound card has a microphone input, it will be a pink color. The microphone input uses an additional gain stage over that of the line-in stage and is not useful for our audio restoration work. If your computer does not have line-in and line-out connections, then a separate converter must be purchased.

One of the converters in my music studio is an external USB powered device, figure 4-4. This converter came from the Focusrite Company and is a model Scarlett 2i2 Gen3. The ASUS computer contains a Realtek Chipset that can also be used as an A to D and D to A Converter. I use multiple converters in my studio for convenience when using other analog sources than a turntable.

The operating system software allows you to vary the gain of the audio conversion for the line-input and line-output levels via *software* slider controls. The best result for music restoration will occur when the software sliders are all set for maximum gain for both input and output and external gain controls; and external knobs used to adjust the overall recording and playback volume.

The sample rate and digital word length for the converters should be set to a rate of 96 thousand samples per second and 16 bits of word length for recording. The adjustment of these values will be covered later in this section and in the transcribing music section, 7.0, in detail. There are several locations in your music system to change these values and you must exercise caution that *you have really made a change and are using the correct values when you record your music*.

4.3.3 Listen while Recording

The ability to hear the sound that you are recording during the transcription process is necessary to avoid a poor-quality resulting source file. You can achieve this using two separate methods. First, the record signal from the preamplifier can be connected to both the line-in connection on the converter and to an audio amplifier and speaker. Some preamplifiers have multiple outputs that enable this type of signal connection. The second method is to use the converters software to perform this ability. The software may refer to this setting as loop-back or listen. Often these settings are in the settings/audio section in Windows software or with the manufacture's converter software.

4.3.4 Turntable & Cartridge

Record turntables to playback the music have been made since the beginning of recorded music. For your restoration work, a used or new turntable can be used. The basic requirements are:

1. Speed Steps from 33 $\frac{1}{3}$ to 78 RPM.
2. Low Mass Tonearm.
3. Adjustable Skate and Stylus Tracking Settings.
4. Ability to mount and use today's Stereo Cartridges.
5. Several different Stylus sizes for the Record types.

While used turntables are available, always purchase a model that is working well and does not need new belts and drive wheels as many of the replacement parts are not widely available.

A current manufactured stereo cartridge should be used; for example, the Ortofon series, Grado Labs and others that accept various stylus sizes.

New turntable models range from simple USB types with a one size fits all to state-of-the-art models. My first turntable was a very used Garrard Turntable. I currently use a RELOOP RP4000 MK2 as shown in figure 4-2.

The USB type turntables that include the A to D converter should be carefully verified that they can provide the needed sample rate, bit depth, and low noise level for your restoration work along with an appropriate preamplifier for Non-RIAA Records (see the next section).

4.3.5 Preamplifier and Gain Control

The electrical connections from the turntable to the analog to digital converter pass through a preamplifier. This preamplifier should have a *flat* frequency response meaning that the gain will be the same for all audio frequencies that it amplifies over a minimum range of 20-20,000 Hz. Many record preamplifiers in the past and many currently available have a frequency response that is not *flat*; rather it conforms to the *RIAA* (Record Industry Association of America) equalization standard. This RIAA equalization is covered in the Equalization Chapter and the need for this EQ is explained in detail in the Reference Section. Some preamplifiers that have an RIAA response also include a means to disable this feature and can be used. Although the DCart11 Software includes methods to remove the RIAA response from the transcription, the best approach is to not use the curve in the first place. The RIAA EQ (equalization) curve is correct only for LPs (Long Playing Vinyl Records) and 45s. The curve is not *correct* for the early records that were made prior to LPs and 45s. Very good restoration results can be obtained by removing the RIAA curve in software while recording non-RIAA EQ records, but the best method is to start all transcribing operations with a flat preamplifier.

The electrical *load* for the cartridge produces a significant effect on its performance to provide a faithful version of the music. Since this is important, the cartridge manufacture specifies the correct load on the output signals in terms of resistance (Ohms) and capacitance (Pico Farads). The preamplifier must present the correct value as a load for each channel. In most cases the value for the resistance is 47 thousand ohms and a very small value for the capacitance. You should verify that the correct load is in place in the preamplifier by checking the owner's manuals for the preamplifier and cartridge. A Diamond Cut Productions Flat Preamplifier DCP-47K-F (figure 4-3) is used in my studio.

All recordings will use the stereo channels, left and right, for records whether they are monaural or stereo. The correct monaural music sound will be created later in software and in no case should the stereo channels be wired to achieve a monaural sound. Separate, individual left and right channels should enter the converter for *all types of records*.

The gain of the recording path must be adjustable and have a maximum value more than adequate to saturate the Converter's input while playing any type of record. The best gain controls are physical knobs that are part of the preamplifier and rotate to change the amount for each channel. While it is possible to vary the gain with a slider software control that is mouse driven the use of a physical knob is the best way to maintain the optimum gain during recording.

If individual gain controls for Left and Right Channels are not possible in your equipment at this time, a method will be shown in the transcribing chapter to modify, after the initial recording, each channel's gain in software.

As I was building my Music Studio, I found that the Flat preamplifiers available at the time often had a fixed gain. One method to vary the gain used a Diamond Cut Production preamplifier (DCP-47K-F) connected to the turntable (figure 4-2), with the output signal of the preamplifier connected to a Behringer analog mixer model Xenyx 502 (figure 4-3). The output from the mixer was then connected to the line-in Realtek Chip Set in the Computer.

Another method used the variable gain available in the Focusrite product. A close look at figure 4-4 shows that the Focusrite converter has individual Left and Right channel gain controls. This allows a direct connection from the Phono Preamp to the Focusrite converter where I set each channels gain (the Focusrite converter has more than enough gain to saturate the A to D process).

4.3.6 Playback of Music during the Restoration Process

While you are restoring your music, you will want to hear the results on a good quality audio system with the speaker's location near the computer display. The front cover of the Handbook shows my first music studio where I placed monitor speakers on a simple wooden desk.

During the music restoration process, you will be listening *and* watching the display as the music is played. While you are watching the display of the music you will be hearing the sound at the same time. This combination of seeing and hearing is very powerful for finding the location of noise events in the music. The best location for the speakers will be on both sides of the display and close to your ears. If the speakers are located away from your hearing, then you may miss a noise burst in the music. Refer to figure 4-1 for a set-up.

The quality of the audio system for playback should be very good so that all the frequencies in the music can be easily heard. The need for a good playback system can be a problem since large physical speakers are needed to reproduce very low notes. A good compromise can be found with bookshelf size speakers as shown in figure 4-1. Another method for music playback is to use high-quality headphones.

The playback amplifier in my Audio Studio is shown in figure 4-5. The power amplifier is located behind the monitor and the preamplifier is on top of the speaker on the right side.

When the noise from the music has been removed and the music file is ready for the final effects to be use, the type of playback system has another important function. The

quality of the restored music is greatly influenced by the type of playback equipment used. When you play your restored music, you may use a variety of devices from headphones, high end audio systems, and portable CD players. A restored song should sound good on as many types of audio systems as possible.

Today's production of commercial music uses special monitor speakers that simulate the most common type of listening that will be used. This extra level of expense maybe useful for some however you can use your own judgement to produce good results by playing the restored music at home or in your car.

4.4 Control of Converters

The Analog to Digital and Digital to Analog Converters require a method to communicate the desired sample rate and bit depth. There are two methods that Diamond Cut software uses for communication; namely software drivers from Microsoft or drivers from the Steinberg company known as ASIO (Audio Stream Input/Output). Other companies may provide different communication but Microsoft and ASIO are the ones supported with DCart11. The ASIO driver operates within the Windows operating system, but it provides its own communication path to the supported device, which provides low system overhead for sending music files to the A to D and D to A converter. In addition, the use of the ASIO drivers provides a direct setting of sample and bit values for the converter, whereas the Windows software can modify the desired value set by the user, to the device. The first step is to check that the soundcard is enabled.

4.4.1 Setting Windows Privacy for Microphone

The Windows Operating System contains a *Privacy* option from the main *Settings* window that controls what software programs can communicate with the record function on your Analog to Digital converter. Windows Software can use the term *Microphone* rather than line-in or A to D.

To allow the Diamond Cut software program to control the A to D converter, the *Microphone* setting in the Privacy control for Desktop Apps must be turned On. Refer to figure 4-6 for the correct settings:

Allow desktop apps to access your microphone

Some apps and Windows features need to access your microphone to work as intended. Turning off this setting here might limit what desktop apps and Windows can do.



Some desktop apps might not appear in the following list or are not affected by this setting. [Find out why](#)

Sort by: Name ▾





	Diamond Cut DCart10 Last accessed 1/8/2021 10:57:14 AM
	Diamond Cut DCFORENSICS10 Last accessed 2/7/2021 9:31:02 AM
	Host Process for Windows Services Last accessed 11/9/2020 3:44:42 PM
	Windows host process (Rundll32) Last accessed 2/7/2021 9:28:02 AM

Figure 4-6 Application Control of A to D

Note: If you have not used Diamond Cut Software in the past then you will not see the Icon in the history under name. You will see the Icon when used. If you have a security program installed, for example Kaspersky, then additional software commands maybe needed to enable the converter.

4.4.2 Communication to the Internal or External Converters

The DCart11 software communicates with the converters using Microsoft or ASIO drivers. The choice is made in the Edit/Preferences/Sound Card setting. Refer to figure 4-7 for the ASIO selection and 4-8 for the Microsoft selection.

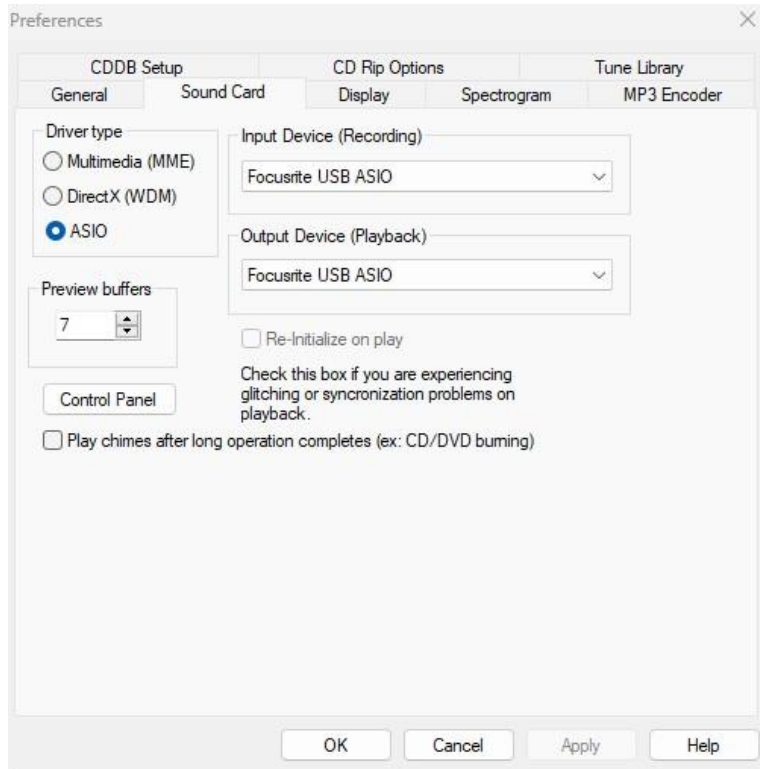


Figure 4-7 ASIO Selected for Software Communication

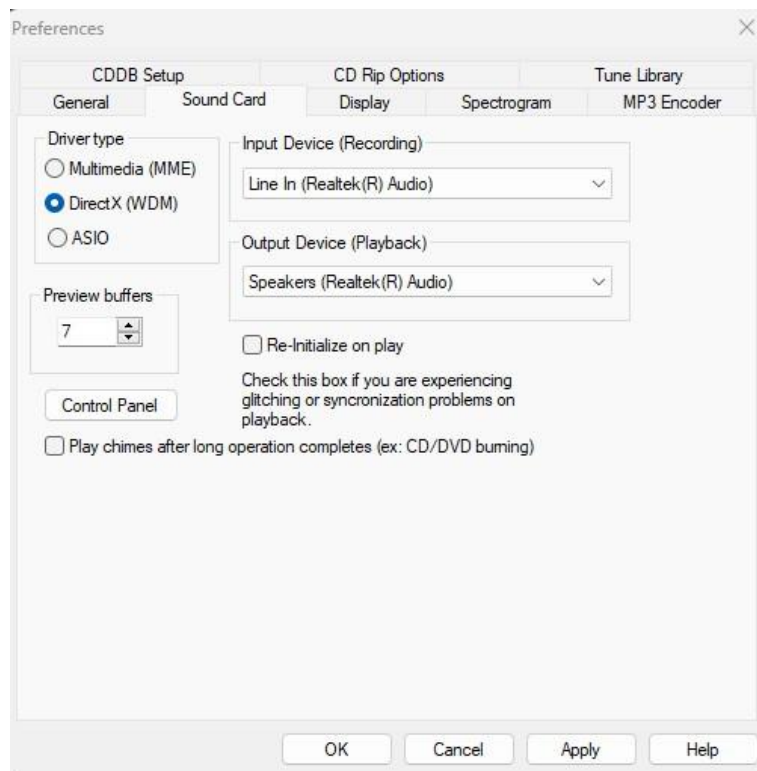


Figure 4-8 Windows Driver selected for Software Communication

4.4.3 Setting the Sample Rate and Bit Depth

When the communication path has been selected, the DCart11 software provides a user interface to the sample rate and number of bits for the file that you will be creating using the Record Functions' options.

When the music is played back, the Digital to Analog converters' settings will also be set to the correct value by the Diamond Cut software which reads the music's header file for the correct value and sends the sample rate and number of bits to the converter.

The *actual control* of the converter will be via Windows Drivers or ASIO Devices. The ASIO software control will next be shown.

ASIO

Figure 4-9 will *show* what has been selected and what the *header file* for the recorded music indicates for an ASIO device.

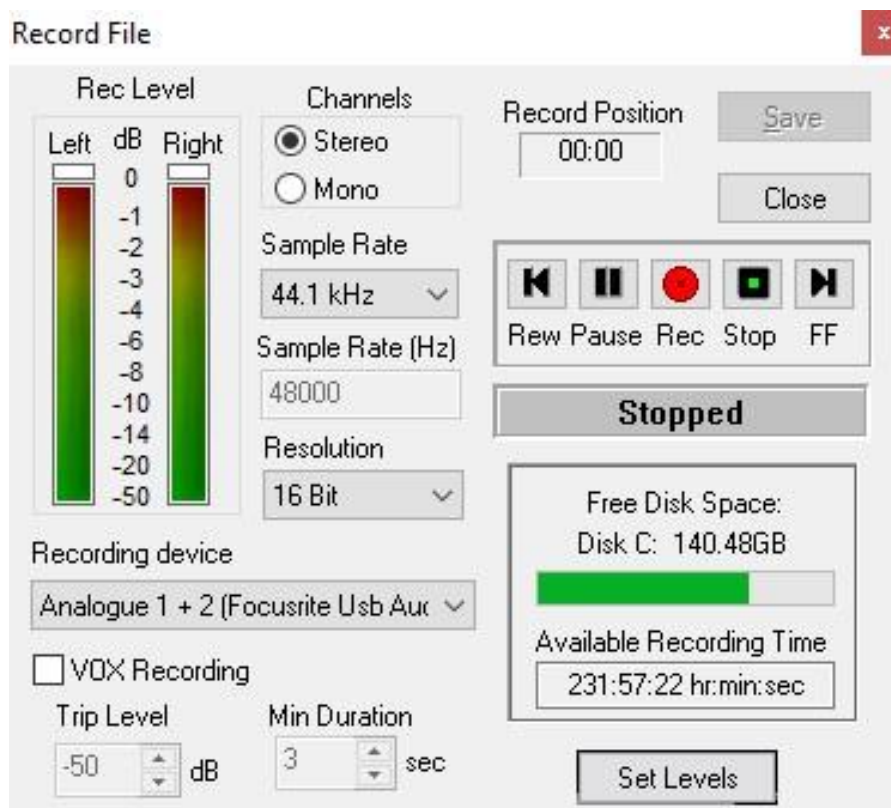


Figure 4-9 Sample Rate and Bit Resolution Settings for ASIO Device

When the Stop button is selected, the sample rate and resolution can be modified. The actual change to the ASIO device occurs when the Pause or Record button is selected. The Focusrite company and other ASIO devices can provide an Icon on your Desktop that shows what your device's sample rate and bit values are. After selecting the desired value via the DCart11 Record function, check that the desired value is set on the ASIO Icon.

The Windows Control Panel/Sound Tab also provides a method to verify the settings from the Record Function in Figure 4-9. If the Converters ASIO software is integrated into the Windows Software, the user selected values are shown. However, values shown in the Sound Panel can be incorrect; the software ASIO Icon provided by the manufacture should show the selected value.

The header file written by DCart11 will correspond to the selected values in the Record Window. Next the same process will be used for converters under Windows Driver control.

Windows Software Control

Figure 4-10 will *show* what has been selected and what the *header file* for the recorded music indicates for a Window Controlled device.

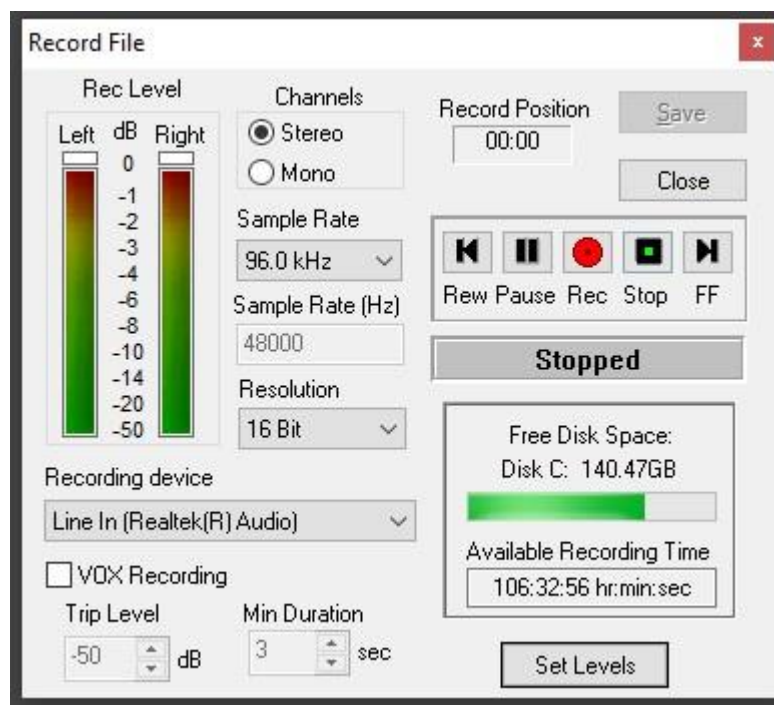


Figure 4-10 Sample Rate and Bit Resolution Settings for Windows Controlled Device

The Windows Operating system controls the sound properties via a software driver. One path to this driver is through the *Control Panel Hardware/Sound window*, and then the various tabs can direct you to the recording and playback sections which in-turn will allow control of the sample rate and bit resolution though the *Advance tab*. The control of the A to D and D to A values at this point may or may not be what was set in Figure 4-10. One requirement to enable DCart11 control of the converter is the use of a software option in the recording tab called *exclusive control*. This box must be enabled so that the DCart11 software has total control over the settings. This exclusive control option may still not work at times due to the Windows Drivers operation. Understanding *the source of this problem* involves understand the internal operation of the Windows sound driver, which is a difficult task. The new ASIO feature was added to DCart11 to avoid any failure to set the sample rate correctly due to the operation of the Windows Driver. The way to set the sample and bit values correctly, is to:

1. Set the desired Sample Rate and Bit Depth via the Windows Control Panel (Also via the Settings Window).
2. Set the same desired values in the DCart Record Function when Stopped.
3. Press the Pause Button.

4.5 *Overview of DCart11 Software Operation for Restoration*

The Diamond Cut Productions Company provides guides to learning how to use their product. The *Getting Started Guide and User's Guide* should be reviewed to learn about the operation of the software. An overall description of the restoration operation will be described, and addition chapters will add more detailed information.

4.5.1 *The Operation of the Software*

After the Diamond Cut Productions DCart11 software is installed on your system for the first time, you will have several options to pick for the look and feel of the product. A screen shot of the initial layout that I use is in figure 4-11:

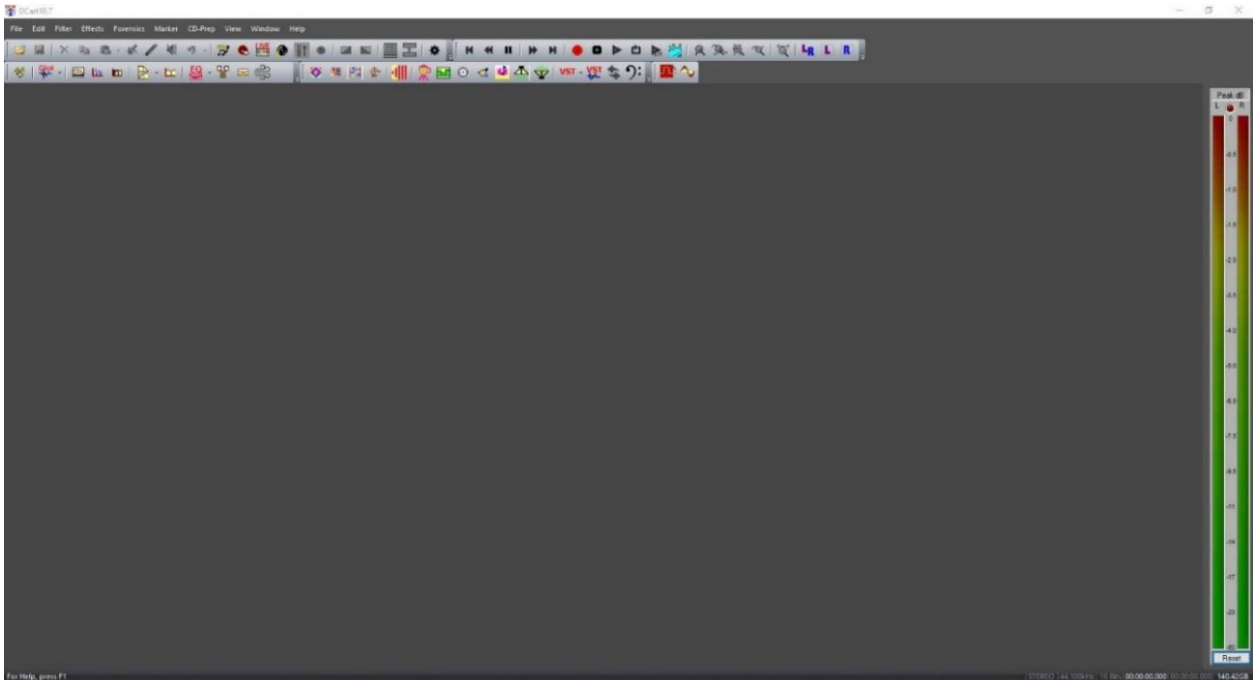


Figure 4-11 Initial Screen

At this point the various preferences have been set under the edit tab, the toolbars that you want are displayed, the amplitude graphs are show on the right, and you have selected *Classic Mode* for editing. Next, a source will be opened with a typical result in figure 4-12:

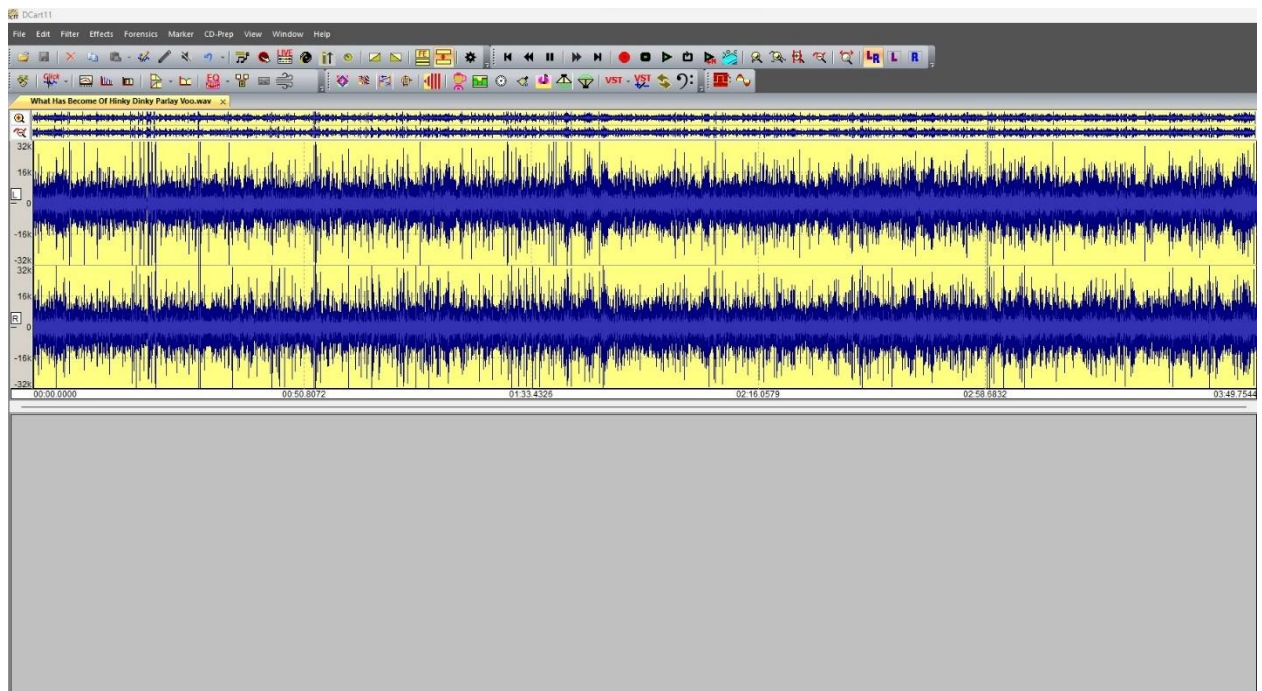


Figure 4-12 Source File Opened

The basic operation of the restoration process is to start with a source file containing your music to restore and then, by using a series of filters and effects, remove the distortions in the music.

The result of the software changes to the source file that uses filters or effects will occur in the destination file. If you are using the *classic edit* mode, then you will have the original source file at the top and the destination file directly below the file. The toolbars in the top of the window allow you to play the selected source or destination file. The result of applying any operation to the source file will, at most times, produce a destination file. The result of a filtering operation on this source file is shown in figure 4-13:

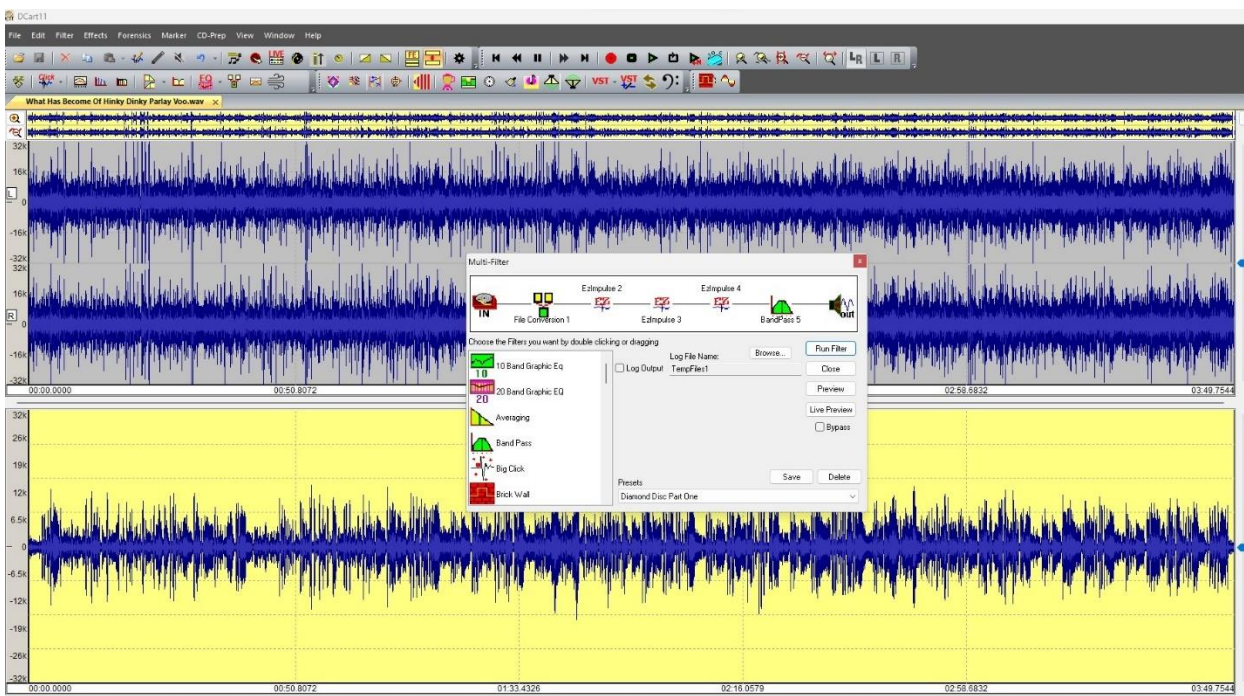


Figure 4-13 Source with Filtered Result in Destination

Notice in figure 4-13 that the original source file has two channels while the resulting destination has one channel. The destination file is highlighted (yellow) to indicate that it is now the active window and the multifilter used to create the destination is shown on the destination and source window.

The documentation for the various filters and effects will show you how to use each one of these software tools and as you use them you will learn what they do and how they work.

Some of the operations that you perform on the source file will only modify the file without creating a destination file. For example, if you want to remove a section of the source

file, then you will highlight the section and *cut it out* by pressing the control and X key at the same time. A new destination file will not be created; instead, the source file will be changed in length. You can recover the removed section via the *undo* edit feature if you removed the wrong part.

The DCart11 software has many features that will become second nature to you after you understand what they do. The software does contain help sections and the reason that the software looks *complicated* is due to the many options available.

4.5.2 Multifilters

The restoration process involves several individual applications of filters and effects to transform the original music file into a final clean sounding piece of music. Each application of a software algorithm upon the source file requires data conversions and a useful technique is to place several operations together in a user defined *multifilter* so that a major operation on the source file can be implemented in one custom filter that you have created. By having the individual algorithms connected in a large multifilter the number of data conversions and potential loss of accuracy will be minimized. This multifilter concept will be used in this Handbook for all the restoration operations.

4.5.3 Music Storage and CD Creation

Within the DCart11 software, you can create a music library database that stores all the restored files along with a useful search and playlist creation. The software also provides a method to create specific CDs that contains your music in this *Tune Library*. The Tune Library will be covered in detail in the Basic Restoration Section.

5.0 *Music File Organization*

How you organize your music files on your computer is important and is the *first* step in the restoration of your music. You will be able to save a large amount of time during the restoration process by creating a specific method for the organization of the produced music files.

The reason that the organization of your files is important is that you will often find new methods for various filtering and enhancement operations after you have spent considerable work cleaning up the noise from the music. If the resultant intermediate work is retained, then you can return and build upon your past work while using your new techniques rather than starting again at the beginning of your restoration work.

Another reason to organization your music files is that you may want to return to the original transcriptions if you made a major mistake by erasing a file or changing the music in a manner that cannot be repaired.

Today's digital storage devices provide the ability to store large amounts of music files for a low cost. The ability to provide multiple versions of the same song in different stages of restoration is very practical.

A consistent method that you use for placing the music in file folders will not only help you find a specific piece of music on your storage device but will also provide a predictable method to easily bring your music into a music library. The DCart11 software provides a useful music tool referred to as the DC Tune Library; and the structure of the music folders will work with this library. A later chapter in the Basic Restoration Section of the Handbook will detail information about the DC Tune Library. The music files are in three major locations which are:

1. **Original Recordings.** This is where the result of the music transfer from the A/D converter is stored. These recordings will have had no equalization applied to them (flat transfer) and they have been played at the correct speed. The initial Raw Recording may need some software processing before they are stored as an original file. This processing will be discussed in the chapter on transcribing the music.
2. **Restoration Files.** This is the working location for the music restoration process. Various sorting of songs and levels of restoration are located here.
3. **Restored Music.** This is the location for the latest versions of the restoration results that can be used for listening or for making CDs, or other digital distributions.

5.1 *Music File Type Used in The Handbook*

The *wav* file type is used in the Handbook. This is the best type of file to use as it is a universal music file and is not compressed. While the size of these files can be large the need for large storage space is not a problem with current storage technology. The *wav* file is the type that the DCart11 software operates on and is a standard within Microsoft Software.

5.2 *General Organization Method*

The folder names for the locations are shown in an *italic* style type.

Music

The top level on your mass storage device will start with *Music* as that is what we are storing.

Original Source Music Recordings

This file folder is below the *Music* top level and contains the Original Music Recordings. It is important to have a special place to keep the initial transfers of your music from the records.

Restoration Files

This file folder is below the *Music* folder and is the location for all your working restoration files. This is the place where you perform your restoration. Various levels of restoration will be created so that if you want to return to a previous level of work for possible improvement you can do so without having to re-do all of what you have previously done. This concept is very important and was learned the hard way!

The layout of the restoration files contains various *Level* folders that correspond to the place in the restoration process that the music files presently reside. For example, *Level 1* contains the original music files that are ready for multifilter noise removal. *Level 1A* contains

the music files that are so noisy that new technology will be needed to fix them. The *Level 2* files have had the noise removed and proper EQ (Equalization) applied.

The layout of the *Level* folders within the Restoration Files will have additional structure to separate the song titles and artists. The specifics of the layout will be shown later in this chapter, while the details of how to do the restoration work will be described in later chapters of the Handbook.

Restored Music

This folder is below the main *Music* folder and contains the final product of your restoration work. These music files may come directly from *Level 2* in the *Restoration Files Folder* or may have some additional enhancement applied to the music.

5.3 *Specific Folder Naming within Major Locations*

Under each major location (*Original*, *Restoration Files*, and *Restored Music*) will be a similar layout for the music files with minor changes to accommodate the difference between individual records, albums, and levels of work. While the original concept for a record *Album* started with paper folders containing 78 records (Album of Music), the use of the phrase *Album* has become associated with LP records, since LP records often contain multiple songs. The LP Albums can also contain songs from either one artist or a theme of music.

Due to the differences between Single Records and Albums, a slightly different folder arrangement for each will be used. The DC Tune Library can work (with slight preference changes) with these different layouts.

5.3.1 *Original Source Music Recordings Layout*

Individual Records

This layout is used for any single recording of a song within the Major Folder *Original Source Music Recordings*. The first folder is the record type such as *78 Electric* or *78 Acoustic* (these terms will be explained in the following chapters and in the Reference section). The

next folder down will be the *Artists Name* followed by the title of the song. An example for a 78 RPM Song is shown with a slash (/) separating the folder names:

Music/Original Music/78 Electric Records/Al Dexter/Pistol Packing Mama.wav

LP Albums

This layout is used for any *LP* record within the Major Folder *Original Source Music Recordings*. The first folder is the *Artists Name* followed by a folder with the *Title of the Album*. Next folder down will be the song in this Album. An example using the same symbols as previous used:

Music/Original Music/LP/Alicia Bridges/Disco Round/I Love the Night Life.wav

5.3.2 Restoration Files

Additional folders are used to define the stages of our music restoration work. The * character will indicate that the number used will be 1,1A, or 2 for the different levels of work. These levels contain your restoration results and music with the same artist/title will be in both *Level 1* and *Level 2*. *Level 1A* will use artist/title for the music with no restoration work performed. The level method will be explained in more detail in the chapter on General Noise Removal.

Individual Records

This layout is used for any single recording of a song within the Major Folder *Restoration Files*. The first folder will be the record type such as *78 Electric* or *78 Acoustic*. The next folder down will be the *Level Number* followed by the *Artists name* followed by the title of the song. This layout will be used in this example with a slash (/) separating the folder names:

*Music/Restoration Files/78ElectricRecords/Level */Al Dexter/Pistol Packing Mama.wav*

LP Albums

This layout will be used for any LP record within the Major Folder *Restoration Files*. The first folder will be the name *LP*. The next folder down will be the *Level Number* followed by the *Artist Name*. Next down will be the *Title of the Album* followed by the song in the Album.

An example using the previous used symbols:

Music/Restoration Files/LP/Level/Alicia Bridges/Disco Round/I Love the Night Life.wav*

5.3.3 Restored Music Files

This folder will be used by the DC Tune Library for your restored music and removes the use of levels.

Individual Records

An example is:

Music/Restored Music Files/78 Electric Records/Al Dexter/Pistol Packing Mama.wav

LP Albums

An example is:

Music/Restoration Music/LP/Alicia Bridges/Disco Round/I Love the Night Life.wav

5.4 Summary

A structured system of file locations for restoration work will provide access to a specific song by an artist and the ability to return to your restoration work so that new methods of restoration can be applied without a re-do of your previous work.

6.0 *Music Equalization & Removal*

6.1 *Overview*

The reference section describes how audio technology began by using only the sound power of the music to directly record the sound waves and then progressed to the use of electrical devices to amplify the music collected by the microphone before it was placed into the master recording. The time during which only the sound power produced the recording is referred to as the *Acoustical Era* or *Acoustical Recording* and for electrical recording the word *Electric* will be used. LP and 45 RPM will not need any extra labels as these recordings used only electrical devices.

Electric recording of music modified the music before making the recording master disc for two important reasons. The first reason was that modifications to the lower music frequencies were needed to avoid a very wide groove width to contain the amplitude movements. The second reason helped to reduce the surface noise of the record which allowed the higher musical frequencies to be heard. With these special modifications made to the frequency of the music the low frequencies could use the same groove width as the medium and high frequencies and the surface noise of the record would be reduced.

The application of frequency modifications *prior* to recording the master record and the application of the inverse of these modifications *after* playback of the record produces an accurate sound that was heard originally by the recording engineer in the studio. Details will follow with additional information in the Reference Section.

6.2 *Definition of Terms*

1. Acoustical: Only sound energy was used to produce the master recording. This was the original method to record music.
2. Flat: Means that the amplitude of the music frequencies that passes through the electrical circuits to the A to D converter are not modified depending on their frequency value.
3. Equalization or EQ: A modification is made to the frequencies' amplitude as it passes through the circuits to a specified frequency response curve.
4. Constant-Amplitude refers to a modification of sound waves so that the amplitude of the signal will remain constant as the frequency decreases, for constant input signal

power. To achieve this, the signal power of the actual recorded sound wave must be reduced in relationship to the frequency as it changes.

5. Constant-Velocity refers to a property of sound waves such that the maximum velocity of the signal stays fixed as the frequency changes, for the same value of signal power. The amplitude of the signal must change in relationship to a change in frequency to maintain a constant-velocity and will steadily decrease with frequency, for the same value of signal power.
6. Turnover Value: A frequency where the response of the sound wave changes from Constant-Velocity to Constant-Amplitude.
7. Pre-Emphasis: A time constant in the filter's circuit which corresponds to a specific frequency, where the Constant-Velocity Response changes to an Increasing-Velocity with increasing frequency (Acceleration); for the same value of signal power.
8. Rolloff: The amount of amplitude correction needed to remove the added boost from the Pre-Emphasis application.
9. Reference Point: A frequency that had no modification made to its amplitude; used for dB (decibels) measurements which requires a reference or 0 dB value.

The Tradition way to refer to the modifications are:

1. The modification to the playback signal uses the Curve Name.
2. The modification to the recording signal uses the phrase "Reverse" before the Curve Name.

For example, the words Reverse RIAA Curve describe what is used to record the initial master record. The RIAA Curve would then remove the original record modification. While this usage may seem backwards it is the method used in the music business.

In the chapter on transcribing records, it is recommended that for the transfer of the music from the record to the digital domain a *flat preamp* should be used which means that no equalization will be applied by the preamp circuits.

If a flat preamplifier was not used, specific methods will be shown to produce a flat recording from an initial EQ applied by the preamplifier. Since the use of a flat preamplifier means that no correction of equalization is applied; when the music is heard using this type of preamplifier from an electric recorded record the sound will be quite different then when the correct EQ is used. The electric recorded music with no EQ (flat preamplifier) will sound tinny and thin to you and that is O.K. at this time. If the preamplifier applies EQ when an

Acoustic record is played, rumble and lack of high frequencies will be heard. If the Acoustic record is played with a flat preamplifier the music will have a natural sound.

When you listen to the music during the restoration process you will want the proper equalization applied so that you can hear the music in a natural manner. The reason for the flat recording for the original recording is related to the operation of the noise filters and to give you flexibility to apply the specific EQ that you will later use to create a restored music file.

Several different methods will be shown to apply and remove the frequency modifications in software.

6.3 DCart11 Equalization Tools

The DCart11 software provides two distinct methods to apply a frequency modification to the recorded music to achieve the proper EQ or to remove specific types of EQ from the recording. The Paragaphic EQ software has a large selection of different frequency responses to apply to the music along with a clever graph that displays the response. Another method uses the Virtual Phono Preamp that has options selected with software buttons. Both methods have several preset responses available. Refer to Figure 6-1:

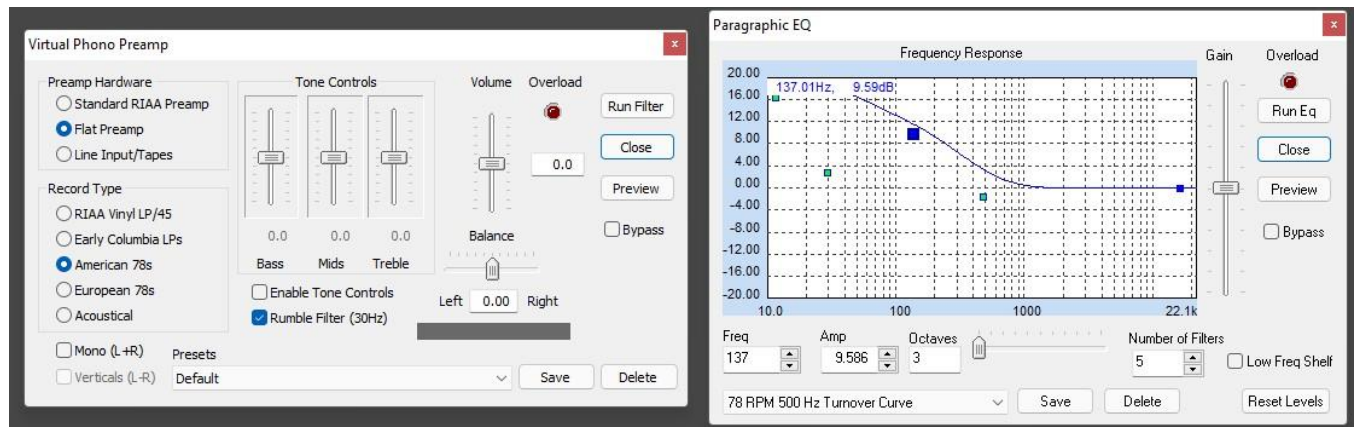


Figure 6-1 Software EQ Methods

The Paragaphic EQ with a 500 Hz turnover is shown in figure 6-2:

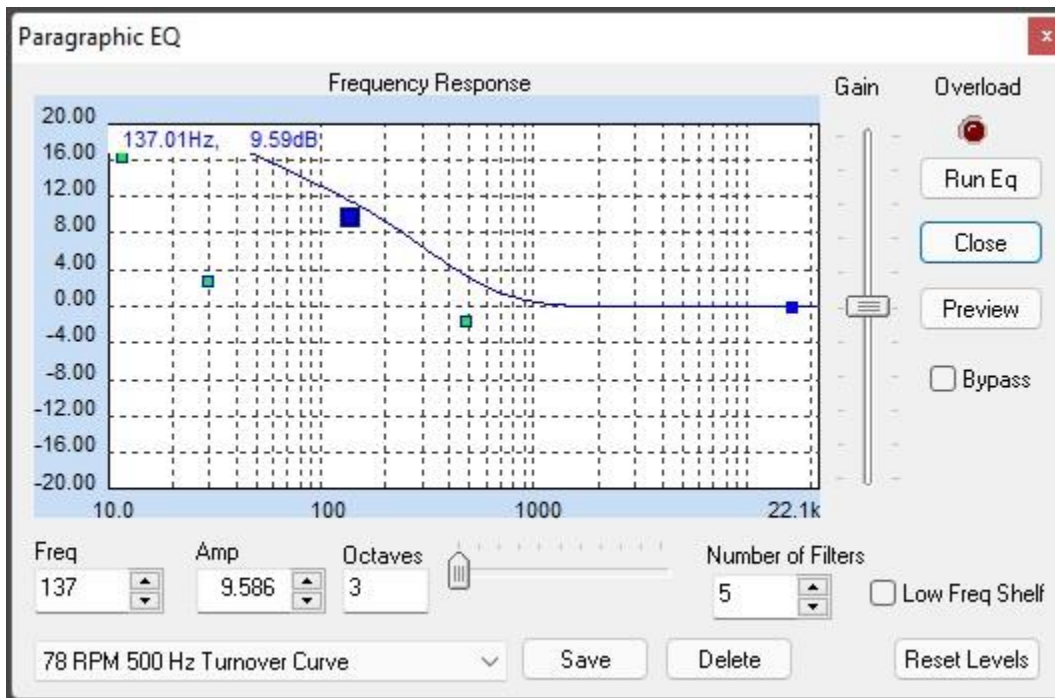


Figure 6-2 Turnover at 500 Hz

Notice that the low frequency response changes at the turnover frequency to have a gain increase of 6 dB per octave (doubling of frequency) and this gain continues to increase with lower frequencies. Prior to the turnover value, the gain versus frequency is a flat response. The reference point for the turnover frequency is 1000 Hz and at 500 Hz the response is +3 dB from the 1000 Hz value. This reference value of 1000 Hz is often assumed but not stated.

There is an additional option on this filter, the Low Freq Shelf option. This option takes the normal 6db per octave slope of the curve and changes its slope to a flat shelf around 150 Hz. This option can reduce the amount of low frequency boost to the recording at the lowest part of the music frequencies where rumble and other sources of noise could be present. Refer to figure 6-3 to see the effect on the same 500 Hz turnover curve.

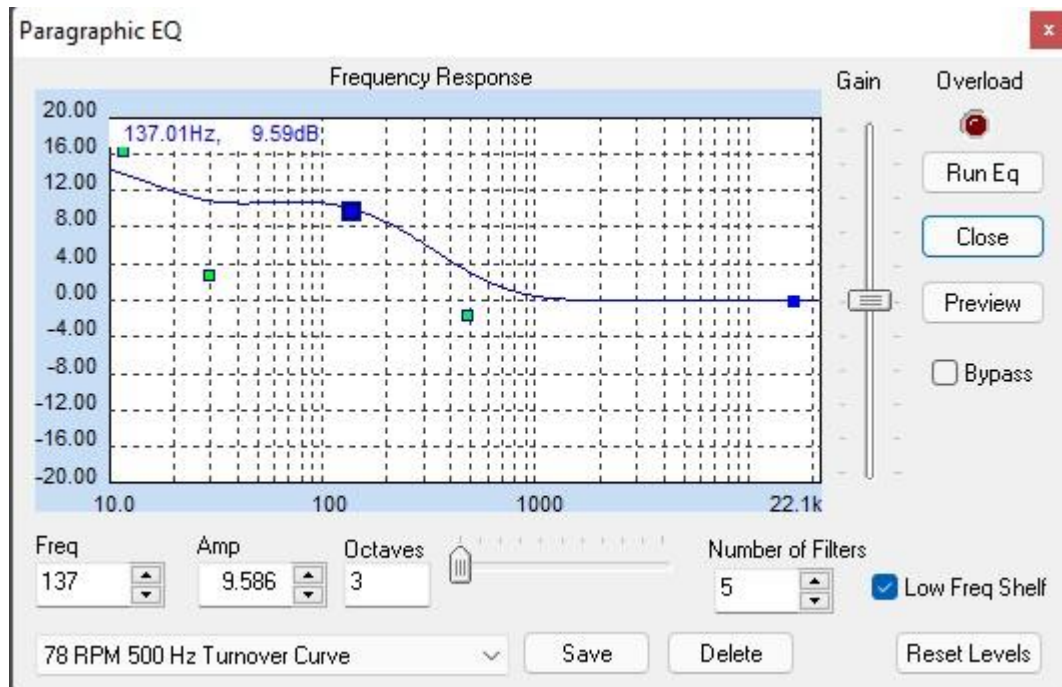


Figure 6-3 Turnover at 500 Hz with Low Freq. Shelf

Another method for adding or removing EQ is the Virtual Phono Preamp. This multipurpose software algorithm may reduce steps in the restoration by combining steps in one function. Refer to figure 6-4:

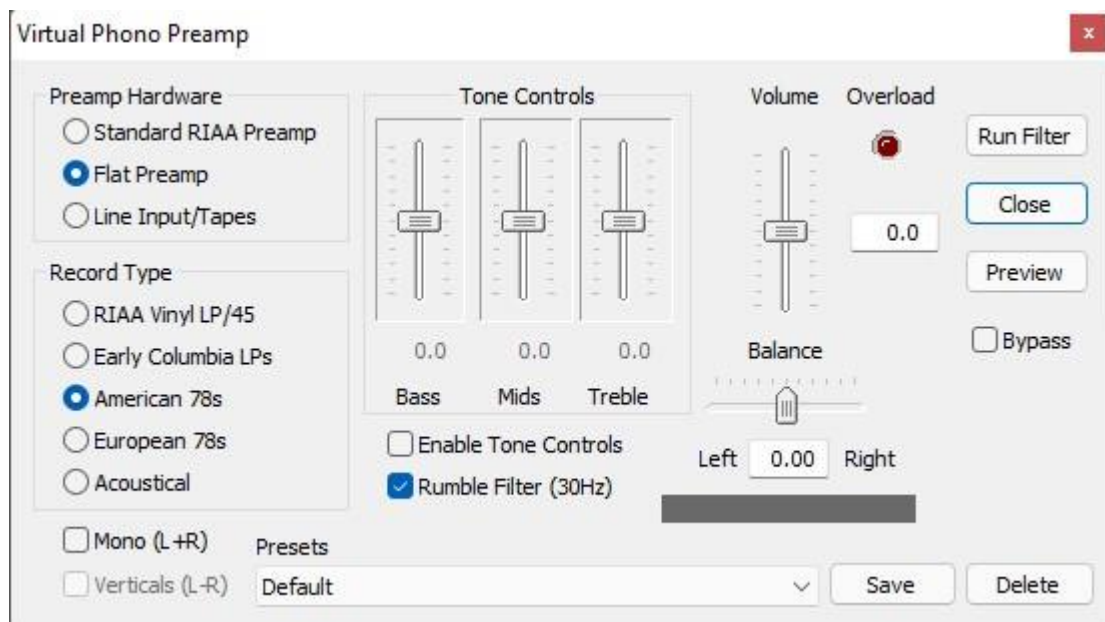


Figure 6-4 Virtual Phono Preamp 78 Record

The selected settings indicate that the original recording was from a flat preamp and the desired playback curve is for an American 78. These setting applies the same 500 Hz turnover setting as shown before with an additional filter selected (30 Hz) to remove rumble. In addition to the turnover response, this filter allows modifications to each channels volume, amplitude balance and changes to the tone (Shelf Tone Controls). Below the Left & Right setting is a meter, indicating channel balance, active when the software is in use.

The comments *American* and *European* are not to be used as a universal setting for either 78 records made in USA or Europe. Many records made before World War Two used a turnover of 250 Hz and this value is labeled European in this filter. The American setting uses a turnover of 500 Hz and was common for records made after WW2.

6.4 *How to Produce a Flat Recording from a Non-Flat Preamplifier*

When the first electric record players were introduced, the controls were very basic and consisted of a volume control and a simple turnover circuit. Later, the audio equipment became more complex and the record companies developed unique equalization curves for their records. This led to preamplifiers with numerous switches to allow you to set the different EQ values. The audio equipment companies, and the record companies finally settled on a RIAA EQ that was standard for all records and this EQ became built-in to many of the preamplifier's phonograph input circuits. Today these preamplifiers with selectable EQ values are not common and preamplifiers with just the RIAA curve are the ones likely to be available.

6.4.1 *Two Types of RIAA EQ Filters are Present, Which One to Use?*

The DCart11 software contains two software filters that can remove the RIAA curve from the preamplifier. The Paragrophic EQ Filter was the original method used to apply different types of EQ curves while the Virtual Phono Preamp was a later development. The Virtual Phono Preamp follows the various EQ curves to a higher precision than the Paragrophic EQ software option and for this reason this *filter should always be used to remove the RIAA curve from a Preamplifier*, for those who do not have a flat amplifier. The Virtual Phono Preamp uses the phrase *Acoustical* which can also mean *Flat*.

To produce a flat recording from a RIAA Preamplifier when playing a 78 Electric Record you would set the software settings to those in figure 6-5:

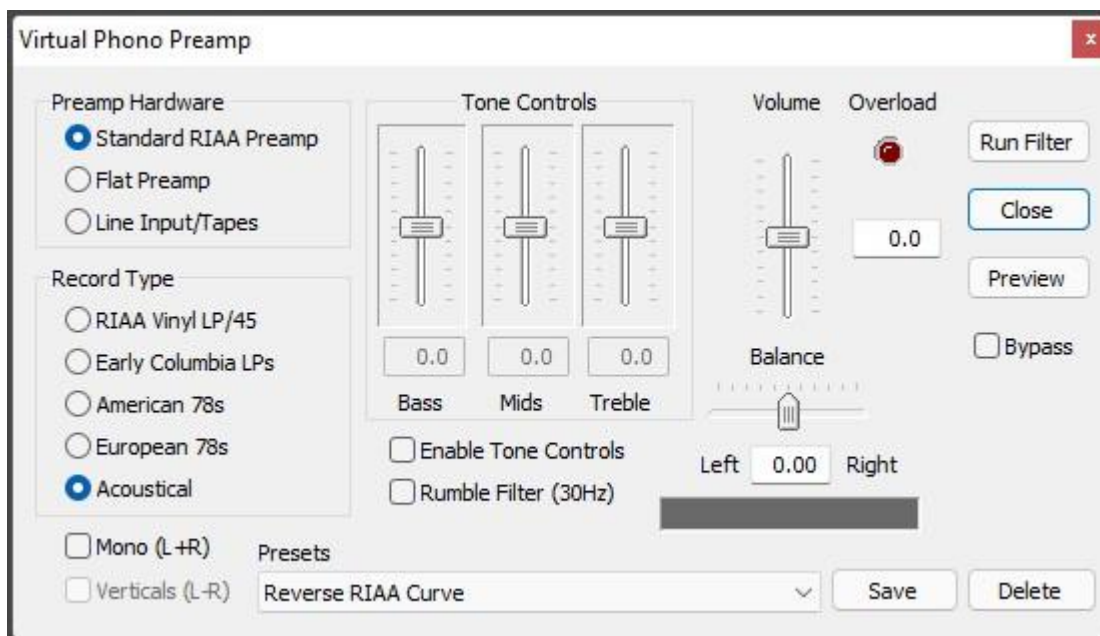


Figure 6-5 RIAA to Flat Recording

In this example the acoustical setting gives us a flat recording for the restoration work by *incorrectly* telling the software what type of record we have. The record type is really an American 78 RPM record but by using the acoustical option, the Preamp RIAA curve will be removed. The new flat recorded file is then saved for further processing as the Preset title indicates.

The Paragrophic EQ filter contains a preset to remove the RIAA curve but performance will be better by using the Virtual Phono Preamp.

6.4.2 The Types of EQ that can be Removed

The DCart11 software has many options to *apply* EQ to the music waveform but the software can only *remove* two types of Standard Equalization Curves. These two types are the RIAA EQ and the NAB (National Association of Broadcasters) Tape curves. The RIAA curve can be removed using the Virtual Phono Preamp software and the NAB Tape curve can be removed with the Paragrophic EQ software. Because the DCart11 software has this limitation on the number of Standard EQ curves it can remove, the various noise removal methods were developed to apply the EQ in specific restoration stages.

6.5 *How to Determine if the Record was Acoustical or Electrically Recorded*

Knowing what type of recording that was used for the record is necessary for music restoration. The choices are either Acoustical or Electrical recording; Acoustical recording is a mechanical operation where the sound energy moves a recording cutting head in synchronism with the music waves to store the music on a master record versus Electrical recording that captures the music with a microphone and after amplification and modification provides power to the groove cutter to place the sound into the master recording.

One method to determine the type of recording is to listen to the music as it is played back. This method uses the source recorded from a *flat* preamplifier while you toggle the virtual phono preamp on and off as you listen to the song. The Record Type Option should be selected for what you think the record type is. The use of the preview control and bypass makes it easy to apply and remove the EQ while listening to the song. For an example, if you have the *flat* recorded file as the source, bring up the phono preamp, select preview and play the music as you toggle the bypass option. Refer to the figure 6-6:

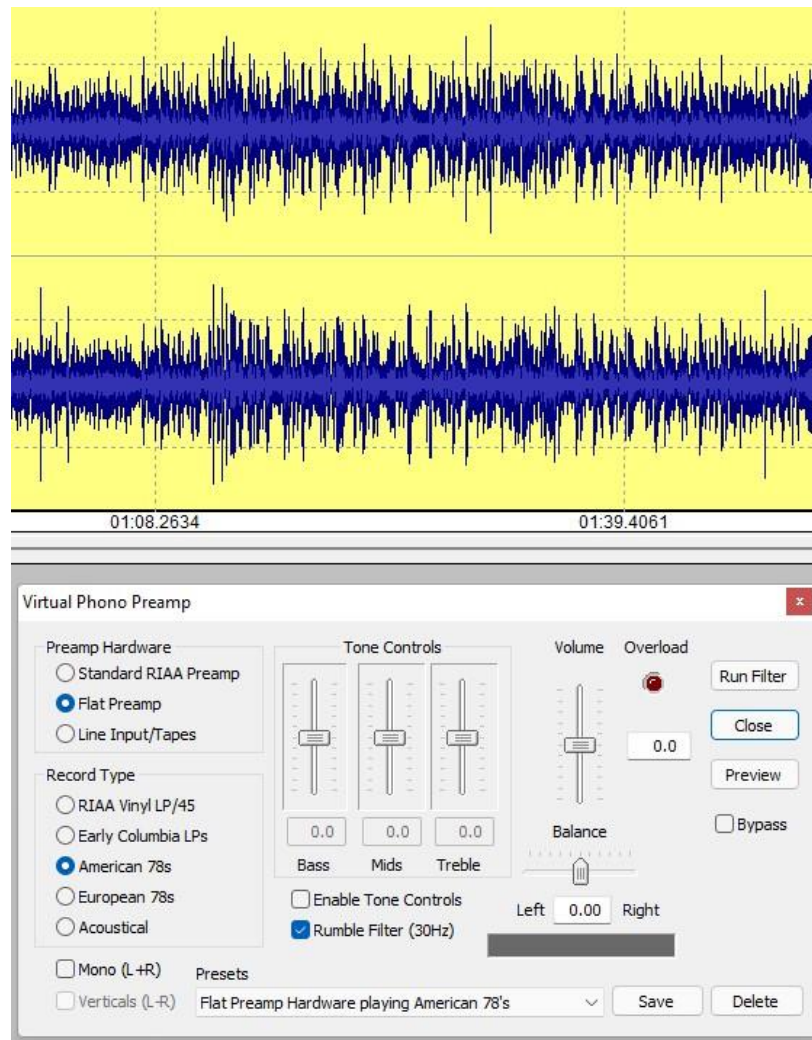


Figure 6-6 Checking the Music for Equalization

In this example turning the bypass on and off during a Preview makes a difference that is easy to hear and since the music sounds correct with the Virtual Phono Preamp on, you know that the song was electrically recorded. The opposite condition with the sound changing slightly or not at all means that the song was originally acoustically recorded.

After hearing a few songs, the difference between the Acoustic recordings and the Electrical recordings will become second nature. If you have a questionable song that could be either, there is another method that uses the software's spectrum analyzer's ability to provide a frequency profile of the music to help you decide between the two.

The use of the spectrum analyzer can show the type of recording that was used since the acoustic and electrical recording each have unique frequency responses. In the case of the Edison Diamond Disc records this method is often needed since the quality of the Edison

acoustic records was *state of the art* at the time and the records have a very nice sound for acoustical records that can, at times, sound like an electric recording.

While the use of the spectrum analyzer while playing material from a *flat* transcription will often show differences between electrical and acoustic recorded records; the difference is somewhat easier to see by using the American 78 setting on both types of records along with the use of the Spectrum Analyzer. Next is an example using some arbitrary 78 Electric and 78 Acoustic records. Both music selections were recorded with a *flat* preamplifier and then had the American 78 setting applied with the software Virtual Phone Preamp while the spectrum analyzer was running. The specific American 78 setting may not be the best for this electric recording, but the method shown here works using this type of setting.

The acoustical recorded spectrum analyzer picture (with the American 78 setting on) is shown in figure 6-7:

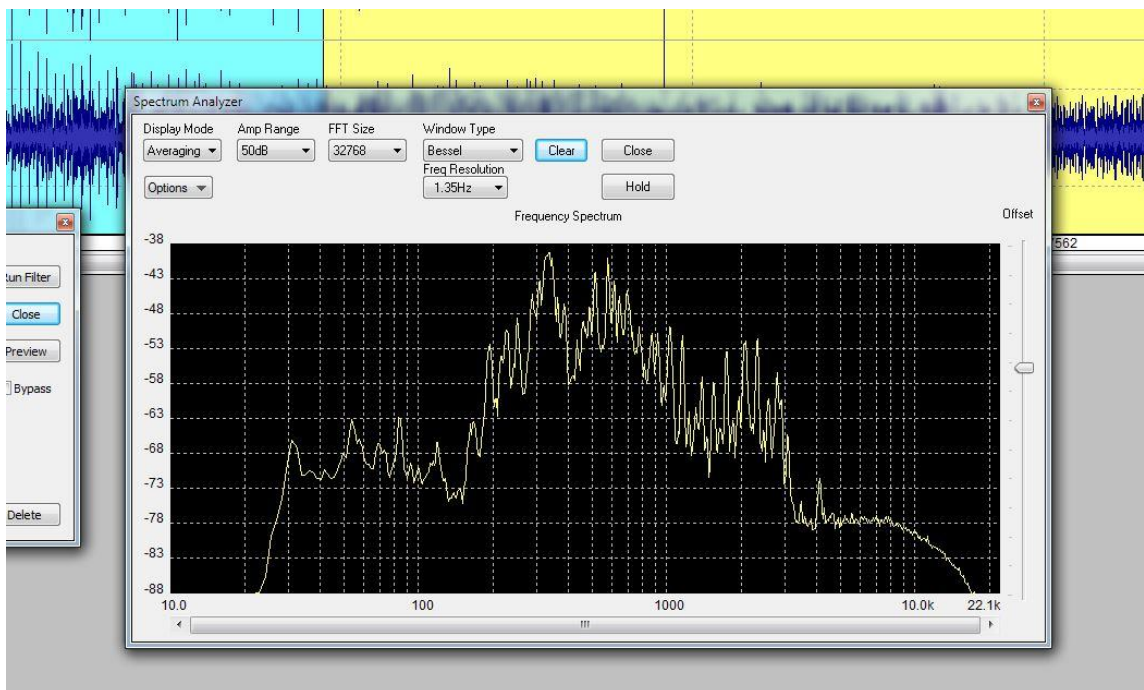


Figure 6-7 Acoustical Record with American 78 EQ applied

Next, the Electrical recorded spectrum analyzer picture is shown in figure 6-8:

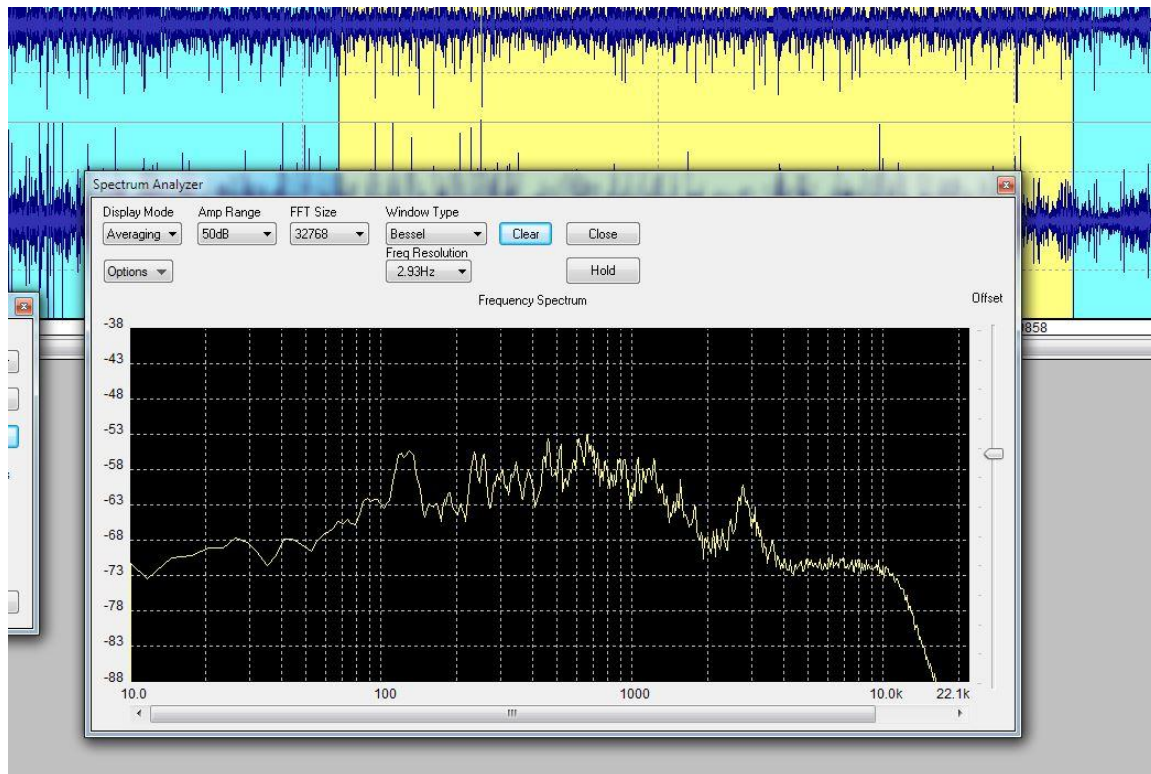


Figure 6-8 Electrical Recording with American 78 EQ

In both cases, the American 78 EQ, applied a boost to the lower frequencies. This boost only works if there is music energy present in this low frequency area to boost. In the case of the acoustical recorded music the spectrum below 200 Hz falls off to indicate that there is not any music present to boost. This lack of lower frequencies contrasts with the electrical recording in figure 6-8. Another area to look at in the frequency spectrum is the higher frequencies for a contrast between the two recording methods. Figure 6-7 shows a sharp drop-off over 3 KHz whereas figure 6-8 shows music energy present to 10 KHz.

6.6 *The Correct Equalization for a 78 Electric RPM Record*

For 78 Electric recordings, a starting value for a Turnover Value is available in the published tables in the DCart11 software help files. Refer to Table 6-1.

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

Table 6-1 Turnover Values from the Help File

The text in the help file states that this equalization can be applied using either the Virtual Phono Preamp or the Parabolic EQ options; however, the Virtual Phono Preamp will apply a more accurate result but is limited to two 78 RPM settings; American with a turnover of 500 Hz or European with a turnover of 250 Hz. The Parabolic EQ has a wider range of values to choose from.

This table provide a useful starting point but the record producers have, since the beginning of recorded sound, imposed their own biases on the sound of the music. What this means is that as you restore the music you can modify the equalization that you apply to shape the sound for what you believe to be correct. The modification that you make to the published EQ numbers will be made to the music after the *Level One* operation (details in the noise removal chapters). In this manner, you can always *return* to the original sound and make different types of modifications without having to start at the very beginning with an original recording. The only EQ curves that can be removed are the RIAA and the NAB Tape curves. You cannot remove a 78 RPM curve once you have applied it to a flat recording.

This lack of standardization by the record companies reinforces the reason why you want to start with a flat recording. When the starting point is a flat recording, the conversion to a specific type can always be *changed* later when you learn more about the recording process on a specific record since the original source file has no EQ applied.

An example will clarify the use of equalization, starting with a recording of a popular swing tune from the late 1930's. The record was made by the Victor Company, so you have some choices from Table 6-1 of a turnover value of 200 Hz or 500 Hz. The first application of EQ uses the American 78 RPM (500 Hz Turnover) setting. After the file has had the noise removed the music file is placed in the Level Two music folder using this EQ. Up to this point you have chosen to use EQ settings that have been recommended to you *independent* of knowing what was specifically done during the mastering of that record. The next step in the restoration is to use *your hearing* to modify, if needed, further changes to the records sound. If you find that a turnover value of 200 Hz sounds better than the 500 Hz value then you can return to the non-EQ file (Level 1) and easily produce the better sounding file (Level 2) by changing the EQ in the Level 2 production and then use it for your final restored music.

There is no reason that your restoration work must adhere to a strict method of using the recommended EQ settings. As your ability to restored music improves, you can always return to the starting point, Level One.

These terms Level 1 and Level 2 will be explained in more detail in the noise removal chapters.

6.7 *Regarding Pre-Emphasis on 78 Electric Records*

The original technical work performed by the Western Electric Lab in 1925 suggested that an improvement in the recording of high music frequencies was possible by boosting their amplitudes above a certain frequency during the recording process and in turn followed

by an inverse operation during playback. Additional information can be found in the reference section.

While this technique was known to the early music recording companies, widespread uniform implementation would have to wait till the adoption of the RIAA curves. However, some of the record companies did implement this method and the literature in audio books and technical articles confirms that it was possible for a record to have its high frequencies increased from the original recording.

The location for the restoration work of modifying the upper frequencies for 78 Electric Records should be performed in the enhancement chapters and therefore the EQ for 78 RPM electric recorded music should stay as only a *turnover* curve for the music up to Level 2 without adding a *roll-off* value.

6.8 *Special EQ for Edison Electric Recordings*

The history of the Edison Phonograph Company showcases an organization that was independent from the rest of the music industry in all matters. Edison used a disc groove that moved up and down or *hill and dale* for the music. The Victor company and many others used a back and forth or *lateral* variations for the music. While Edison's original patent for the phonograph claimed the use of both lateral cut and vertical motion to capture the sound, he believed that the vertical method gave a better acoustical performance than did the lateral method. Edison's many patents helped to create the invention of the vacuum triode (Edison Effect) and the microphone, yet he did not adopt the new electric recording method first used in 1925 until much later than his competitors in 1927. Very little written information is available regarding the methods that were used by the Edison Company for the electrical recordings. From the discussion on the need for equalization recall that the stylus motion for the Edison Diamond Disc record is a vertical motion and while the amplitude does increase with low frequencies, the potential *swing* into an adjacent groove does not have the same limitation as with a horizontal motion.

The Edison Electric recording process improved the frequency response of the record assuming that most playback would be via acoustic or mechanical means. The result is that some of the low frequencies would not be reproduced by the record player so that the low end could contain significant rumble and noise that would be present in the record but *not heard* using an Edison Record Player. The use of a standard 78 RPM turnover curve with the Edison Diamond Disc Electric records will often yield a noisy and un-natural sound for the low frequencies. The low frequencies can be boosted and will improve the overall sound, but this

method requires some extra steps to just an application of EQ. Specific information will be shown in the enhancement chapter for Edison Electric Diamond Discs. For the Diamond Disc records that were electrically recorded the best result is to not use any specific bass boost or turnover value during the Level Two or Basic restoration work.

For the Edison Needle Type lateral records, the table in the Diamond Cut help file (Table 6-1) shows a turnover value of 500 Hz. These records used a lateral motion for recording sound and were produced for a very limited time.

6.9 *EQ for LP and 45 Records*

In a previous section (6.4.1) the virtual phono preamp was used to *remove* any EQ from a preamplifier as it implemented the RIAA curve better than the Paragraphic Filter. When LP or 45 RPM records require that the EQ is *applied*, either filter can be used. The reason is that these records can lose some high frequencies due to groove wear and the Paragraphic Filter may provide an improved sound because of a slight difference in high frequency implementation.

The chapter on the details of the LP and 45 RPM record noise removal will use RIAA EQ from *either* the Paragraphic or the Virtual Phono Preamplifier software depending on the condition of the record.

7.0 *Transcribing Records*

The process of taking the recorded music and converting it to a digital file is known as transcribing a record and starts with converting a recording to a raw music file. Next, any changes to the raw recording to correct the speed, balance the channel gains, and make it a flat recording, are done. Lastly, trim away any extra recording during the beginning or end of the music to create an Original File.

7.1 *Definition of Terms*

1. Raw Recording. The phrase *raw* refers to the first conversion of the analog signal to a digital value. No additional signal modifications have been performed on the resulting digital file. The recording will contain the result of the conversion from start to finish.
2. Flat Recording. Any EQ applied to the Raw Recording has been removed.
3. Balanced Channels. The phrase *balanced* means that the individual gains of the Left and the Right channels are adjusted to produce equal recorded Left and Right Channel Amplitudes.
4. Trimmed Recording. The phrase *trimmed* means that the sections of the digital conversion from the analog recording that occurred when the stylus was *not* in the record groove have been removed.
5. Original Recording. The phrase *original* means that the recording has had the results of terms 1 through 4 applied to the digital file.

7.2 *Equipment Set-up*

The tools chapter described the audio equipment needed for the recording and playback process. Refer to figures 4-1, 4-2, 4-3,4-4, and 4-5 for a possible setup for the transcribing process. Some of the material from chapter 4 will now be repeated with added detail. Check that your privacy setting to verify can use your microphone to be used.

7.3 *Preamplifier and Turntable Speed*

In the tools chapter a flat preamplifier with adjustable channel gain control was recommended for transcriptions. If you do not have a flat preamplifier i.e., the one you have was designed to play LP records with the RIAA curve, then you can still use this type of preamplifier, but you will need to apply an extra step to the raw recording. This extra step will occur after the raw music file is made but before the file becomes the original recording. If you do not have individual channel gain adjustments in hardware, then a software feature within DCart can be used to change the gain for each channel.

DCart can compensate for speed differences between the correct record speed and the turntable speed. For example, a 78 RPM (revolutions per minute) record, is not compatible with turntables that only have a position for $33\frac{1}{3}$ RPM or 45 RPM. In this case, the record will be played at 45 RPM and using software, converted to sound as if it was played back at 78 RPM. While there maybe times that a record benefits from being played back at a slower than designed, the best method is to playback the record at the speed it was designed for. If the record is played back at a slower speed than the original, music frequencies will be changed in proportion to the speed change. For example, a 78 RPM record may have a strong bass note at 50 Hz. When that same note on the record was played back at $33\frac{1}{3}$ it would be at about 21 Hz! Some of these new low frequencies could be below the frequency that your preamplifier can respond to and will be lost for your transcription. The best approach is to playback the records at the same speed that they were designed for.

The one exception to this rule is for the Edison Diamond Disc records which were recorded at 80 RPM. The conversion from 78 RPM to 80 RPM is very small and is needed since 80 RPM turntables are not common.

7.4 *Cartridge and Stylus*

The selection of the cartridge and stylus for the playback of the records is very important. The music information is contained in the *motion* of the grooves on the record and the job of the stylus and cartridge is to convert this *motion* into an electrical signal. Therefore, you want the stylus size to correctly *fit* the specific record groove. The reference section provides addition information regarding the choice of Cartridge and Stylus for your transcription.

The Cartridge should be a Stereo type that fits your turntable and can accept different types of styli. Many people have used a Stanton 500.V3 cartridge since the price was moderate and different types of styli were available. This brand is not currently being produced although old stock can be found. Some choices available today are Ortofon, Grado, Nagaoka, and others. When the cartridge is selected the preamplifier should apply the recommended electrical load.

The stylus size is critical to the quality of the playback of the sound. If the tip rides low in the groove for a lateral record it will scrape the groove bottom and add significant noise to the music. On the other extreme it can ride too high and miss some of the recorded sound in the groove. Styli are described using a dimension in either mils (thousandth of an inch) or μM (millionth of a Meter) for the radius of the tip for a conical shape; and two numbers for a complicated shape. When two numbers are used one is for the larger radius and the other is for the smaller radius (an elliptical stylus has a major and minor radius).

For Edison Diamond Disc records a stylus must ride in the bottom of the groove. Most Diamond Disc records can use a “DJ” type of stylus that has a conical shape and a radius of 0.7 mils. For very worn records, a 3.75 mils conical shape stylus from the Expert Stylus Company (located in England) works well. The Reloop OM Black 0.7 mil DJ stylus from Ortofon is used for Diamond Disc records in my Audio Studio.

For the 78 Acoustic and Electrical records, a 2.7 mils stylus (Stanton D5127) has been used in my Audio Studio. For today’s LPs and 45s the standard size is 0.7 mils x 0.4 mils and is a good choice. It is possible to spend quite a bit of time and money to obtain a large set of different styli for the best transcription result. The amount of wear on the groove of the record can often be avoided by having the stylus *ride* the groove in a new location that is removed from the distortion and damaged section of the groove. Trial and Error in stylus size can find this new location.

7.5 *Channel Selection & Gain*

All types of records will be transcribed using a Stereo Cartridge with Left and Right Channels. Preamplifier channel gain is individually adjusted for equal recorded channel amplitudes. Signal processing will be performed in software to the original left and right channels during the noise removal work to produce the desired outcome and if needed additional channel balance.

A stereo recording for the transcription allows, if needed, the later separation of the vertical or horizontal components of the stylus motion via software routines. The reason that

this separation can be done is that when the stereo recording method was originally developed the designers chose a technique that would be compatible with previous recordings that had One Channel, Monaural, of music. Since the previous monaural records used a horizontal groove motion the solution to add a second channel for Stereo, was to change the stylus horizontal motion to one using two 45-degree vectors with the left channel on one side and the right channel on the other. This 45-degree motion allowed a new vertical component to be added to the lateral component resulting in two channels of music, left and right, in the record groove. The original horizontal (monaural) motion would come out equally on the left and right channels which allowed the older records to be heard. The older hill and dale technology that Edison used was merged with the lateral motion to create Stereo recording.

Since the cartridge coils respond to 45-degree vectors, by performing math operations you can separate the Vertical or Lateral Motion. The fact that the Vertical or Horizontal groove motion can be *pulled* out of the stylus motion means that all types of records can be played back using a stereo cartridge for the original transcription.

7.6 *Sample Rate and Bit Depth Selection*

A sample rate of 96 kHz (96 thousand samples per second) is recommended for all transcriptions. This high sample rate is useful for restoration work since the software algorithms for noise removal will benefit in two ways; first the higher sample rate allows very high (ultrasonic) energy to be recorded which will aid in finding impulse noise (which is largely high frequency) and second, the large number of samples per second provides many pieces of information to the algorithm that *restores* the missing music section during noise removal operations. Since the noise filters have many samples per second available to them this allows a *more aggressive* setting for these filters with a resulting cleaner sounding music verses a lower setting of 44.1 kHz samples per second.

A sample rate of 44.1 kHz (44.1 thousand samples per second) can be successfully used if the equipment only supports this sampling rate. The noise removal can be more difficult and will require more manual noise removal. While this rate has the advantage that the size of the music files will be less than the 96 kHz, the cost of digital storage is low and should not be a factor in selecting this sampling rate.

The DCart11 software allows a sample rate conversion via a re-sampling algorithm, however, it is NOT recommended to use this conversion to a higher rate than the original recording was transcribed at to achieve better noise removal. If the original rate is 44.1 kHz, then maintain this rate for your restoration work.

The word length of the recording is set to 16 bits of digital depth. This value is more than adequate to maintain a high signal to noise ratio needed for digital processing. Refer to the reference section for additional information that justifies the use of 96 kHz sample rate and a digital word length of 16 bits. Some analog to digital converters provides only one choice of 24 bits for the digital word. This value can be used but the best method is to select 16 bits by using the options in the converter driver or the control panel in Microsoft operating system.

In Chapter Four the control of the sampling rate and bit length was described as needing both the software drivers for the converter and the DCart11 software to communicate with each other. This communication will not work at times with some external converters and internal converters that use Windows drivers. An additional complication is that Microsoft audio drivers will re-sample digital files to different rates at times to accommodate user needs without the user aware of the re-sampling. *It is very important that the desired sample and bit rate is performed to produce the music file.* The next method will ensure that it happens for the music restoration work.

7.6.1 Setting the Converters Sample Rate and Bit Depth for Recording

ASIO Driver

This example uses an external USB converter made by Focusrite (Scarlett 2i2 Gen3) that communicates via an ASIO driver. The Focusrite companies' products are used in many DAWS (Digital Audio Work Station) that communicate using the ASIO (Audio Stream Input/Output) driver. This example uses the ASIO driver selected in the preference window (Figure 4-7).

The DCart11 Record Function is selected and the desired Sample Rate and Bit Depth variables are entered. Refer to figure 7-1.

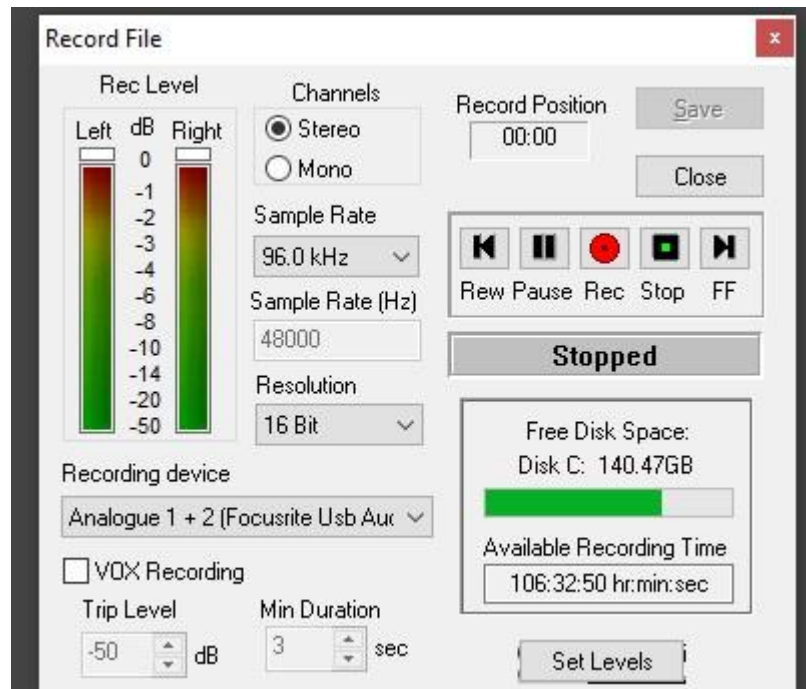


Figure 7-1 Focusrite (ASIO) device and Settings Selected.

Refer to Figure 7-2 for the Windows response to setting the Focusrite to 96 kHz and 16 Bit Depth via the DCart11 Record window.

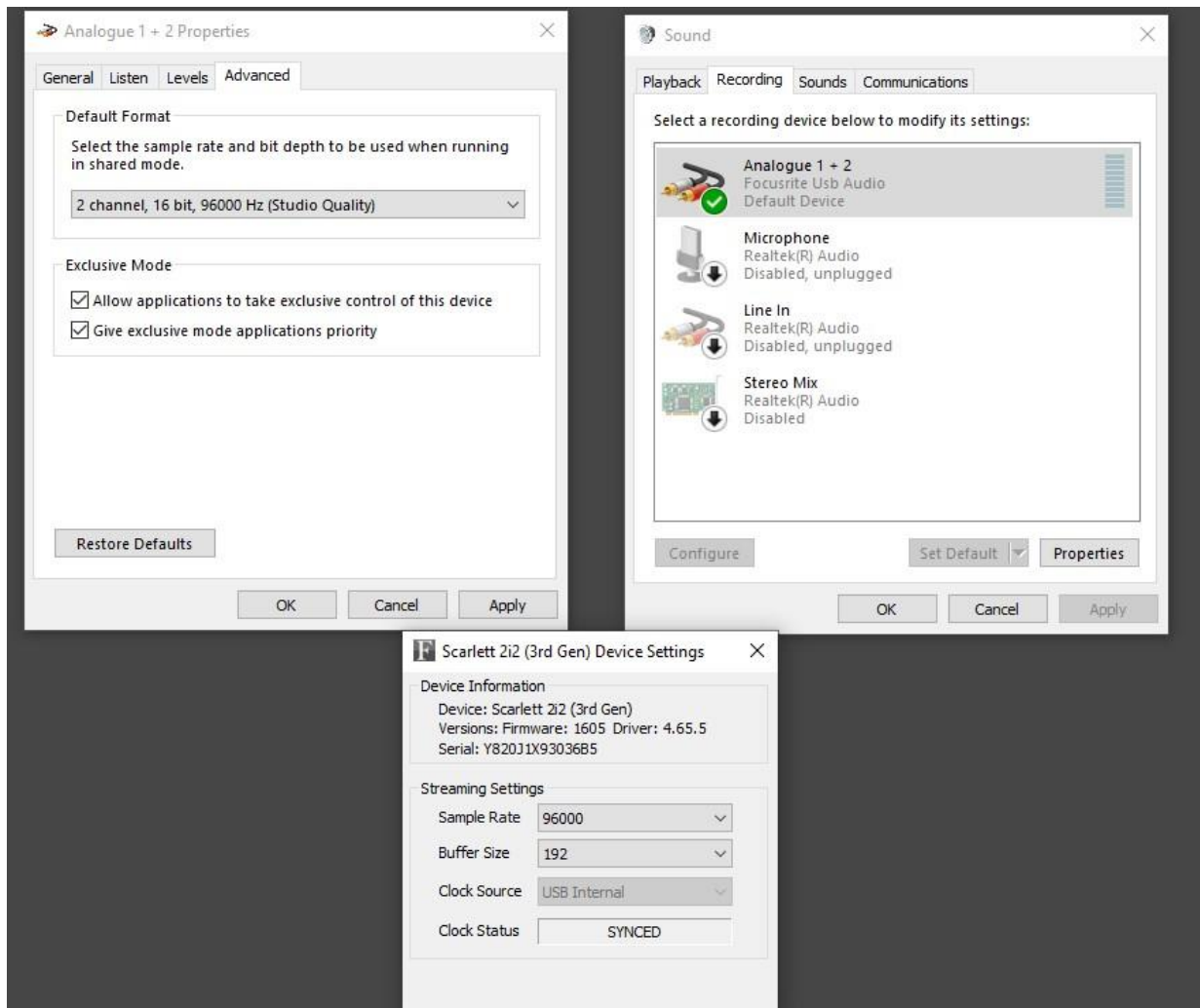


Figure 7-2 DCart11 Preferences sent to Focusrite Converter

The Focusrite company provides additional information regarding the operation of their device with the Windows operating system on their web site. In this case, the Windows advanced tab shows *what* the Focusrite driver *set* the sample rate to. The small window in the bottom of 7-2 is where the actual sample rate is set. The Windows software will not *set* the sample rate.

Windows Software Control

This next example is when the A to D converter is located on the computers main board. A Realtek Chip Set is used for the conversions and will respond to the Windows software via

the Windows Sound Panel settings. Refer to figure 7-3 for the results of setting the converters' values.

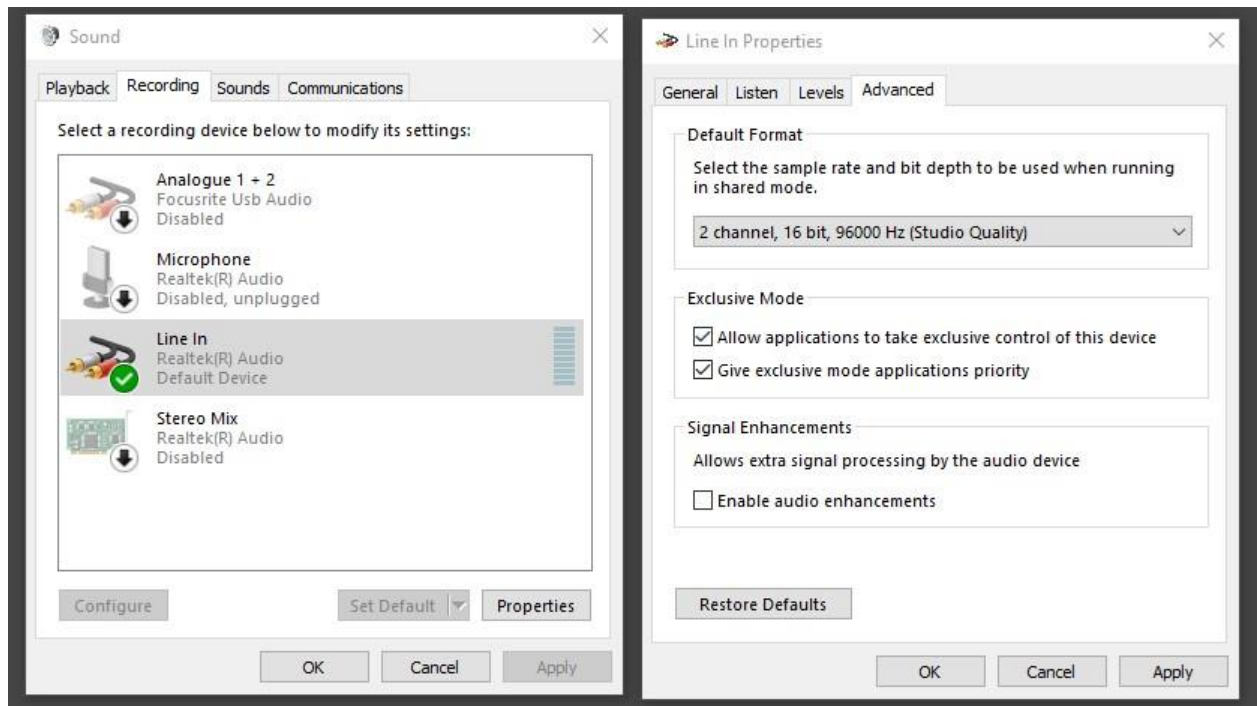


Figure 7-3 Realtek Settings to 96 kHz and 16 Bits

In this example the A to D and D to A Converters Sample Rate and Bit Length has been correctly set for the transcription by the Windows Sound Panel. The preferences tab under Edit should have the correct converter shown (Figure 4-8). Sometimes the preference for the sound card will show a phrase *primary sound driver* or *primary sound capture driver*. It is best to use the arrow besides the software option to select the *actual device name*.

Now that the converters' values are correctly set and the DCart11 software has the device connected, the DCart11 Record Window can now be set to the *same* values that were previously used for the converter.

If the sample rate must be changed, the controls in the DCart11 Record Window may not change the actual sample rate. The header file written for the new wav file *will* contain the selected rate from the Record Window, but the actual rate *may not* be written.

7.6.2 Checking the Recording Window

Figure 7-5 shows the recording window (press the red button in the toolbar to bring it up) set for 96 kHz and 16 Bits using the Focusrite Converter, while 7-6 shows the window with the Realtek Chipset.

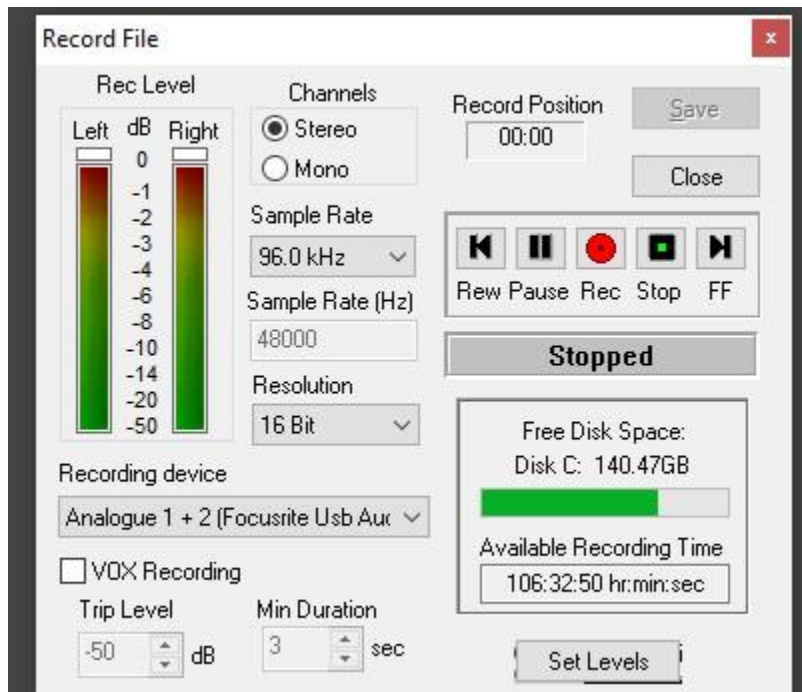


Figure 7-5 Recording Window with Focusrite Converter

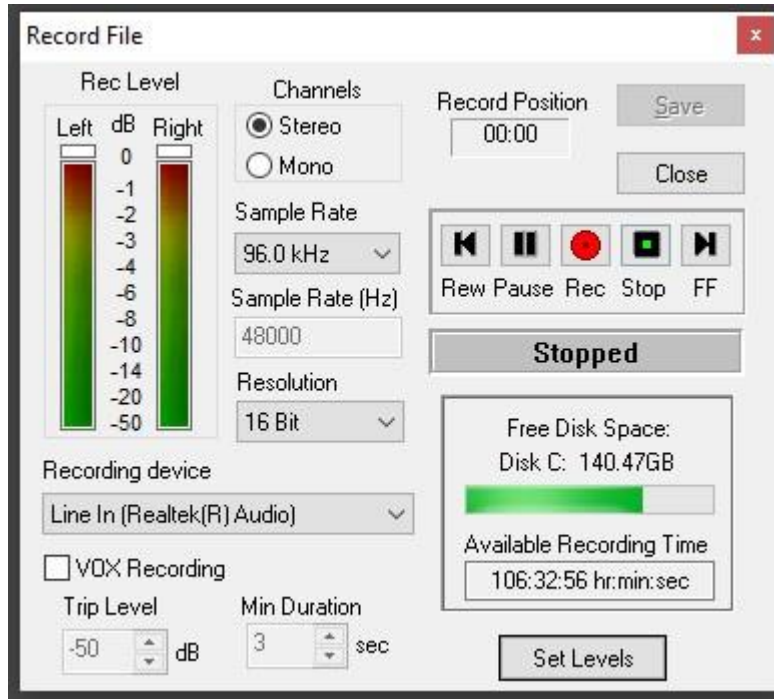


Figure 7-6 Recording Window with Realtek Chipset

The DCart11 software displays the recording variables that will be written to the header file when it is produced. At this point, you can *only* change the converters settings by using the recording window settings if you have an ASIO device and driver selected. For the Windows Driver, the converters values are first changed via Windows Operating Software (Windows Sound Panel or Settings). Then a change is made via the Record File window so that the files header is correctly written.

In conclusion, the way to record the sound at the desired sample rate and bit resolution for the Windows Software Driver is to proceed in this order:

1. Set the A to D and D to A Converter to 96 kHz and 16 bits by the manufacture's device in the Control Panel or via the Settings panel.
2. Adjust, if needed, the values in the Recording Window to agree with 96 kHz sample rate and depth of 16 bits.

For an ASIO Device, set the desired values in the Record Window while Stopped and press Pause. Then check the ASIO devices Icon, if provided, to verify the settings.

7.7 Playback System

7.7.1 Hearing the Recording

During the transcription it is important to hear the music, as it is being recorded, so that you will hear if any distortion is occurring from a wrong gain or speed setting and when the music starts and stops. The level of playback should be low so that any possible audio feedback does not occur between the recording and playback. This option, to hear the recorded music, can be set via the software settings in the recording properties or by using a preamplifier with multiple outputs so that one output can go to the A to D converter and the other to the playback amplifier. Refer to figure 7-7 for the *listen while recording* settings using the Focusrite converter. The same method works for the Realtek Chip Set or other converters.

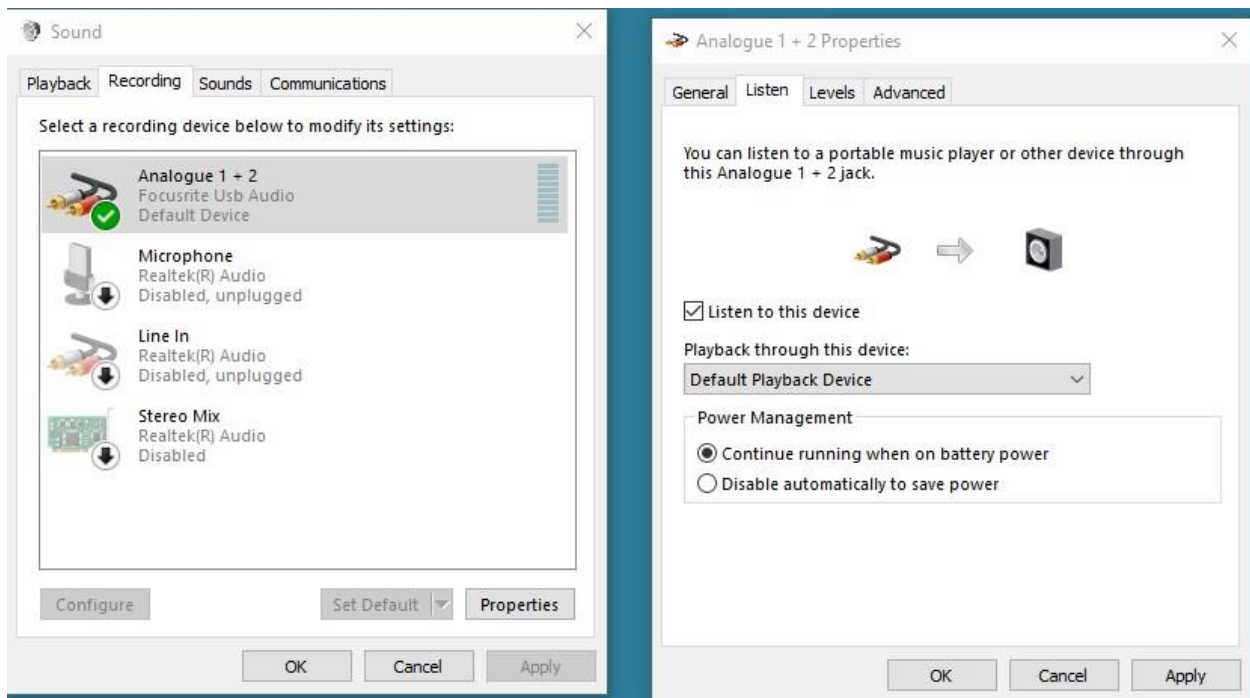


Figure 7-7 Listen while Recording

7.7.2 Playback Sampling Rate

The same communication problems using the Windows Driver can occur during playback with the result of having the wrong sampling rate used during playback. If the

Windows Driver is used, the value set in the Windows Sound Panel or Settings will be used *independent of the actual music file header*. This means that when you playback a file using a Windows driver, the desired value will be set by you manually in the same location as you set the recording sample rate/bit depth.

No manual setting is needed with the ASIO driver. For an ASIO device, the sampling rate of the D to A converter will be set by using the header file present on the wav audio file. As you change from one file to another during playback with an ASIO driver, the converter will automatically change the conversion rate as it reads the header file.

7.8 Details of the Transcription Process

7.8.1 Clean the Record Surface

The surface of the record should be cleaned of dust and dirt. Each type of record has a specific cleaner that can do an excellent job *and* another cleaner that can destroy the record! For example, water will clean a vinyl LP record but will damage an Edison Diamond Disc record. The reason that water will damage the Edison records is that this type of record was made using a wood powder core material sandwiched between plastic surfaces. The wood core will absorb water and change the shape of the record. Alcohol will clean an Edison Diamond Disc record's surface but will damage a 78 RPM record due to alcohols' ability to dissolve the shellac present in the record used to bind the rock material.

7.8.2 Set the Speed to the Value for the Record

Except for Edison Diamond Disc records (set to 78 RPM not 80 RPM), the turntable should rotate at the correct speed for the record. If the turntable cannot rotate at the correct speed, set the speed as close as possible and refer to the later section in this chapter on speed correction.

The DCart11 Help Files provides a path to a printable Strobe Disc that you can use to check the speed of your turntable by using fluorescent lights for illumination.

7.8.3 Prepare the Software

Set the sample rate and bit resolution using section 7.6. With the Record File Window selected set the sample rate set to 96 kHz, resolution to 16 bits, and the recording software to the pause state just before you are going to record the music. Refer to Figure 7-8:

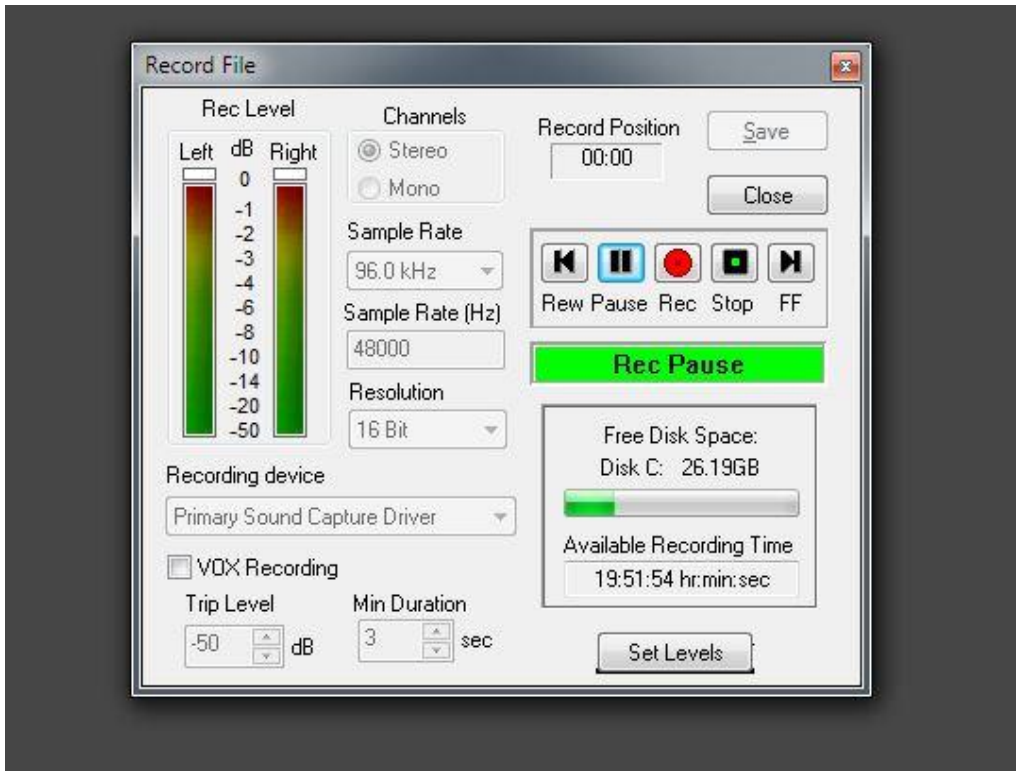


Figure 7-8 Recording Window Paused

7.8.4 Recording of Raw File

One method for recording the song would be to start the recording just after the Stylus drops into the music groove a couple of seconds before the music starts. The ability to press the recording button at this exact moment is hard to do and is not needed. Instead, after the recording has been made the sound of the Stylus dropping and entering the groove and groove noise before the start of the music can be later removed in software. Therefore, the Rec button can be pressed before the stylus has been placed onto the record. The method used for a single song is to:

1. Press the Pause Button in the recording window.
2. Start the turntable rotating.
3. Press the Record button in the recording window.
4. Place the tonearm either at the beginning of the record or prior to the start of the music you want to record.
5. When the music, you want to record, has finished, press the Stop Button in the recording window.
6. Press the Save Button in the recording window and place the file in a temporary location as a Raw File.

For an LP or other record that has multiple songs on one side of the record you can start and stop the recording process for each song, or the *entire* side can be recorded as one big file and then split apart using the *chop file into pieces* option under the CD-Prep Tab. To record the whole side containing multiple songs treat the recording as one big file using the same steps as previously shown. The individual songs will be separated from this big file in a later section.

While you are recording carefully watch the Rec Level meters. The meters will indicate if the signal exceeds the maximum range of the analog to digital conversion. Since the music has both loud and quiet passages the preamplifiers gain must be set to handle the total range of the music and should not be changed during the recording process. You can tell if you have set the levels correctly by having the recorded music stay in the green for the entire song. This is not always easy to do as noise events, which are not music, can often cause the level to hit the red or saturate the conversion process. A good recording will have the quiet passages at least around -10dB to -20dB down, and the loudest passages down around -3dB. The music will have noise spikes present and these spikes are allowed to hit the maximum level, while the music cannot. To state this another way, the music should range from about -3 to -20 dB while any noise event can hit the red level. The -3dB upper level provides head room for the later EQ curve and file conversions. Later figures will help to understand this relationship between the music level and noise level.

While you are recording, you also watch the Rec Level meters to observe correct balance between the Left and Right Channels. This process can be difficult when noise events occur as they are often specific to a channel. The recording balance can again be checked and corrected after the raw recording and before the original file is saved.

If the recording does *overload* the A to D conversion, lower the gain control for the preamplifier and repeat the recording process. Refer to figure7- 9 for a typical display for the

level meters while recording and figure 7-10 when you stop the recording after the music has stopped.

When the Raw file has been recorded, this file will undergo several steps to create the final Original Recording. These additional steps will be shown in section 7.9.



Figure 7-9 Recording

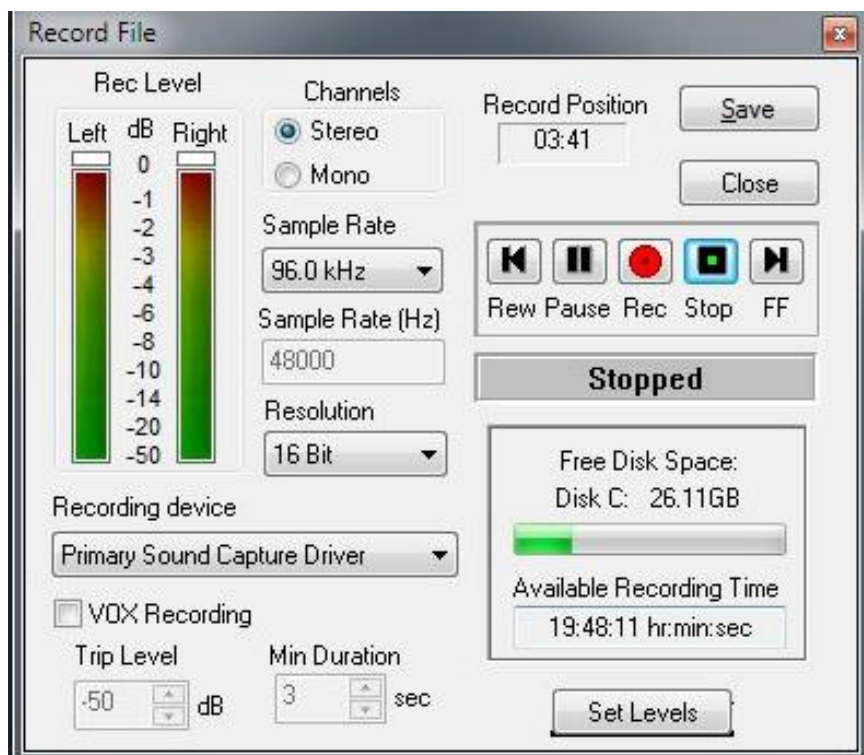


Figure 7-10 Stopped

7.8.5 Creating a Single Raw File

If you are ready to store this recording press the Save button. You now have a *Raw file*.

7.8.6 Creating Raw Files from a Multiple Song Recording

If you have recorded a large file that contains several songs within one continuous recording, as would happen from a LP record, then you can split this large file into individual raw recordings by using two DCart11 Software tools in the following sequence and shown in figures 7-11, 7-12, and 7-13.

1. Save the large file with multiple songs as a source file.
2. Under the CD-Prep Tab use the *Find and Mark Silence Passages* feature on the source file to create *Markers* that occur during the silence between songs. The marker location for silence can be manually set if desired.

3. Playback the recording to verify and adjust if needed the marker locations so that the marker is in the *dead space* between songs and not just during the fade-down of the music.
4. Use the *chop file into piece* feature under the CD-Prep Tab to create individual original recording from the previous large file.
5. Change the file names for the newly created music files to correspond to the correct names and save them as raw files.

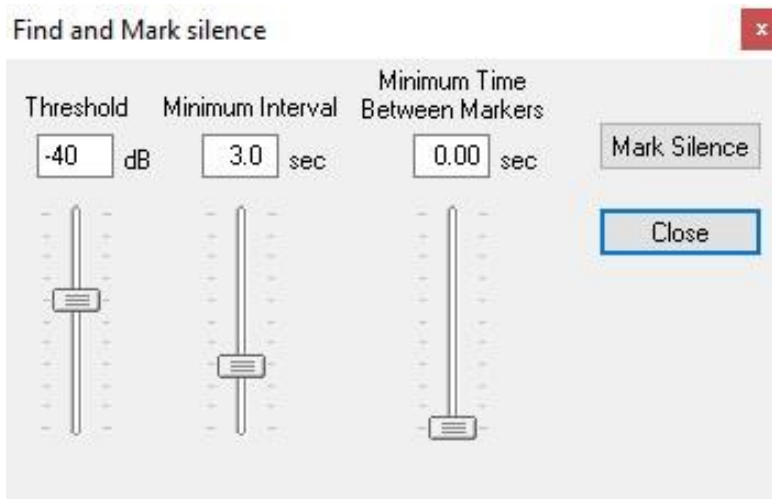


Figure 7-11 Find and Mark Silence

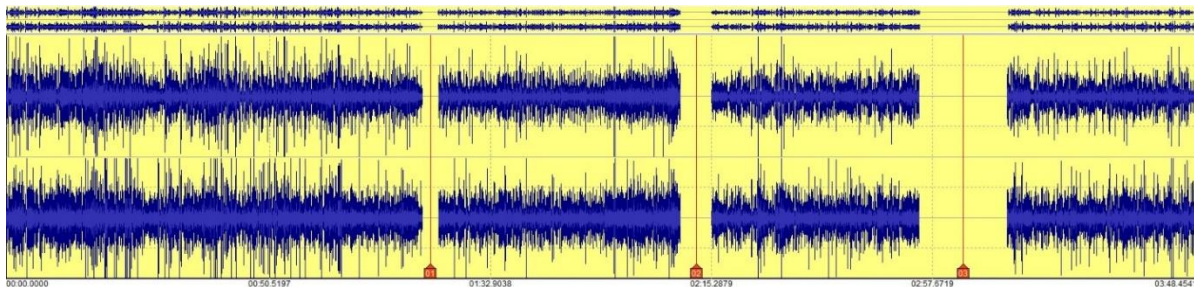


Figure 7-12 Markers at Silence Locations

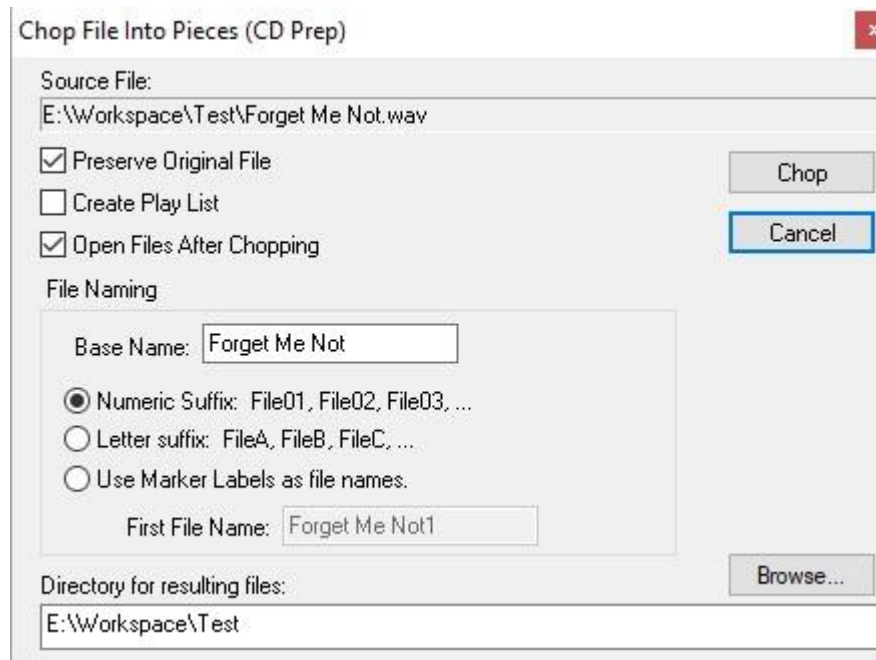


Figure 7-13 Chop File into Pieces Command

The original use for this algorithm was to prepare a CD with individual songs from a continuous long music file with multiple songs. That is why the *Create Play List* option is shown. Splitting an originally recorded long file into separate files for restoration using this software demonstrates how flexible DCart11 Software is.

7.9 Steps to Create the Original File from the Raw File

Before you can save the file, you have just recorded as an *Original File*, some important steps may be needed.

7.9.1 Checking and Correcting the Channel Balance

The channel balance, which refers to having equal amplitudes for the left and the right channel, is difficult to perform when you are watching the music amplitude via the recording meters. A better balance can be obtained by using a software tool in the Virtual Phono Preamp. An example will demonstrate the method. Refer to Figure 7-14.

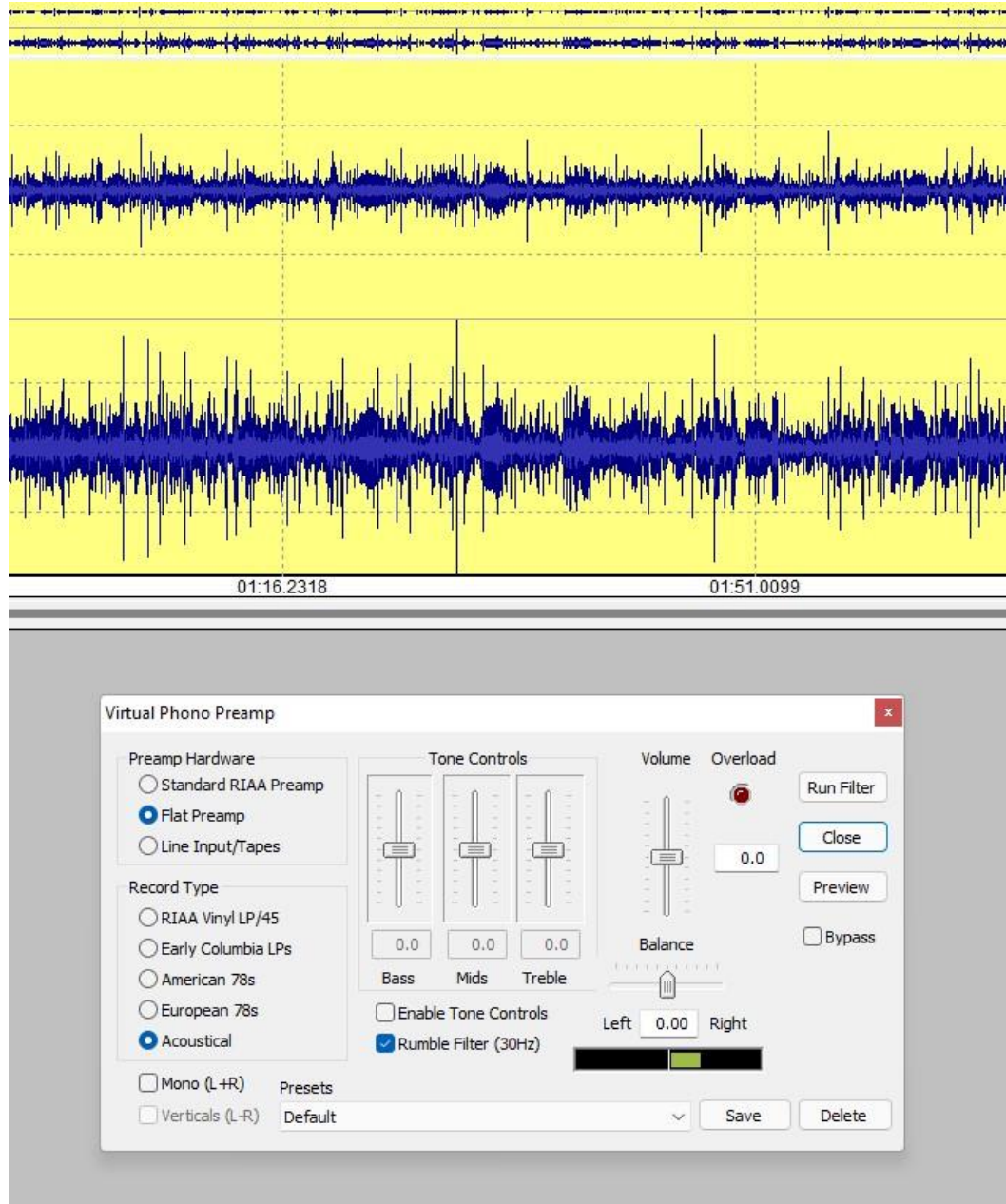


Figure 7-14 Left and Right Channel Out of Balance

In figure 7-14, the Top or Left Channel is smaller in amplitude than the Right channel, with the meter displaying the relative amount of un-balance, as it is played using the Preview Button. Since the L & R channels maybe combined for a Monophonic Recording or used as a Stereo recording, the balance between the two is important to remove noise or to have a correct stereo sound. The Virtual Phono Preamp should have the flat preamp and acoustical settings selected when the balance is measured and adjusted. When the preview button is

activated, the result on the meter under the Balance setting shows the amount of imbalance present as the music is played.

By moving the Balance slider above the meter, you can move the value on the meter to a neutral setting. At this point, you have a *raw* recording so there will be noise spikes at times. Also, due to groove wear in the record, you may see the meter movement move from the neutral position at times. Set the balance slider for the best overall setting while using the preview option during the song. Refer to figure 7-15 for a balance value selected for the waveform in figure 7-14.

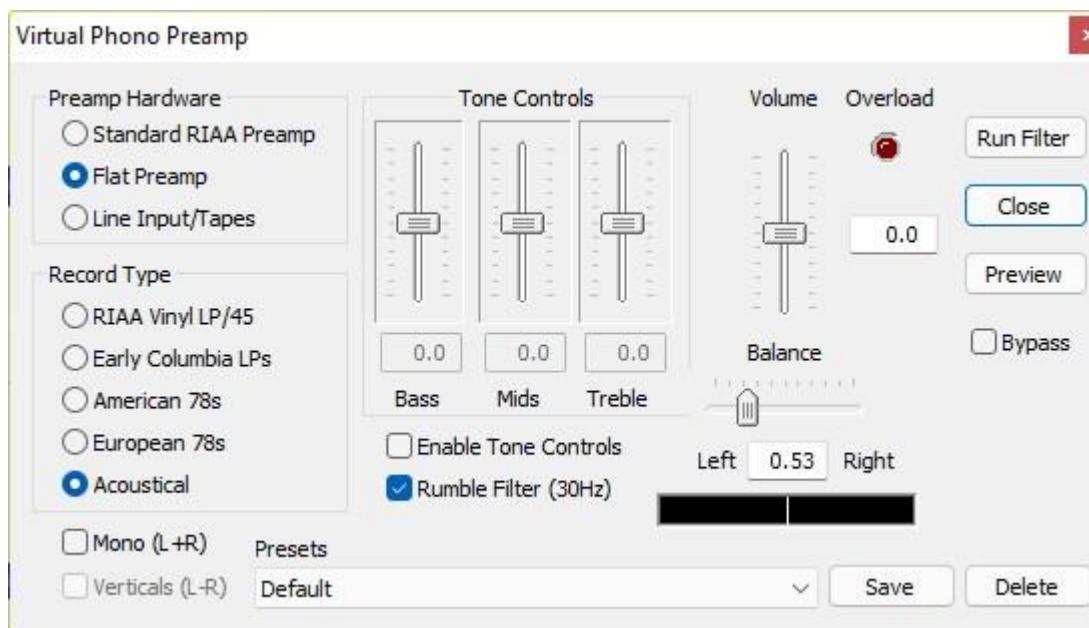


Figure 7-15 Balance Setting to be Used.

After using this setting, the Filter is run on the source with the result shown in figure 7-16.

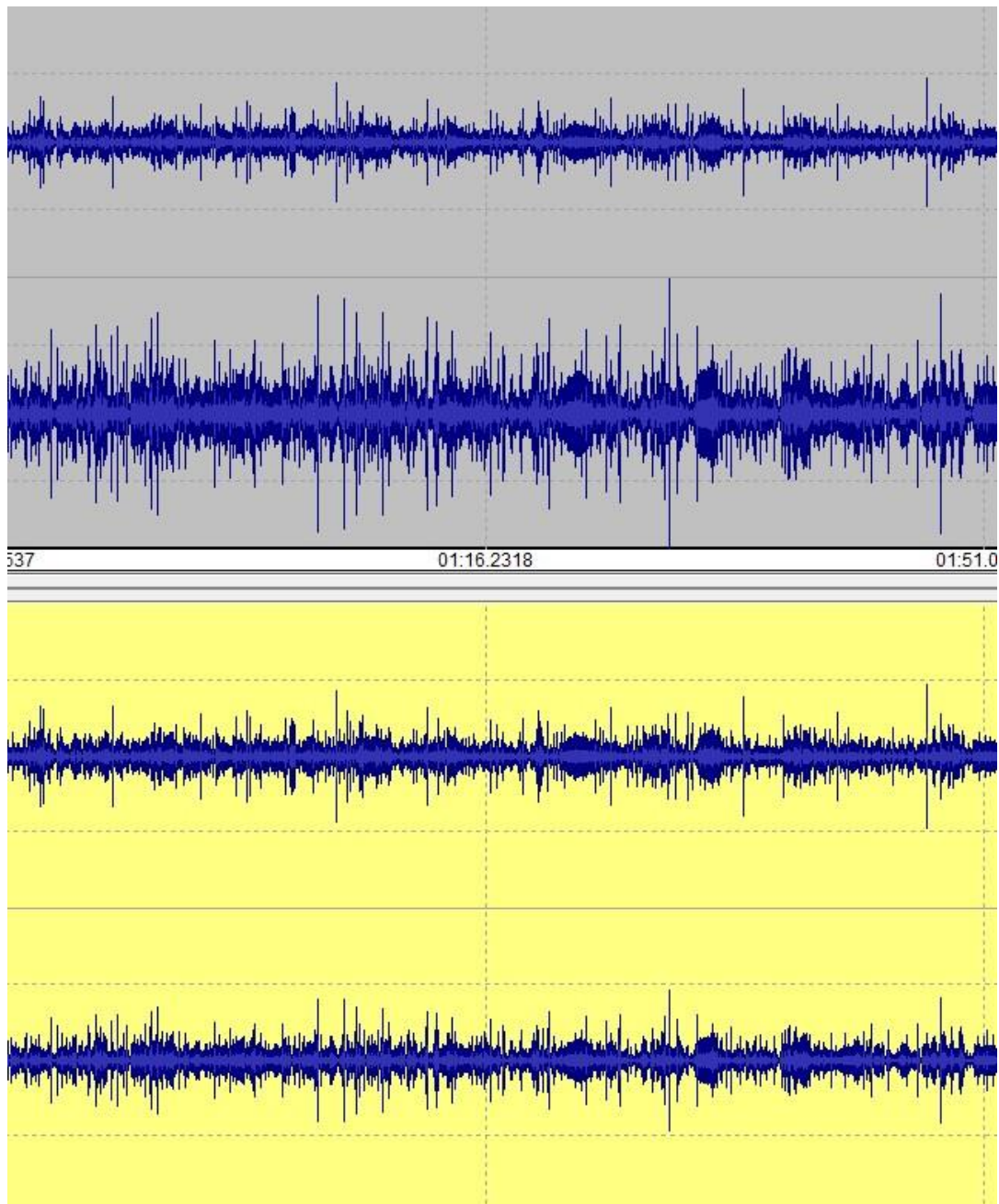


Figure 7-16 Destination (Bottom) Result from Balance Setting

After running the filter, the channels Left, and Right will be close in amplitude over the length of the song. Because the *flat* and *acoustical* setting was selected, no frequency modifications were made to the raw recording. Proceed to Trimming and checking for overload in section 7.9.2

7.9.2 Trimming File Length and Checking for Possible Overload

Check the raw file to see if you have overloaded the analog to digital conversion process by displaying the entire song and then looking for any *music* touching the very top or bottom of the chart *disregarding* short noise bursts. If the music touches the top or bottom, then the song should be recorded again, and the previously saved file deleted. Refer to figure 7-17 for an example of a good raw file. The noise spikes can touch the top or bottom as they will be removed later.

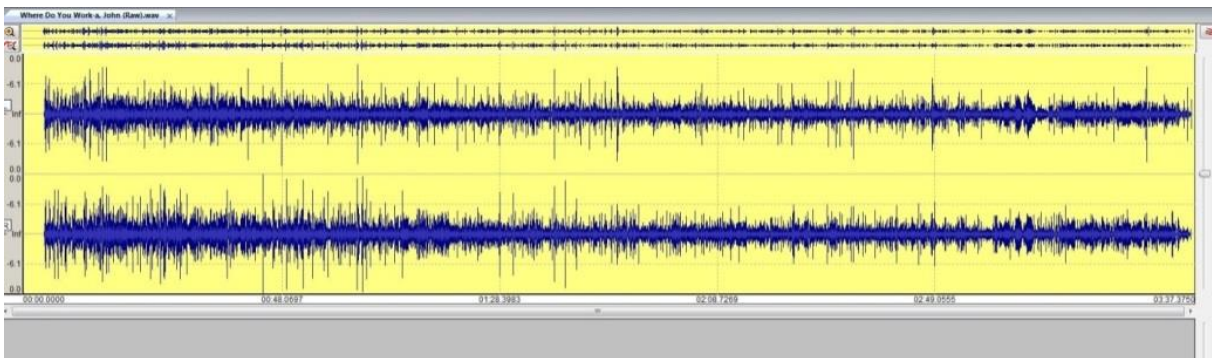


Figure 7-17 Raw Recording

Notice in figure 7-17 how the heavy waveform is well below the upper and lower limits of the analog to digital conversion process, while some noise bursts (spikes) are close to and even touching the limits at times. The high levels for the noise bursts are not a problem for the restoration process. Details of the raw recording from figure 7-17 will now be examined.

The beginning of the raw recording corresponds to the time before the stylus has settled down into the groove (dead time) and can be removed. An example of this would be when the stylus is moving via the turntables automatic starting motion of the tone arm or when the stylus is hunting to find the music groove.

This dead time may be useful for an advanced signal measurement that is found in the reference section where the noise of the recording system is measured. This section, however, will be removed for all the restoration work.

The next interval of time is after the stylus is in the music groove and before the song has begun. This section is important to keep as it provides an indication of the surface noise present on the record surface and can provide the noise filters with important noise removal information. This section should be retained for the original recording.

You can locate these various sections on the raw recording by playing back the recording and hearing the change in background noise as you watch the playback position on the waveform. Figure 7-17 has been magnified and the beginning area is shown in Figure 7-18:

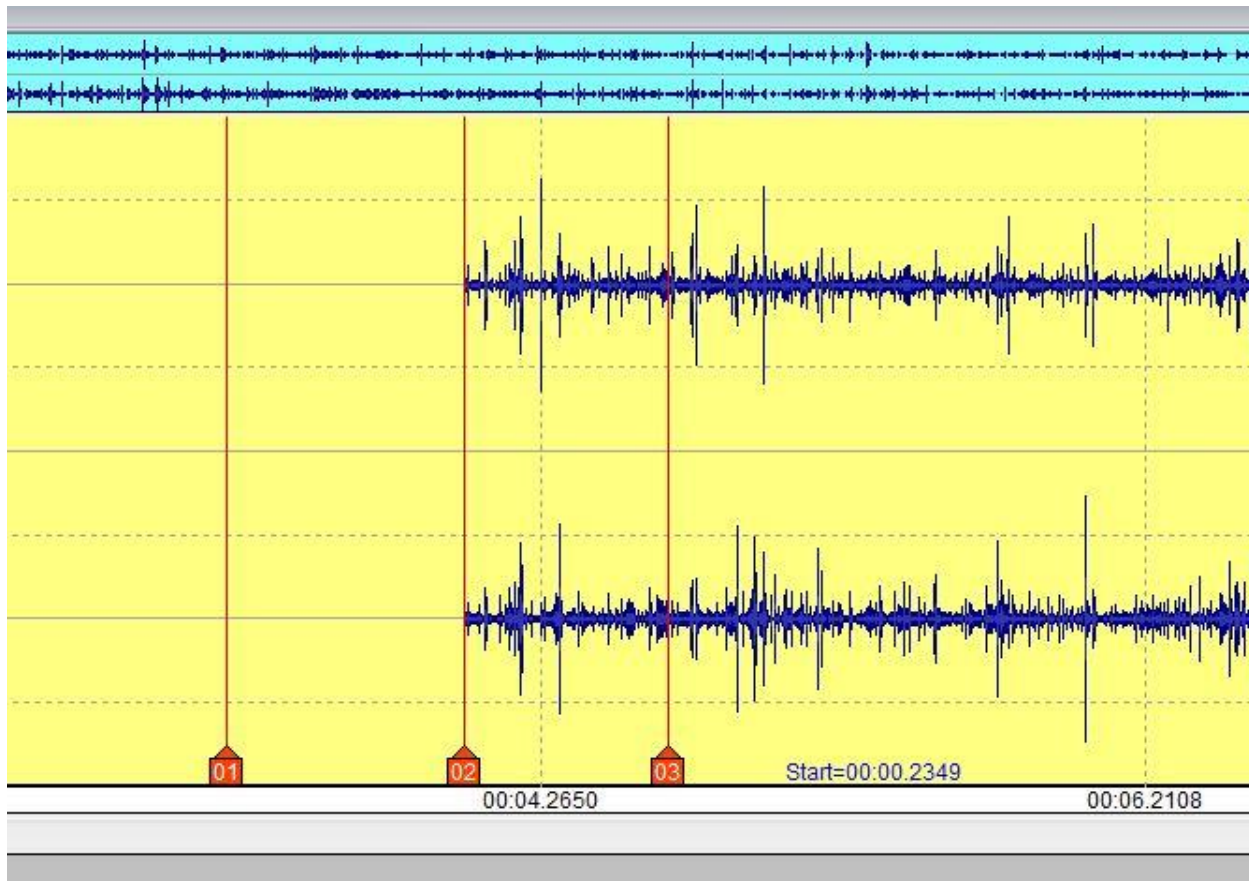


Figure 7-18 Beginning Section

Marker 1 is in the section before the stylus has contacted the record and is quiet. Between Marker 2 and 3 you can hear the sound *before* the stylus falls into the record groove. At Marker 3 the noise has changed, and the stylus is *in* the groove. After Marker 3 location the display does not clearly show the actual start of the music due to noise. This section where the music starts can be heard and identified using your hearing. The section from the beginning of the file to marker 3 can be removed so that what is left of the recording is the time when the *stylus is in the music groove and before the song has begun*. The amount of time to keep for this type of groove noise can be as short as a couple of seconds.

A similar method occurs at the end of the recorded music. When the music has ended, the groove may spiral into the center of the record which in turn may cause the tonearm to lift

and proceed into a shut-down sequence. As the song ends the level will often fade down to a quiet sound with the stylus riding in the groove with no music. When the groove spirals in and lifting of the stylus occurs, you will often hear a *thunk* as the lifting mechanism moves the stylus. Refer to Figure 7-19 for details of this ending part of the music.

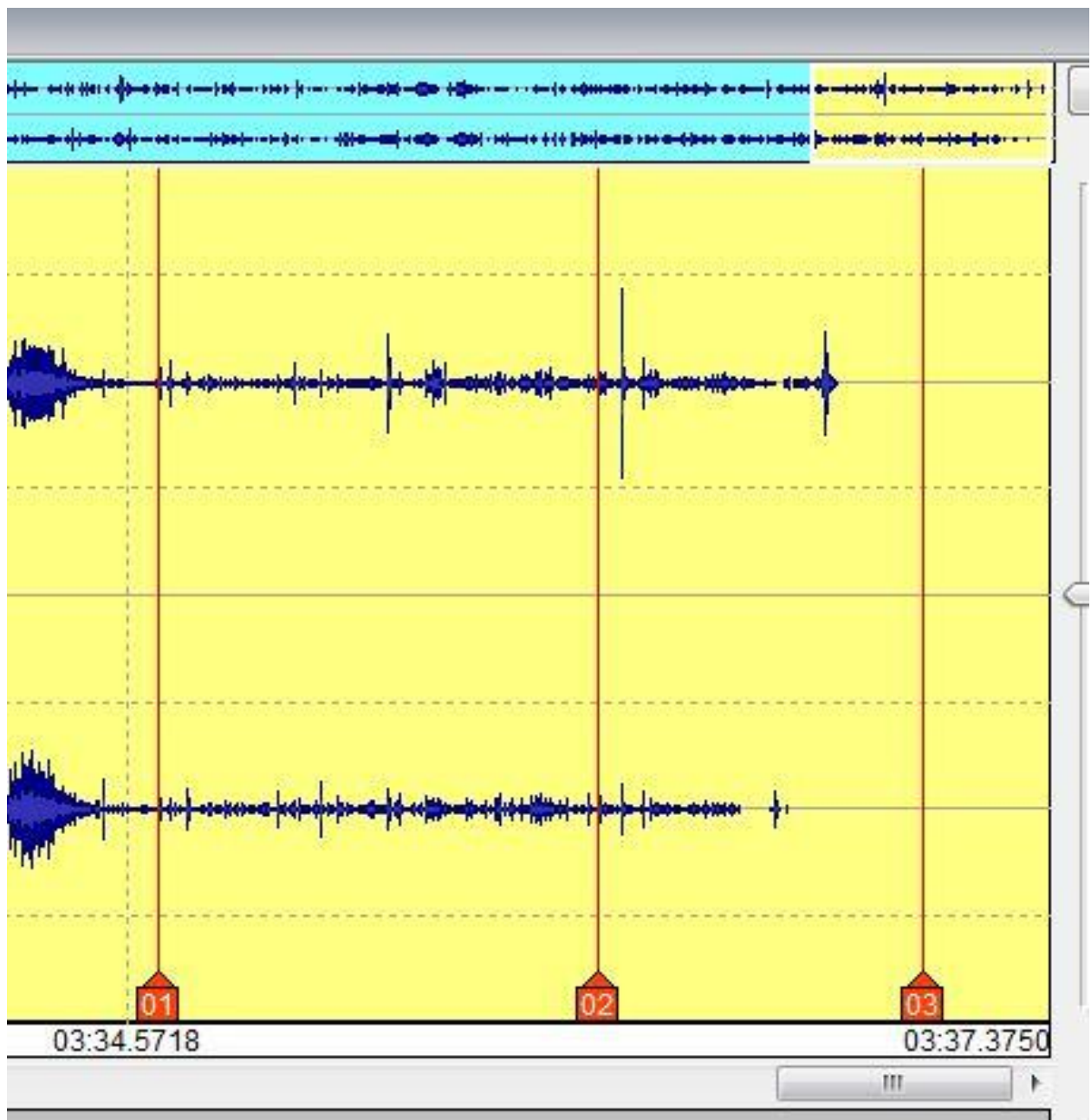


Figure 7-19 Ending of the Music

Marker 1 is the location where the music has faded to a level where it cannot be heard. Marker 2 is the beginning of the stylus lifting, and Marker 3 shows the resulting silence as

the stylus is not in a record groove. You can remove the section from Marker 1 to the end of the file. Turntables that do not have an automatic lifting mechanism will continue to keep the stylus in the groove after the music ends. This section with no sound should be removed.

After you have trimmed the beginning and end of the song the file can be closed and placed into the file location for your original recordings *if a flat preamplifier was used, the correct speed was used for playback, and the channels are balanced*. Place this file into the *Original Music* location for the artist and record type that you have previously established with the Music File Organization chapter.

If the speed of the turntable was not the correct speed for the record and/or the preamplifier applied equalization, then extra steps are needed on the trimmed file before it can be stored as an original file. Section 7.9.3 details any extra steps after section 7.9.2 have been performed on the *Trimmed Raw Recording*.

7.9.3 Extra Steps to Remove EQ and/or Correct the Playback Speed

Remove Equalization from the Recording

If equalization was applied by the preamplifier during the *Raw* recording, then this must be removed to produce a flat recording to become an *Original File* recording. You can create an equivalent flat recording with the DCart11 Software tools.

An example to clarify this will start with a raw recording of *any* type of record that used a RIAA EQ preamplifier. To create a flat recording, you must *undo* or remove the RIAA curve. The correct equalization for the record will be applied later, during the noise removal process; the only equalization to remove is the one that was applied during the transcribing operation in the preamplifier.

The chapter on Equalization and Removal contains information on the method to remove the preamplifier equalization (RIAA EQ) and that method will be repeated. After the file has been trimmed the software filter shown in figure 7-20 will be used to process the file with the settings shown.

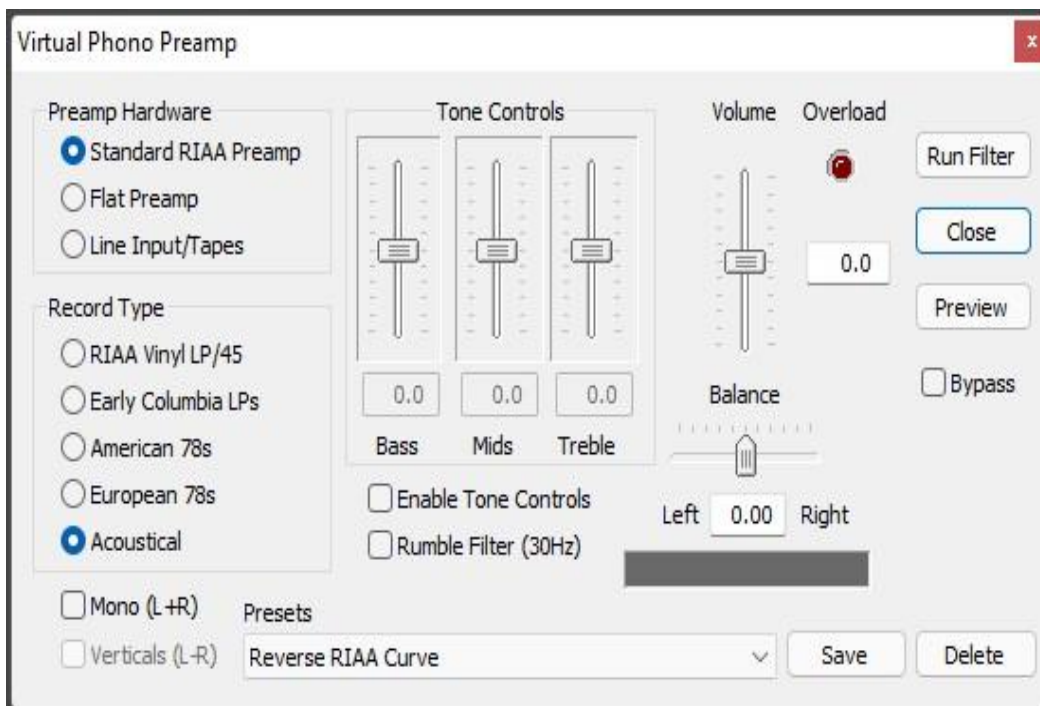


Figure 7-20 Removal of RIAA curve from a Preamp

Notice that Record Type Acoustical was selected even though that may not have been the actual record type. This software option will convert the recorded file to a flat preamplifier type if the Acoustical option was picked. We are *using* the software to remove EQ with the Acoustic Setting. After the file has the RIAA curve removed from it, it can be saved as an *original* file in your music file structure if the correct record speed was used during the transcription.

Correct the Speed

The speed for recording the record should have been the correct speed. In some cases, the speed during the transcription may have been slower than the correct speed due to either the record speed was not available on the turntable, or a damaged record required a slower speed to keep the stylus in the record groove. For the case where the recording was made using a *flat* preamplifier, DCart11 software provides a method to convert the frequencies on the recorded file to correspond to a higher (or lower) record speed. The change speed software is located under the effects tab. Refer to figure 7-21.

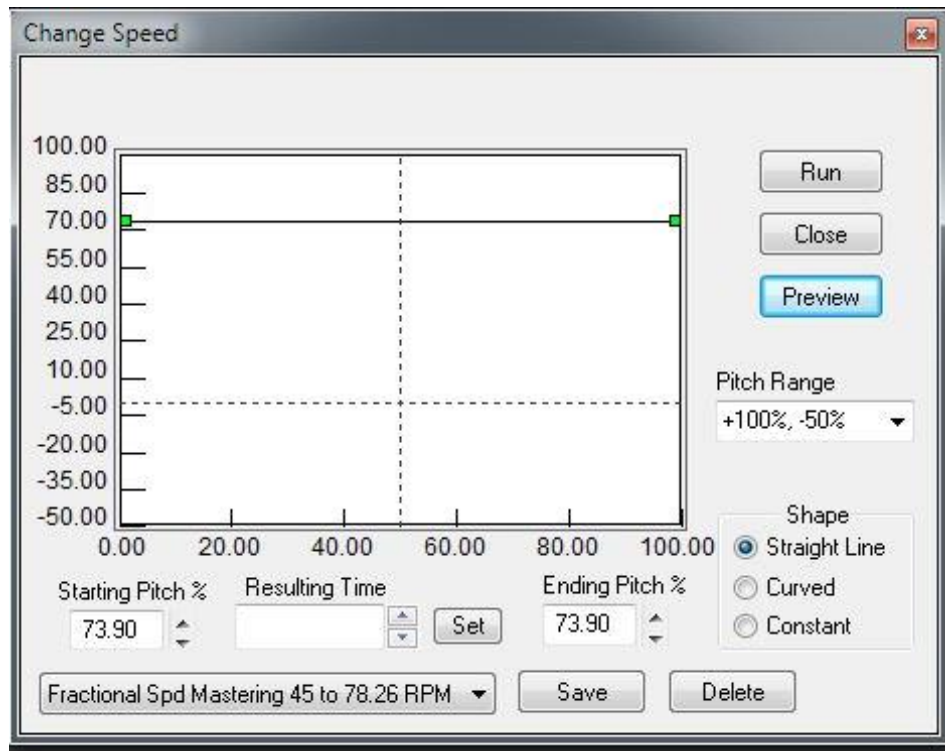


Figure 7-21 Speed Change

This software algorithm performs a speed change by modifying the digital values in the file to the desired turntable speed from which they were initially recorded. Note that the change from $33 \frac{1}{3}$ to 78 RPM must be performed in two separate steps; first to 45 RPM then to 78 RPM because of the operation of the software. If a lower speed was used to record a 78 RPM record, the preferred turntable speed would be 45 RPM.

If a *flat* preamplifier was used, then after using the speed correction you are finished, and the resulting file can be saved as an Original Recording.

If any type of equalization was performed by the preamplifier *and* a speed change is needed, a specific order in applying the software filters is required. This specific order *is very important to perform and to understand*.

Specific Order of Speed Correction and Equalization Removal

If the speed was incorrect for recording and RIAA equalization was applied, then a specific order of removal and speed correction is needed.

An example will explain why the equalization must be removed *prior to the speed correction* by picking a frequency within a song and see what happens to the level of this

frequency with speed change and equalization. Start with a frequency of 1000 Hz that has an amplitude of $\pm 10,000$ counts (full scale ± 32000 counts) which was originally recorded at 78 RPM on an electric recorded record. The 1000 Hz tone was in the *constant-velocity* part of the frequency range and no change was made to its amplitude when it was originally recorded. Now reduce the turntable speed to 45 RPM and transcribe the music with a preamplifier that has a RIAA curve in the circuit. The original 1000 Hz tone will become ≈ 570 Hz due to the speed change. The RIAA EQ curve in the preamplifier will now increase the amplitude of this tone by ≈ 3 dB or an amplitude of $\approx \pm 14125$ counts. Refer to figure 7-22 for the RIAA curve:

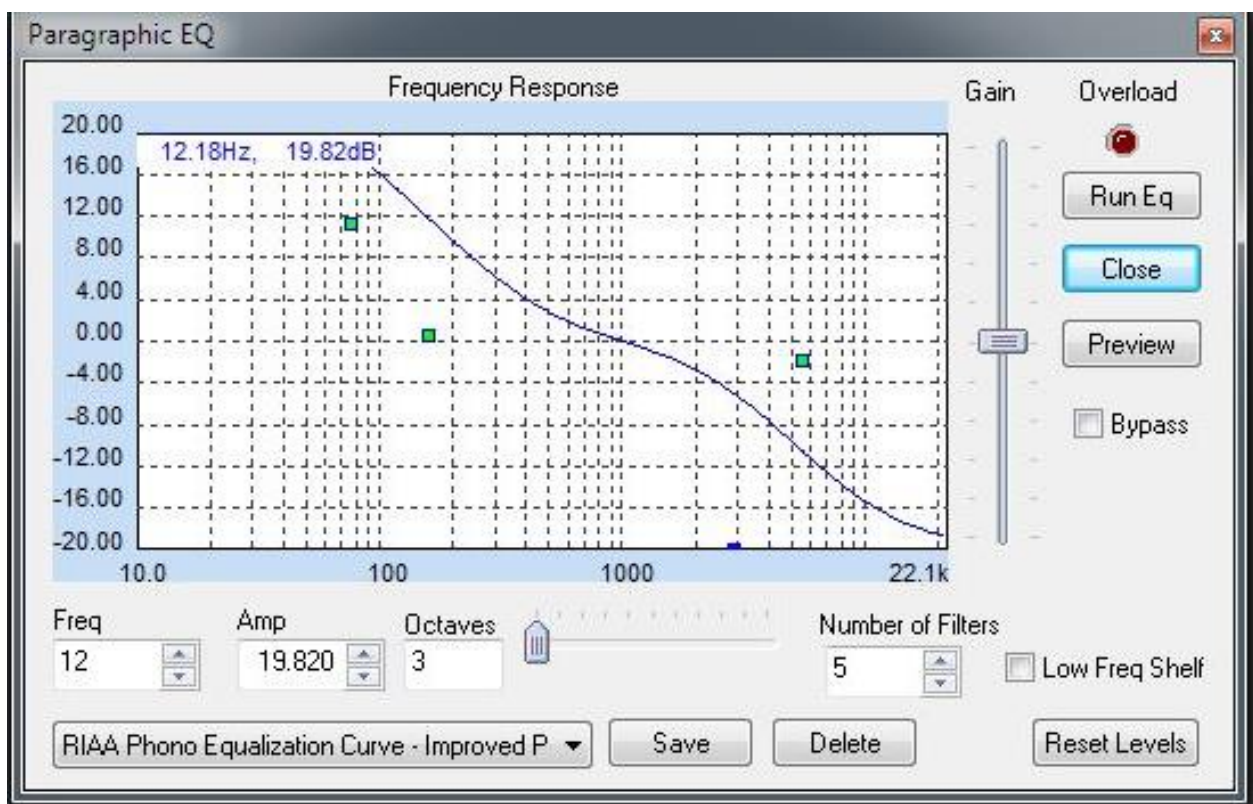


Figure 7-22 RIAA Curve

At this point two different outcomes are possible. If the transcribed file is changed to an equivalent speed of 78 RPM by using the software speed change (figure 7-21), then the 570 Hz tone is restored to the correct 1000 Hz tone but *at the wrong amplitude of ± 14125 counts*. When you next apply the reverse RIAA curve to make a flat recording, the amplitude will stay at ± 14125 counts since the RIAA EQ reverse curve provides no amplitude change at 1000 Hz. See figure 7-23:

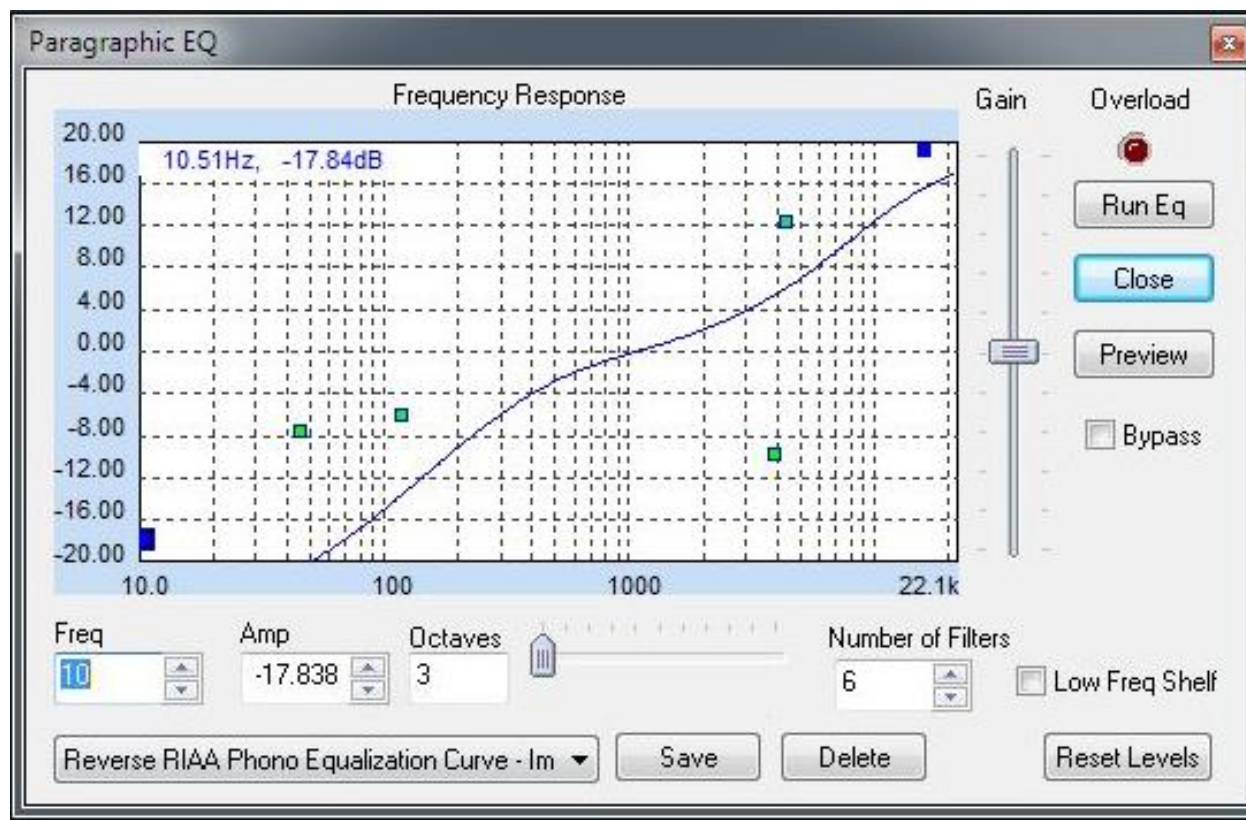


Figure 7-23 Reverse RIAA Curve

The resulting music will not sound correct for 1000 Hz and many other frequencies. Now perform the techniques in the correct order. While the music file is still at the lower initial speed of 45 RPM, apply the reverse RIAA EQ curve. This will take the amplitude of the 570 Hz tone down to $\pm 10,000$ counts since the curve is a mirror image to the RIAA curve in the preamplifier. Next when the speed change is made to 78 RPM, the 570 Hz tone becomes 1000 Hz at the correct amplitude. The order of applying the equalization is very important.

The specific sequence is to *first remove the equalization before any speed correction has been made*. After the equalization is removed from the recording, at the speed it was transcribed, the file is now equal to a flat recorded file. Next, the speed correction can be used to bring the music tones to where they would be if they had been played at the correct speed. This method is the only way that can be used for a successful conversion of this type of raw file to an original type for music restoration.

The original file should always be preserved as it is. Make a copy for any working file.

8.0 *Foundation Methods of Noise Removal & Level Creation*

This chapter covers the Methods for removing noise from the music file and the locations for the results. The location for your noise removal work will be in both *Level One and Level Two* music files. Later chapters will describe *Specific* noise removal multifilters for different types of records that build on the methods in this chapter.

These methods form the underlining techniques to remove the noise from the recording. The structure of locations for the music files provides a method to return to your Restoration Work without the loss of your previous noise removal effort.

8.1 *Definition of Noise*

For music restoration, noise will be anything that has been added to the original recording that was not wanted. The distortion that occurs to the music from record wear is one example.

Different terms for noise are used to qualify the type of noise. For example, in the DCart11 software there are terms for noise using a specific word to describe the character of the noise. Terms used are impulse, crackle, narrow crackle, and big click. A further distinction is made between *impulse noise* and *continuous noise*. Impulse noise is any type of interruption in the music that is of short duration. An example would be the sound from a crack in the record when played. The other type of noise is continuous or present though out the music. An example of this type of noise occurs just before the music starts and continues throughout the song. These two general types of noise will be removed by two different Multifilters labeled Part One and Part Two. These multifilters will be described in detail for specific types of records in later Chapters.

Another aspect to record noise is that the shape of noise *is unique for each type of recording media*. The shape of the noise waveform found on a 78 RPM electrical record has a different shape than that from an Edison Diamond Disc record. The stylus motion for the vertical cut records gives a different noise signature than the lateral cut motion. The noise software filters while have unique settings for these different types of recordings.

Multifilters are a useful concept within the Diamond Cut Software that allow a series of individual filters to be strung together into an overall block. While it is possible to perform each operation in a separate step, using multifilters have important advantages. One advantage is that all the operations in the block are performed as digital words in a continuous mathematical operation whereas performing each operation separately involving

mathematical conversions. Another advantage is that you can review and see your total operation on the music in one filter containing all the unique settings for each individual part.

8.2 *File View used in DCart11 Software Display*

DCart11 software has two modes for viewing and editing the audio data; namely *classic edit* and *fast edit*. The classic edit mode will be used in the handbook as the ability to view both the source and the destination while removing noise will be the *Key* to the success of the noise removal process. This advantage will be later shown in this chapter under the Sight & Sound section.

8.3 *Sample Rate and Bit Depth*

A sample rate of 96 kHz for the music will allow the impulse noise filters to work at their peak performance. If you have a music file that was *not at this sample rate*, then the DCart11 software may not remove as much noise as a higher sampling rate could do. Also, the noise filters can usually be set to more aggressive setting at 96 kHz than a lower sampling rate. If your original file is at 44.1 kHz you can still achieve excellent results, but the noise removal may take longer.

The DCart11 software provides an algorithm to increase or decrease the sample rate and bit depth via software however; this is not a substitute for an original recording at the higher rate. If the source file was originally recorded at 44.1 kHz, then the noise should be removed at this rate and not converted to a higher value. Useful noise information cannot be added to the original recording via an algorithm that increases the sample rate or bit depth as the algorithm has no knowledge of the noise properties.

The continuous noise filter settings are designed for a music sample rate of 44.1 kHz. Higher sampling rates do not improve the CNF performance. The files sample rates will be converted, if needed, after the Impulse noise is removed to 44.1 kHz.

16 Digital Bits to represent the music is more than adequate for the Signal to Noise ratio of all types of flat records.

8.4 *How Much Noise can be removed by the Filters?*

This question is important to understand. Any software algorithm that removes noise from a music file does not have the ability to consistently remove just the noise and not some of the music. The different software algorithms must make decisions on millions of digital bits and while they do use very clever engineering techniques, the algorithms will fail at times. The use of these noise filters can remove some of the desired music so you must be careful with the settings. When the noise filters are being used, it is very important to listen to the music *before and after* the filtering to determine if your settings were too aggressive and caused some of the music to be removed. When I first started to restore music, I would set the filters to very aggressive setting so that after just one pass through the music I could not hear any noise. Later, I realized that I had also removed some music!

Universal settings for all noise filters are not possible due to the variation in the condition of the records *and* how the content of the music affects the performance of the noise filters. The best method is to start with the filter settings at a low level and then try a pass through the music. If you have a relatively small number of noise events left after using the filters, then the remaining noise can be removed by a manual technique called the *Sight & Sound* method. While very aggressive filter settings can give the impression that the noise has been removed; some music will have been removed along with the noise.

It may not seem correct that you may have to be remove noise manually when you have software that can do the removal for you; however, a balance is needed between time to manually remove noise and removal of desired music.

There will be records that have more noise that you can remove. There were many millions of records produced and if the recording you have is not *cleaning up* as well as you want then it is best to stop at the best you can achieve. You can often find better versions of the recordings as you continue to add to your collection.

Brass instruments create a music waveform shape that can look like a noise event instead of music. There is a box that is checked for all the filters to help avoid the removal of brass sounds from the music. When filters start to remove brass notes, the sound becomes somewhat “mussy” instead of having a sharp and clean sound.

8.5 General Noise Pictures & Analysis

For an example of the noise removal process, an original song file before the use of any noise filters will be used. This working copy is placed as a source file. The transcription was performed with a sample rate of 96 kHz and a digital word length of 16 bits. The file is displayed using DCart11 in classic mode. Refer to Figure 8-1 below.

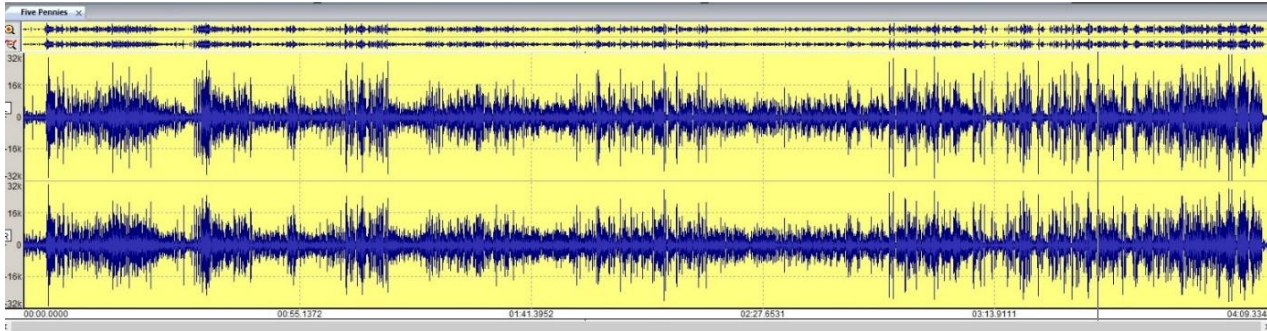


Figure 8-1 Original Song File

When the total song is displayed on the screen, waveform detail is hard to see, however, there are several large spikes that show sudden changes in the amplitude of the music. Specific sections will be Highlighted and Magnified using the zoom in Button on the Tool Bar.

8.5.1 Magnified Picture of Noise

Figures 8-2 and 8-3 magnify some noise events for more detail.



Figure 8-2 General Noise in Music

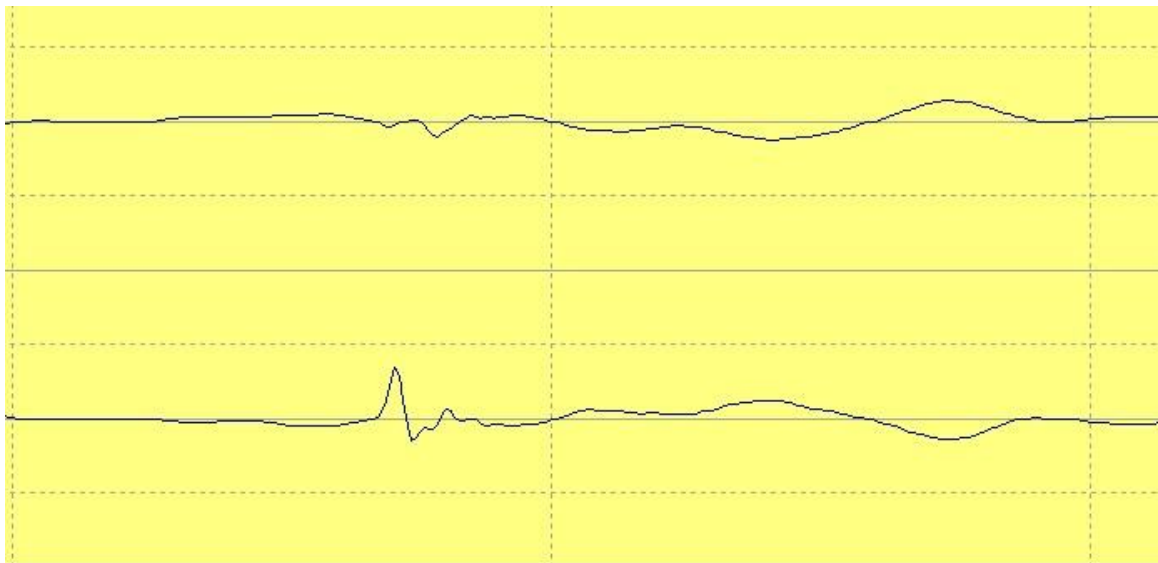


Figure 8-3 More Magnified Detail on Noise

Notice the shape of the noise has fast amplitude changes and that there are differences in the shape between the right (top) and left (bottom) channel at the same instant in time.

8.5.2 Use for the Section before the Music Starts

Figure 8-4 is a zoomed in display that shows the moment when the stylus is in the record groove and when the music starts. The location between Marker 1 and Marker 2 in figure 8-4 shows the time before the music starts, which is the surface noise of the record. The music starts at marker 2.

The reason that the waveform line does not appear as thick as it does in this same area in Figure 8-1 is that the time scale is different between the zoomed part vs the original and now the signals are not *bunched up* in time.

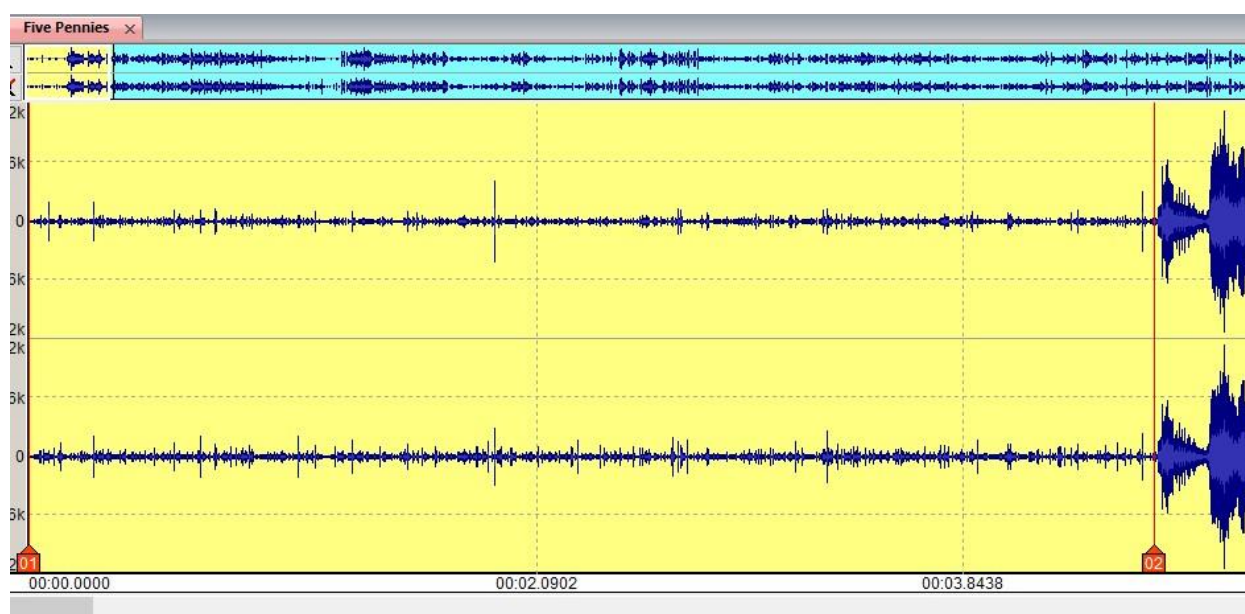


Figure 8-4 Time before the Start of Music

There is noise before the music starts that comes from the texture in the surface of the record which causes the stylus to move in a somewhat random motion. This section just before the music starts is used by the *EzImpulse Filter* in its noise removal algorithm. It is important to keep the record noise before the music starts for a short amount of time (couple seconds or more).

This section before the music starts has another use during the application of the *Continuous Noise Filter (CNF)* in the Level Two File Generation. The methods later shown for specific CNF settings will start with a general setting that is effective for most records. A change to the CNF from the general settings can occur in the Level Two file creation using

this period from Marker One to Two as a new *sample* for the CNF for a specific song. Later in the final part of the restoration process, this section is removed.

8.6 *Noise Masking Technique for Beginning and End*

There is a useful noise reduction method that can be used when the music is loud, which is the ability of the music to over-ride or mask the record noise. Many songs have a constant level of background noise present on the song. Records made prior to the use of a vinyl material have a built-in noise that is noticeable. The way that the ear hears and processes sound, can be used to reduce this background noise; namely that when the music is loud, your hearing cannot distinguish the background noise from the music. Therefore, when the stylus first settles into the groove and the music has not started, the listener will hear the background noise and be conditioned to having this noise present in the recording. When the beginning of the song starts, often the music is loud and now the background noise cannot be heard...however... the listener had heard the noise earlier and to some extent that impression will stay regarding the quality of the song. As explained in section 8.5.2, this noise before the music starts will be useful for noise removal, however when the final restored version is ready this part will be removed. By removing this *initial* noise, the overall quality of the music will be increased. If a moment of silence is desired prior to the start of the song, then the software can add silence, void of any noise.

Another aspect to record noise and the ability of the music to mask its presence occurs at the end of songs during the common fade down ending. Many songs do not suddenly end. Rather the volume is slowly reduced to the point where you hear both the music and background noise at the same time. Now the record noise that is always present will stand out. One method can help to reduce noise at this point by adding additional fading to the music over and above the original fade down performed by the recording engineer. Since additional fading *at this point* reduces both the music and the noise, the overall effect on the noise presence can be improved. You must use careful judgement when additional fading is used since you do not want to remove a part of the song when the music is still prominent.

An example will demonstrate this method. Figure 8-5 shows a song with a strong presence of background noise remaining while the music is still playing during the slow faded down ending.

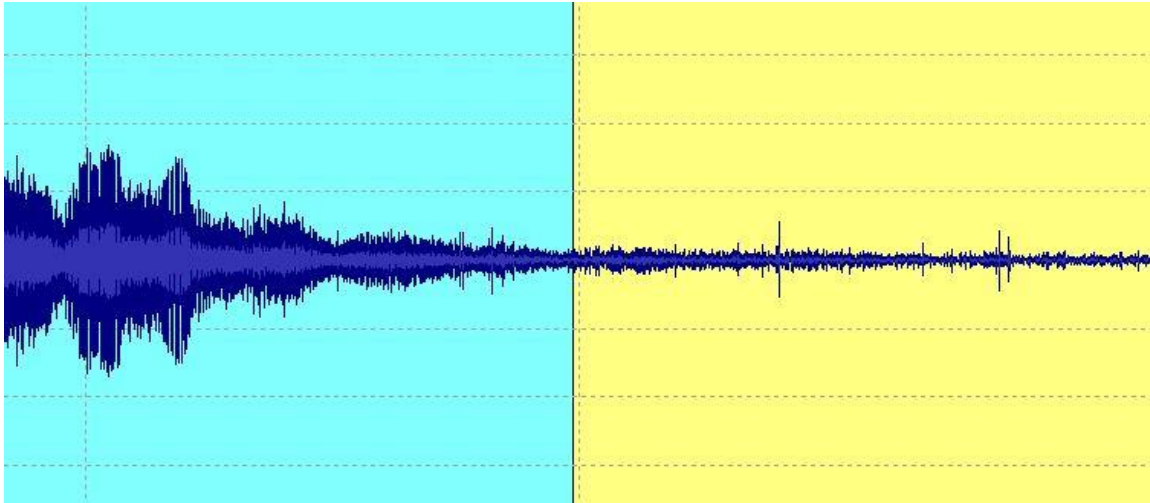


Figure 8-5 Ending of the Song

At the point where the vertical black line is located, the level of the music has decreased to a point where the noise dominates the music. The music is still undergoing a reduction (fade down) so that a simple stopping of the song at the vertical black line would not sound natural. To improve this song the fading after the black line will be increased so that the highlighted area to the right of the line will undergoes more amplitude reduction. The DCart11 edit command fade down is used. Refer to the figure 8-6:

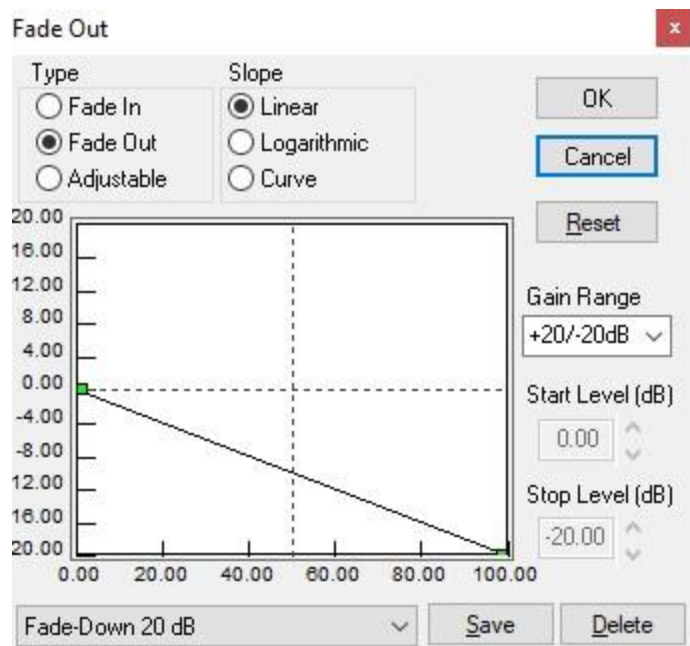


Figure 8-6 Fade Down

This software option shows a line indicating the reduction in the signal level giving a fade down. The result of the use of this signal reduction is shown in figure 8-7:

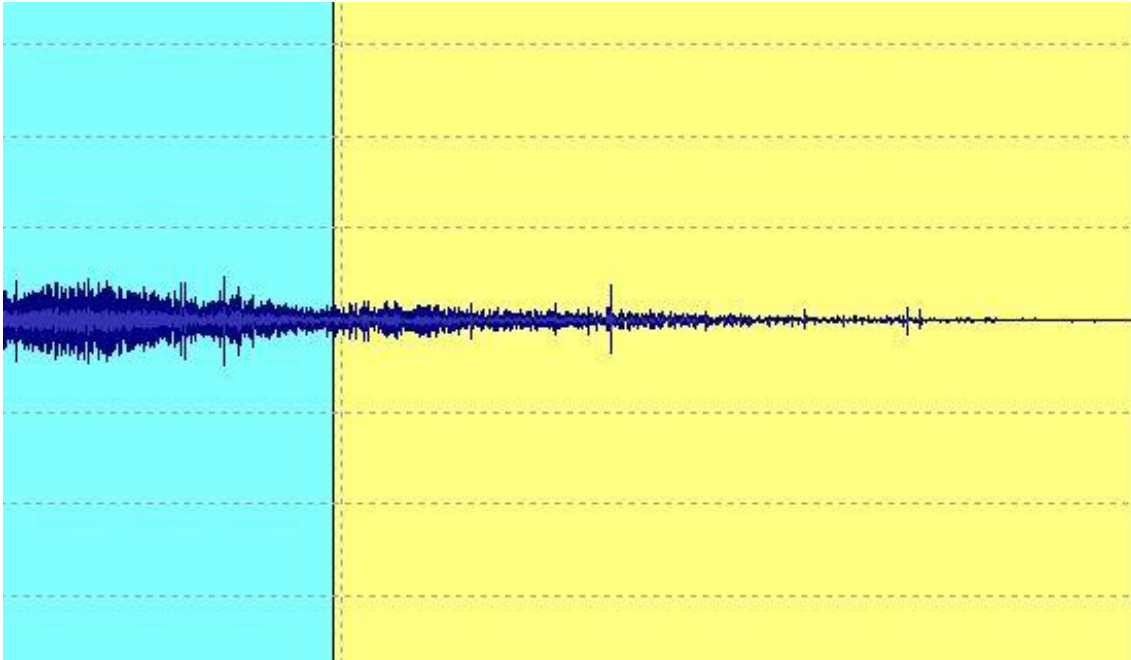


Figure 8-7 Fade Down Result

When the song is played the additional reduction in level at the end of the song will still seem natural and the background noise will be reduced. The value of 20 dB was found to provide a smooth ending to the music. The remaining noise events will now be hard to hear.

The use of both noise removal prior to the start of the music and possible extra fading towards the end of the song will be performed at the end of the Level Two file creation.

8.7 *Noise Methods for all Types of Recordings*

8.7.1 *Two Sequences of Noise Removal Are Required*

The removal of noise for all records involves similar steps and uses Multifilters named *Part One* and *Part Two*. The terms *Level One* and *Level Two* describe two *Sequences* of using the Multifilters along with additional steps.

The reason that the noise removal process will be performed in two separate levels is due to the nature of the noise, the variation in record equalization, and the operation of the software along with the ability to return to previous work.

The results of using the Multifilter and additional operations are stored as separate Folders in the Music Restoration Level Folder. For 78 RPM Electric recorded records the Multifilters used for noise removal will be 78 Electric Part One and 78 Electric Part Two. For other types of records, a similar labeling process will be used.

As seen in figures 8-1 to 8-4 there are two types of noise: impulse of short duration and continuous noise. The DCart11 Software has specific tools to deal with these types of noise and they are different in operation and will be performed in separate parts of the noise removal which are called Part One Multifilter and Part Two Multifilter. Part One Multifilter performs operations on impulse noise, and Part Two performs operations on continuous noise. The CNF (Continuous Noise Filter) works best after the impulse noise has been removed in Part One and is performed during Part Two.

The DCart11 software uses high frequency information to help remove the noise. A flat recording will have the greatest amount of high frequency noise since the high frequencies present on the recording have not been modified by an EQ (equalization) curve. In the Part One Multifilter the noise is removed first and then the EQ is applied as a last step.

Another reason to use two types of Multifilters is due to the ability to see and hear the exact location of the noise event. The use of the CNF can cause the introduction of a noticeable *time lag* between events in the source file and the destination file. This time lag during noise removal increases the difficulty in finding the exact location for the noise during the Sight & Sound manual noise removal. Also, the reduction in background noise with the filter causes small impulse noises to be harder to detect with your hearing.

Location of Stereo to Mono Conversion

In the Part One Multifilter for vertical records, the first operation performed is conversion to a Monaural file followed by noise removal. For lateral or 78 records, this conversion is performed after the noise removal. The order of conversion from Stereo to Monaural within the Part One Multifilters was found to depend on the type of groove modulation; lateral or hill and dale.

Filters for Impulse Noise

DCart11 contains several types of filters to remove impulse noise. For Level One use, a progressive series of EZ Impulse Filters will be used. The use of multiple filters can provide increasing noise reduction with increasing settings. Refer to figure 8-8 for an example of an Impulse filter using preset values from DCart11.

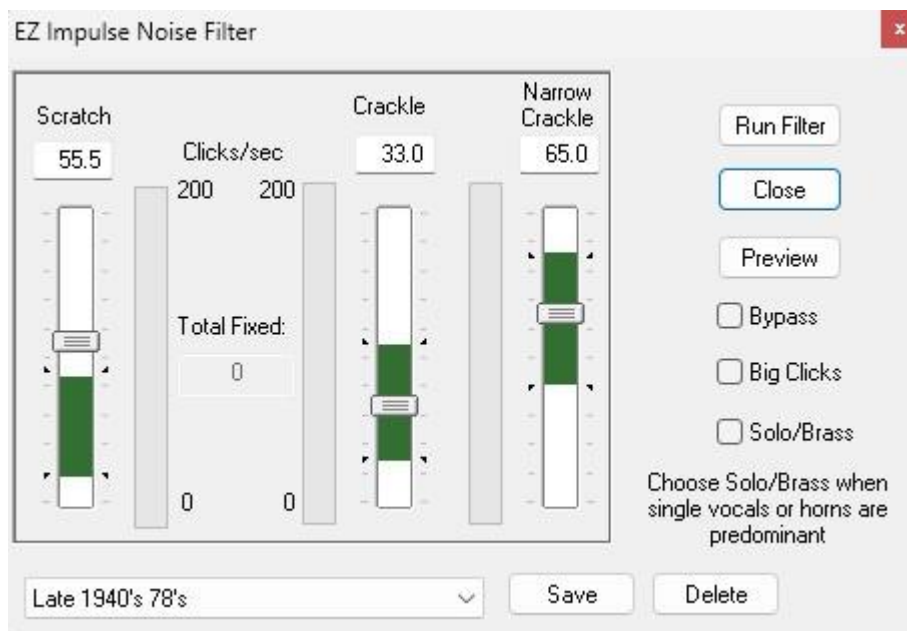


Figure 8-8 EZ Impulse Noise Filter

How to Manage Possible EQ Changes for Electric Recordings

In-order to preserve your noise removal work, and to allow for changes in the applied EQ for the music, a specific order of using the Part One and Part Two Multifilters are needed to create, in-turn, the Level One and Level Two music files.

The desired EQ will be applied as the last step in the Part One Multfilter (None for Acoustic Records). However, the manner that Level One and Level Two music files are created will allow a future change to the EQ without losing your noise removal work. The method to allow *possible* EQ changes is have no frequency modifications made to the Level One Files and allow frequency modification to the Level Two Files.

How it is Performed

The desired record equalization will be applied after a Level One File has been created. The Part One Multifilter and Part Two Multifilter will *both* be used to create a Level Two file. The Part One Multifilter will be used to remove the impulse noise from the Source file, but the Destination file, minus noise, will not be saved *at this point*. *The source file will be saved later for the Level Two file along with the desired EQ (Or Not for Acoustic)*. Later examples will help to clarify this concept.

The creation of the music located in the Level One folder, requires time for your restoration due to the potential manual noise removal. Since the application of EQ to the record cannot be removed in software for all but RIAA and NAB Tape curves, you can always return to the Level One file and try another EQ without redoing the labor-intensive manual noise removal operation. The fact that all the Level One files came from a Flat Recording means that possible changes to the EQ can be made without losing your noise removal work by repeating the generation of a Level Two file.

Various EQ options may be used while listening to the results for the song and the desired one will be used to create the Level Two file. The EQ used in the Part One Multifilter, can be changed to create another Level Two file. The Level One File will *still* be a flat file since the *Source* will be saved as the Level One file and not the *Destination*.

Acoustic Recordings

For acoustic records, no EQ is needed. The methods are similar except that the EQ stage is not present in the Part One Multifilter. Separate Level One and Level Two folders are still needed, due to the application of the CNF.

This order of noise removal operations will be repeated for each type of record within the Level One and Level Two file generation sections shown in later chapters.

8.7.2 Creation of a Level One File

The result of these steps will create a music file for the folder *Level One* so that when the noise removal multifilter Part One is used on this file, a noise-free file will remain that needs no additional *impulse* noise removal.

If you have manually removed noise events, that could not be removed with the software filters, you may want to return to this file in the future, without losing the manually removed noise events. If you keep the music file containing noise that *can* be removed at this time with your noise filter and in the future improve the filtering even more, you can *return* to the same point, with your previous noise removal work saved.

An example is in Figure 8-9. In the music the *source* file contains a completed Level One file. The destination contains the *result* of applying Part One Multifilter. The section in the source contains two channels and a noise spike just before Marker 1. The Destination contains a single channel and the removal of the noise spike.

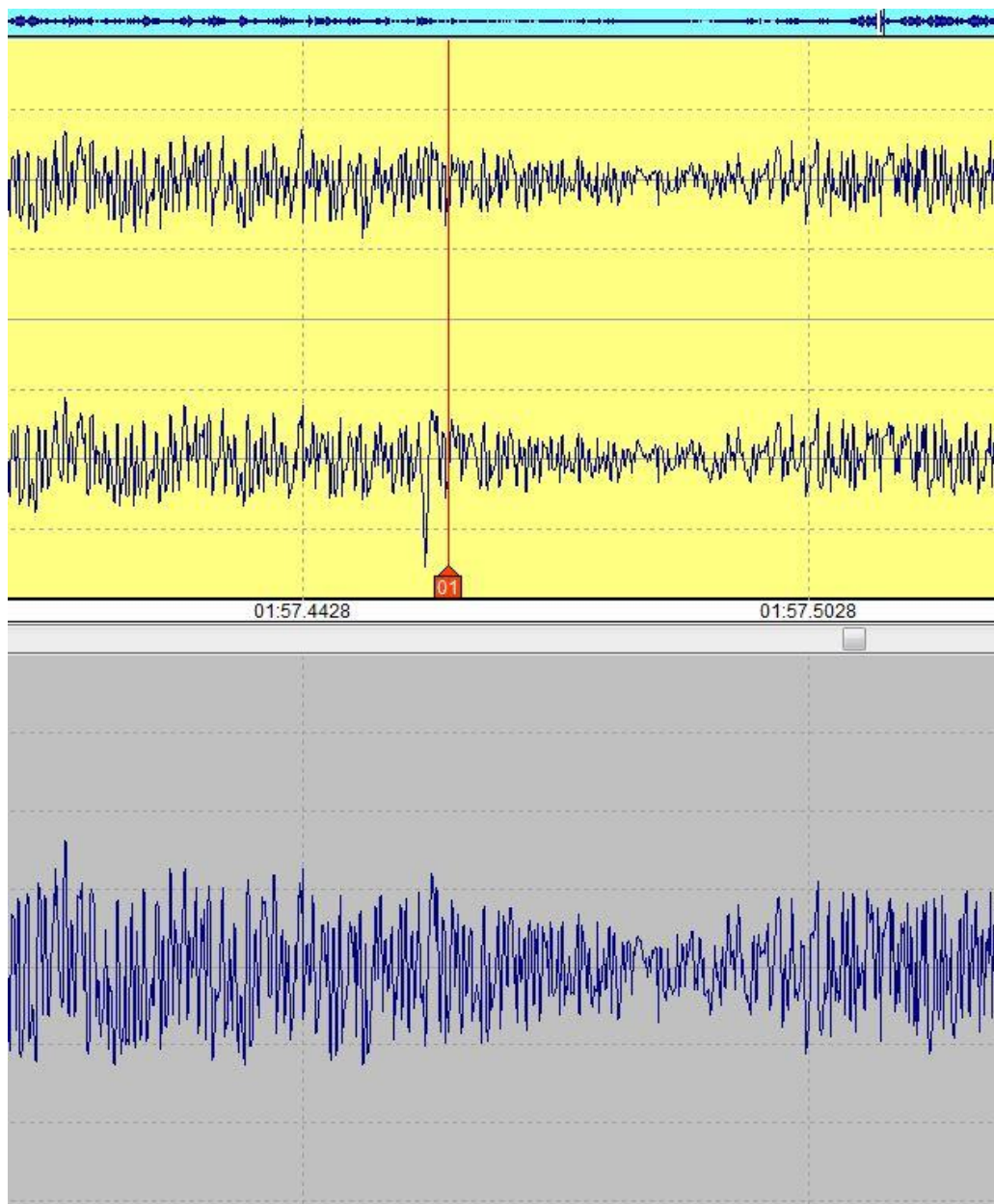


Figure 8-9 Noise Removed from Source using Part One Multifilter

All files that are in the Level One folder will have impulse noise removed *only when* the corresponding Part One Multifilter is used on them. If the noise could not be removed at first by the Part One Multifilter then extra manual removal work is needed to create the Level One file. This manual method is in 8.7.3. After any manual noise removal is performed; *A final Level One File requires ONLY the application of the Part One Multifilter to produce a noise free result.*

8.7.3 Level One Manual Noise Removal Sight & Sound Method

After the DCart11 noise filters have removed as much of the noise as possible without removing the desired music and if *some noise remains*, a manual method is used to finish the removal of impulse noise for the Level One file. The manual method uses a combination of *Sight & Sound* to find the remaining noise event. Although this method uses manual methods, the technique will include the Part One Multifilter to *pre-condition* the music to locate the remaining impulse noise.

The original file is opened as the source file using the classic edit option for file view. Next, a specific multifilter for the type of record is run and the result is shown in the destination window. The multifilter used will be the *Part One Multifilter*. The next step will be to find any noise events that are left in the source from the use of this multifilter and remove those noise events using manual methods. The result of automatic and possible manual noise removal technique allows the desired Level One file to be produced.

The manual removal of the noise events works well since when your ears hear a noise, your eyes will then see the specific location of this noise. This method requires that the source and destination files be in sync so that the specific location at the *source* can be removed when you hear the noise in the destination. In the DCart11 software there is an icon in the toolbar to Sync the source and destination files, which should be selected. Note: Even when the Sync icon is selected, the use of some types of filters will cause a time lag between source events and destination results. The filters selected in the Part One Multifilters do not create a noticeable time lag. The Sight & Sound manual noise removal process consists of the following steps:

1. Keep the Multifilter window on the screen and as far down as possible but still able for you to press the Run Filter button.
2. Starting at the beginning of the destination file, carefully listen to the music while watching the cursor move. When a noise event is heard, immediately stop the playback of the music (press spacebar or stop button), and locate the source of the noise in the destination window. When the noise event is found, look above to the same location in the source window using the playback cursor. You will be able to locate the specific event that the multifilter *did not* remove in the source file. The zoom in button and highlight section along with the replay of the zoomed in destination section can be performed several times to work down to the specific source of the noise. Remove the noise event in the *source* by using the manual interpolation *I* key or other manual methods.
3. After the noise event in the source has been fixed continue to play the music in the destination window *just after the last noise event* or re-do the multifilter again on the edited source (remove the previous destination file first and press the run filter button) and continue to listen for more noise and if needed remove additional noise events from the source file.
4. When the destination file has been completely heard and the remaining noise events removed to your satisfaction the edited *source* file now becomes a Level One File.

An example of this sight & sound method using the 78 Electric Multifilter Part One with a 78 RPM Electric Recording is shown in Figure 8-10. The source is the original recording at 96 kHz and 16 bits word length, and the destination contains the result of the filtering.

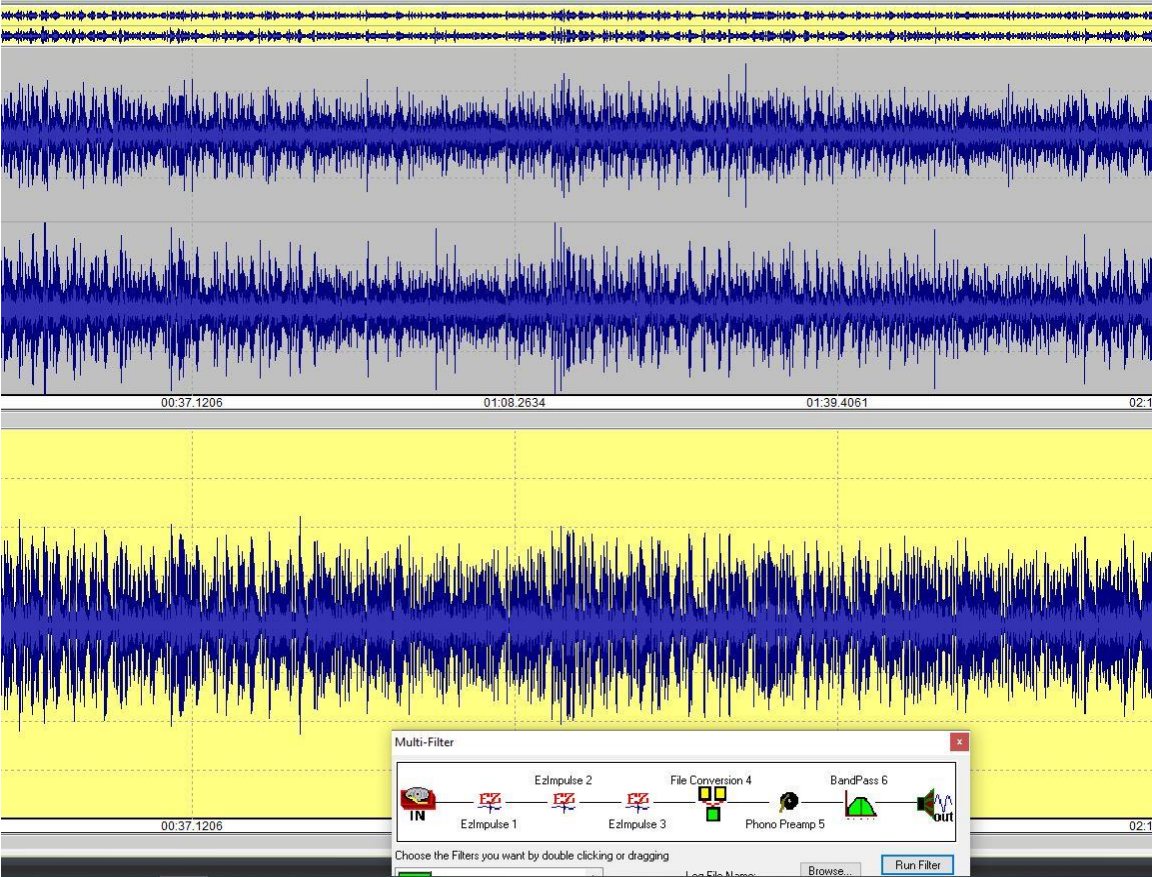


Figure 8-10 Ready for Sight and Sound

Some remaining noise pictures from the use of Part One Multifilter are in figure 8-11.

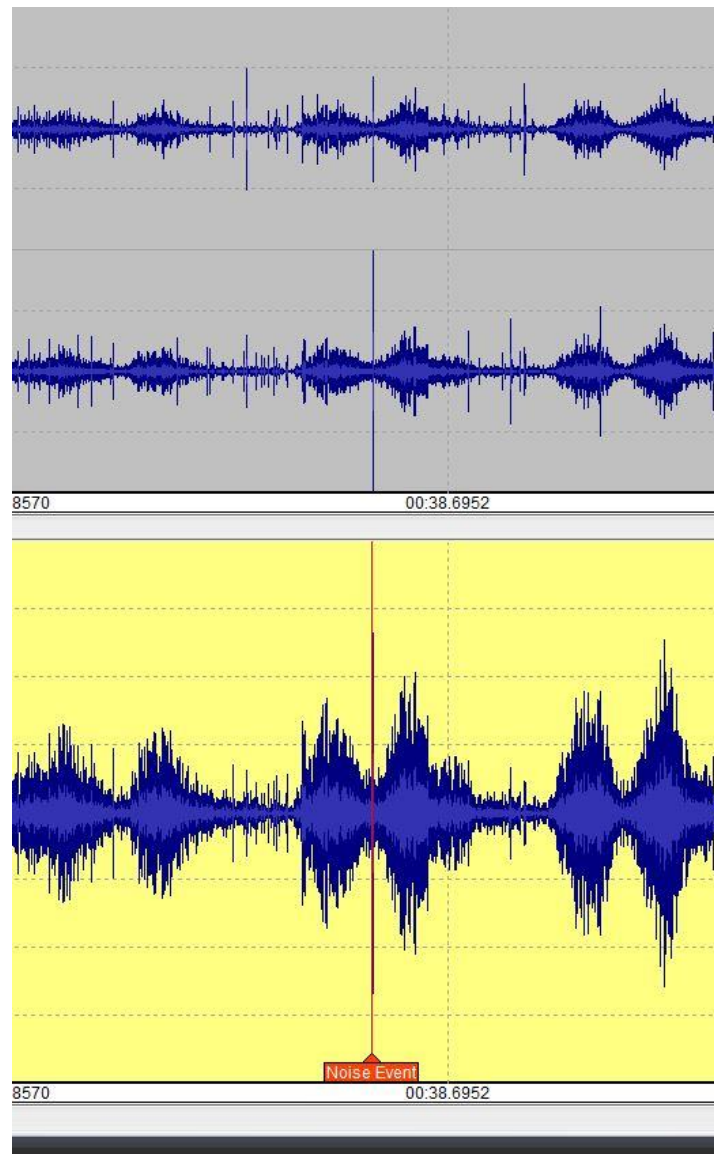


Figure 8-11 Noise Events in the Destination/Source Windows

The marker shows the noise event in the destination window that was not removed by the 78 Electric Multifilter Part One. The corresponding source file location shows the noise event that you want to manually remove. Figure 8-12 shows the noise event in the *source* zoomed in and highlighted prior to the use of the manual *I* key.

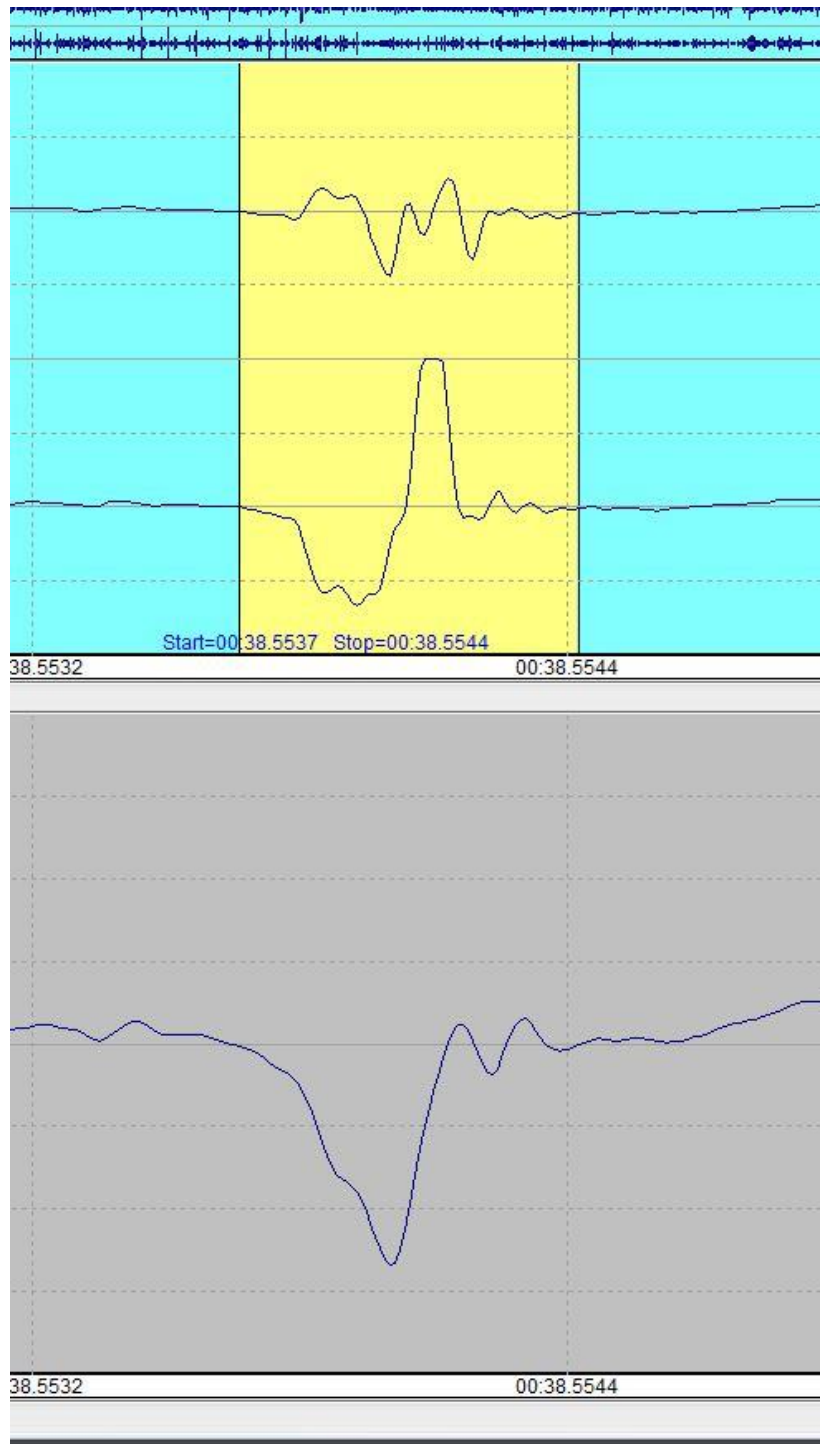


Figure 8-12 Noise Events Prior to Removal

The highlighted section is where the software will remove the offending noise event by using the *I* key. The result of removing the source noise is shown in figure 8-13:



Figure 8-13 Manual Noise Removal Results

Notice that the noise event has been removed from the source, but the noise is still shown in the destination file. The noise removal from the source can be verified by removing the previous destination file and re-running the Part One Multifilter and hearing the music in the new destination file. An alternative method is to advance the playback position on the destination file slightly after this noise event and continue to hear the music and listen for another noise event. By jumping ahead of the last noise, you can save some time by not waiting for the multifilter to run again after the last manual noise removal.

Some caution is needed with the manual method of removing noise since the use of the *I* key requires that you place the interpretation range, highlighted in Figure 8-13, around the offending noise with the correct interpolating interval. The best method for manual removal is to always check the results of the correction by looking at the music before and after the use of the key to verify a smooth replacement of the music's waveform. You can use the undo key and try a slightly different area for the interpolation range if the result is not smooth and natural. While the manual method of noise removal may seem difficult to use it will become second nature as you use it. The *I* key works over a large range of noise types.

This process continues until the source file has had all the noise events removed that the multifilter could not remove. At this point, the source file should contain impulse noise that can be removed solely with the 78 Electric Multifilter Part One except for continuous noise. A good check would be to run the Part One step one last time to check for any noise that you may have missed in the last pass.

Save the *Source* in the *Level One* location in your music folder. The file will remain at a sample rate of 96 kHz and 16 bits of word length. The source now had extra noise removed by the Sight and Sound method.

Help in Finding the Noise during Sight and Sound

The source file that you use during the manual noise removal operation has a lot of variation in amplitude since they come from original files from the transcribing operation. At this early stage in the music restoration, you will want to keep the source file intact and not change its amplitude. Since you are creating a temporary destination file that will not be saved for the future; you can change the amplitude of the destination file to help your hearing find where the noise impulse is located for removal. There are two methods that you can use to increase the music volume to aid in *hearing* the noise, the second method will also help in *seeing* the noise. The first method is to increase the music level by *turning up the volume* of the playback amplifier. This is easy to do and often helps in finding the noise events. The second method helps in seeing and hearing the noise by applying a gain change to the destination file. This gain change amplifies the music and noise waveform. The gain can be changed under the CD-Prep option in the toolbars after the destination file is highlighted. The increase in amplitude of the destination file can range up to where the peaks just touch the maximum values (0 dB).

After you have removed all the noise from the source file through the Part One Multifilter and if needed manual methods, you now have your Level One Music File.

Why the Manual Noise Removal Method Occurs at a Specific Step

The manual removal of noise could be performed at any time on a music file. Indeed, you may later hear an annoying noise event in a music file that was in the Level Two folder or even the Restored folder and want to remove the noise at that location. While it is tempting to remove this newly heard noise in a later part of the restoration work sequence, *you should only remove impulse noise using the manual method during the creation of the Level One file*. This sequence of noise removal during the creation of the Level One File is important for many reasons. One is related to the potential use of batch editing.

The sequence that we use for noise removal is structured so that you can return to the Level One file and then re-run the filters to create the Level Two file. The operations during the creation of a Level Two file can often be automated in a *batch* process (under the filter tab) since any noise that needs manual work has been removed. If you remove the noise later than the Level One file you will lose this ability to just re-run the filters in a batch process. The next reason is related to the nature of the response of the filters.

The impulse noise from the record surface can cause a rapid change to the music waveform's shape. Music frequencies range from 20-20,000 Hz and have a relatively smooth shape change to them. This contrasts with the fast shape changes seen with impulse noise. In electrical and mechanical circuits, a fast change applied to a circuit will cause the time domain output response of these circuits to have a *shape* that is related to the type of elements within the circuit. If the circuit contains components that can oscillate then the time domain response will show a *ringing* related to these circuit values. This is analogous to a push on a swing that will then continue to move back and forth for a while.

An example in music would be when a crack occurs in the record and the resulting playback noise impulse shows a ringing as the stylus oscillates for a short time. This ringing can cause the waveform in the source file to have a longer *distortion length* to it than the original impulse itself. This initial ringing will continue into the multfilter with even more ringing with resulting music waveform distortion. For a picture of these effects, refer to figure 8-14.



Figure 8-14 Large Noise Spike

The original impulse *on the record* was modified by an electrical-mechanical circuit (stylus and cartridge) during the conversion from a crack on the record to the music source file. The stylus in the cartridge produced some ringing that starts at marker one location on the upper source file display. This mechanical ringing was then converted to an electrical signal and displayed as a ringing waveform.

The equalization filter in the Part One Multifilter will also show *a ringing* at a frequency related to the circuit elements in the EQ filter. Notice in figure 8-14 how the destination result from use of the EQ in the multifilter (lower waveform) shows the waveform continuing to wiggle until marker two, both graphs use the same time scale. The impulse noise that started at marker one in the source file continued till marker two in the destination file. This extra stretching of the noise event in turn adds distortion to the music for a longer time than just the initial noise event. When the impulse noise is removed in the source file

then the extra distortion shown in the destination file will also be removed when the multifilter is reapplied. The removal of noise in the source file is much more effective than trying to remove the noise event *and* the resulting additional distortion in the destination file.

When you are listening for noise in a section of the music, the ability to hear the noise and see where it is occurring is *easiest* before the shape of the noise has been changed. Therefore, the noise is removed from the *source* file during the application of the Part One Multifilter and not after additional filtering with the Part Two filter. If you waited to remove the noise after several filtering operations have occurred, then your ability to find and remove just the noise is more difficult. After the application of equalization filters and other filtering (CNF) the noise will blend into the desired music waveform and be harder to find and remove.

The use of EQ (Equalization Filter) in the Part One Multifilter during Manual Noise removal is a compromise to help your hearing locate the noise by making the music sound natural.

8.7.4 Creation of a Level Two File

Both Part One and Part Two Multifilters will be applied to create the Level Two File. The previous file in the Level One location will have no impulse noise remaining when the Part One Multifilter is applied to the file. The Part One Multifilter may perform frequency modification to the music due to a possible EQ stage. Before the Part Two Multifilter is applied, listen to the sound of the destination file after the Part One Multifilter is used and change the EQ, if needed, for the best sound.

The Part Two Multifilter may contain several CNF (continuous noise filter) stages having a general type of response that can be applied to music after processing by the Part One section. If continuous noise remains with the general CNF settings, then different settings can be used after taking a noise sample from the beginning of the file.

The sequence of the Level Two operations will be, removal of the impulse noise using the Part One Multifilter, Conversion if needed to 44.1 kHz samples per second, Normalization of the amplitude, Removal of noise that is continuous throughout the entire music using the Part Two Multifilter, and then final editing steps. During the final editing steps the noise before the music starts will be removed and if needed Extra Fade down near the end. The results placed in the Level Two file can be the final stage of the restoration process if you are not going to enhance the music. The subject of enhancing the music is covered in a later section.

Because you have the previous Level One file in your Restoration Files Folder you can always return to the creation of a new Level Two file later for any reason. An example of Level Two File creation follows.

Apply Part One Multifilter

The effort starts by applying the Part One Multifilter to your Level One music file. Refer to Figure 8-15 for an example using the 78 Electric Part One Multifilter:

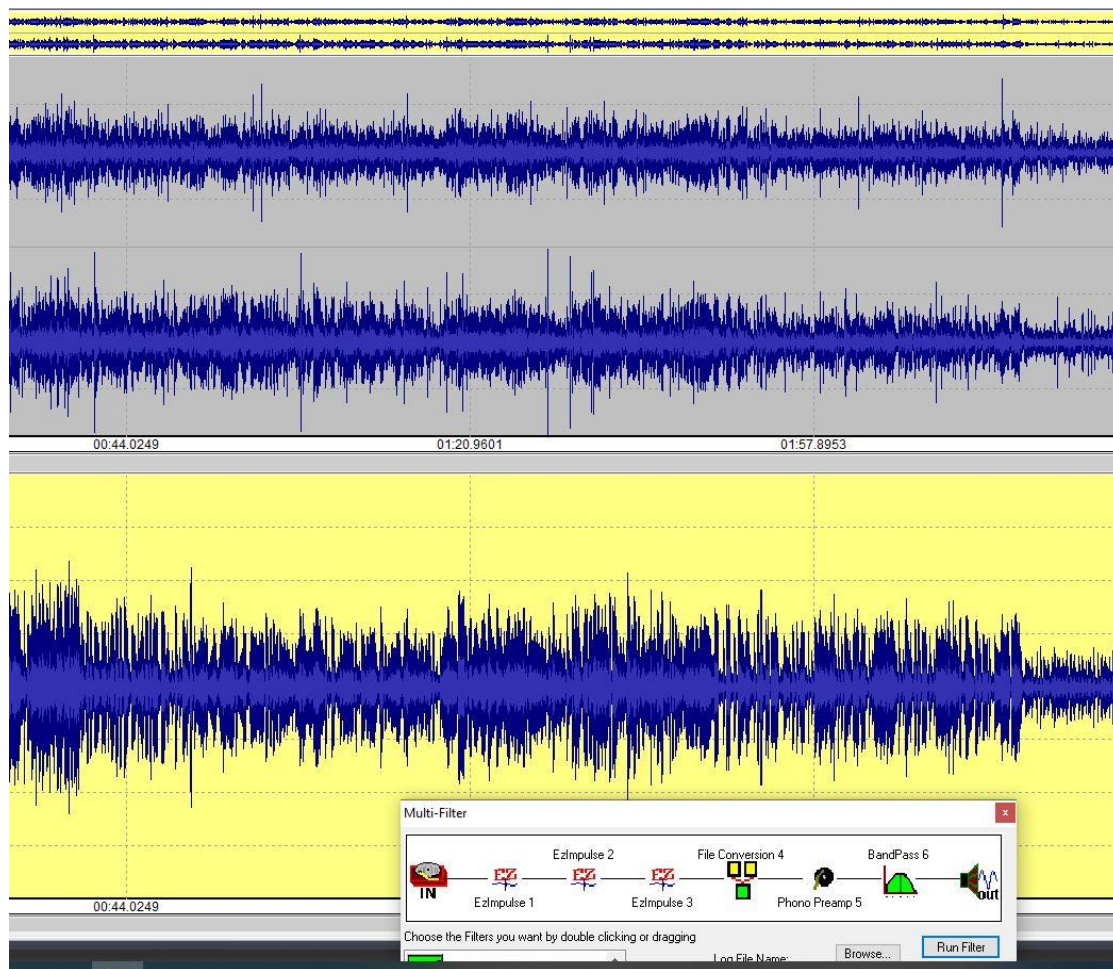


Figure 8-15 Example Multifilter Part One Used

The destination file contains the music with impulse noise removed and continuous noise remaining. As you listen to the destination file verify that the proper EQ was used for the music. If you need to change the EQ for this song then make a temporary modification to the EQ in the Part One Multifilter, delete the previous destination file, and run the filter again

to create the desired result. The destination file will now become the source file for the next effort.

Change Rate & Normalize Amplitude

The previous Part One Multifilter section used a high sample rate of 96 kHz for the music file. This high rate is not needed for the rest of the restoration work and a 44.1 kHz sample rate works well with the CNF to create narrow bin sizes for a given FFT size. The 44.1 kHz sample rate, when used with a digital to analog converter, can accurately produce audio frequencies from 20-20,000 Hz which spans the range of human hearing. Therefore, the sample rate will be changed to 44.1 kHz using the Change Sample Rate Option. Refer to figure 8-16:

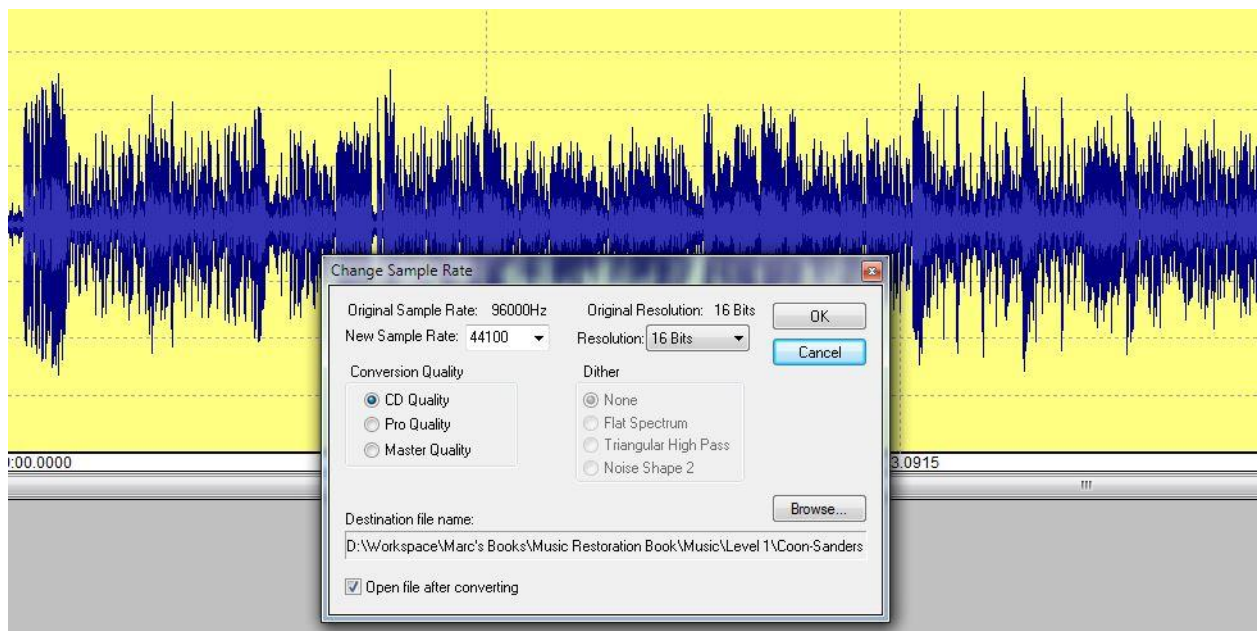


Figure 8-16 Change Sample Rate

After the sample rate has been changed, the file will be the new source if the option to open file after converting is checked, see figure 8-16.

Next, the source amplitude is normalized so that the signal levels are increased to a maximum value of 0.0 dB. There are two methods to normalize under the CD-Prep location. Either one can be used with a setting of 0.0 dB. Refer to figure 8-17 to see the result of the normalization.

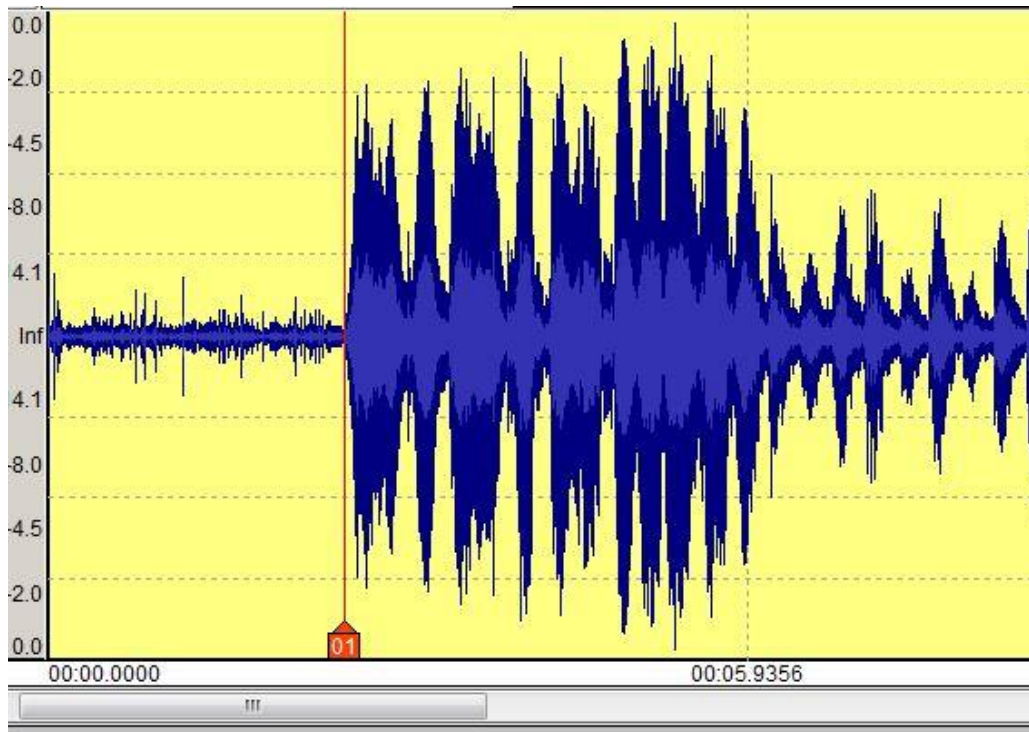


Figure 8-17 Results of Normalization

When the file has been normalized the CNF settings will be useful over several music files without a need to make adjustment for amplitude differences since all the files will have had the same maximum value. The CNF options will be selected *with a normalized music file* in the Part Two Multifilter. For most restorations, the CNF settings with the specific examples later shown will be adequate.

Apply the Part Two Multifilter & Trimming

After the file has been normalized the noise that is continuous will be easily seen. Refer to figure 8-17 to see the noise prior to the start of music at marker 1. If this noise was not present, then the waveform would show a straight line with zero amplitude prior to the start of music. The Multifilter Part Two is now selected and run on the new source file. Refer to figure 8-18 for the results:

If the music still has noticeable continuous noise present after the Part Two Multifilter is used, you can modify the various CNF settings and if needed sample the noise before the music starts to assist in setting the values. Keep in mind that the use of the CNF algorithm

needs careful adjustment so that music is not removed along with the noise. The CNF settings shown later in the chapters are quite mild.

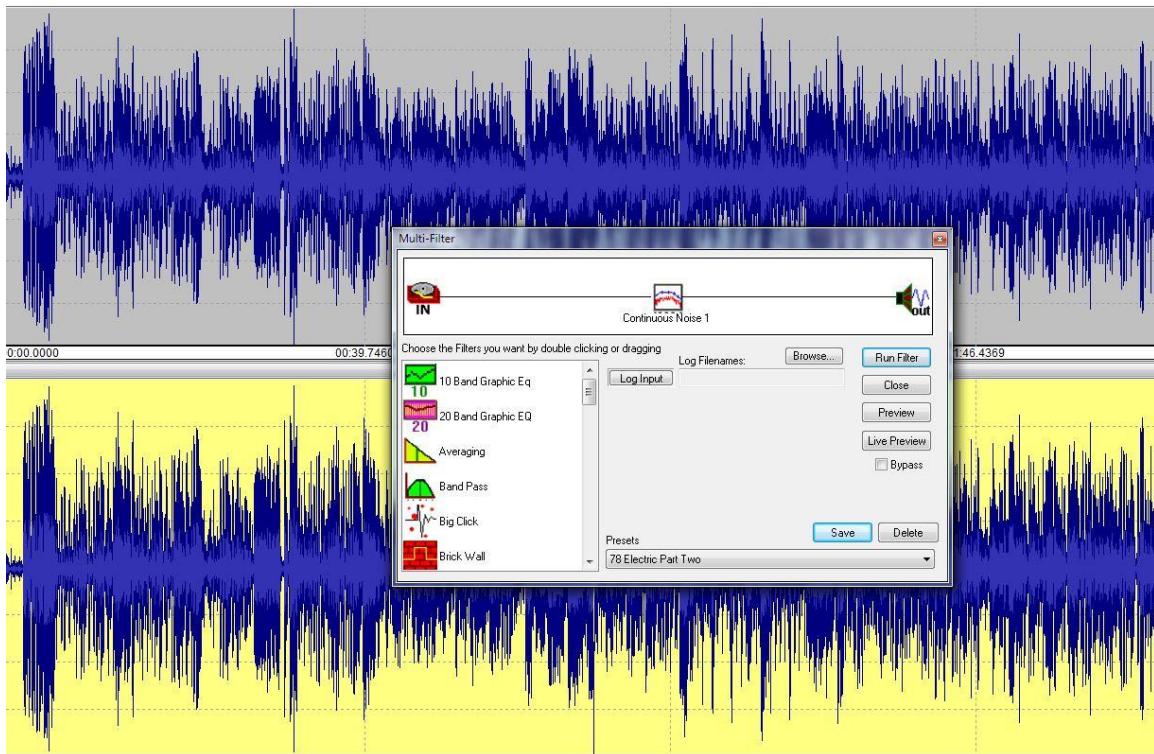


Figure 8-18 Using 78 Electric Part Two Example

The destination file in this example shows a reduction in the continuous noise present before the music starts.

After the 78 Electric Multifilter Part Two has been applied to the source file the destination music file needs one last process: namely the final trimming operation. During this part, the remaining noise prior to the start and end of the music is removed (Using Control & X on a highlighted section works well for removal). The resulting file is now placed into the *Level Two* file location. Refer to figure 8-19 for the final file. If wanted, the fade down from section 8.6 can also be performed to create the final Level Two file.

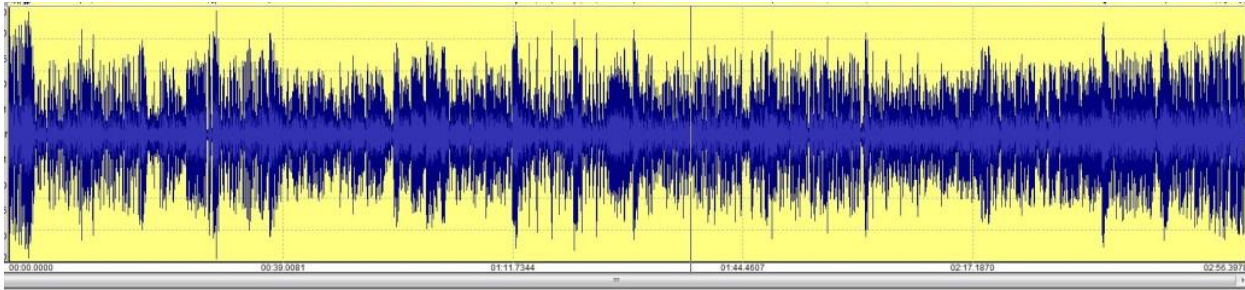


Figure 8-19 Level Two File

If desired, a brief period of silence can be added to the beginning and/or the end of the music using DCart11. This period of silence can be useful when selections of songs are placed on a CD or another device to produce a clean break between songs.

A short one second period of silence before and after the music will be useful for creating both playlists and making CDs of your music. Some of the CD burners provide a method to add the silence. The DC Tunes software will benefit by having a distinct silence section before and after the music as it moves automatically from song to song.

Figure 8-20 shows the DCart11 software option to add silence at either beginning or end of the music. This software is activated only when a file is present in the source location and is found under the Paste option from the main Edit location.



Figure 8-20 Insert Silence Software

8.8 *Specific Multifilters for Records*

Later Chapters will describe Multifilters for different types of records. Each will have some variations in their makeup for the different types of recording methods. For example, the creation of a monoaural file *before* the use of the noise filters works well for vertical recordings, while the conversion to monoaural is best *after* the noise filters for lateral recordings.

The Part Two Multifilter will use one or more CNF devices. These different types were optimized for the type of surface noise on the type of records.

8.9 *Creating a Final Restoration File*

After you have finished with the restoration of the file and it is in the Level Two folder, you may be finished with your work. The same music file in the Level Two folder can be copied into the Restored File location and enjoyed. Later, you may want to add some enhancements to the music or try other effects that are available in the software. These extra steps will use the Level Two file *as a base* for your work and by keeping the Level Two file in its original cleaned up and equalized form you can always return to it if the enhancement work does not work.

8.10 *Method to Keep Track of Your Restoration Work*

As you are creating various Levels you will want to keep track of what files *have been worked on* and *what files need to be worked on*. There is a feature in the DCart11 software that can be used to help keep track of your progress. The method uses the presence of a file that the software creates to display the music, the PKF file, to track your progress.

When the DCart11 software opens a music file the software will create another file with the same name but a different extension. The new file extension has the letters pkf. In figure 8-21, the open software dialog window shows the files before the new Level One or Two files are created. The selection of files to see in this window, is All files (*. *). Note that this setting is not the default for DCart11.

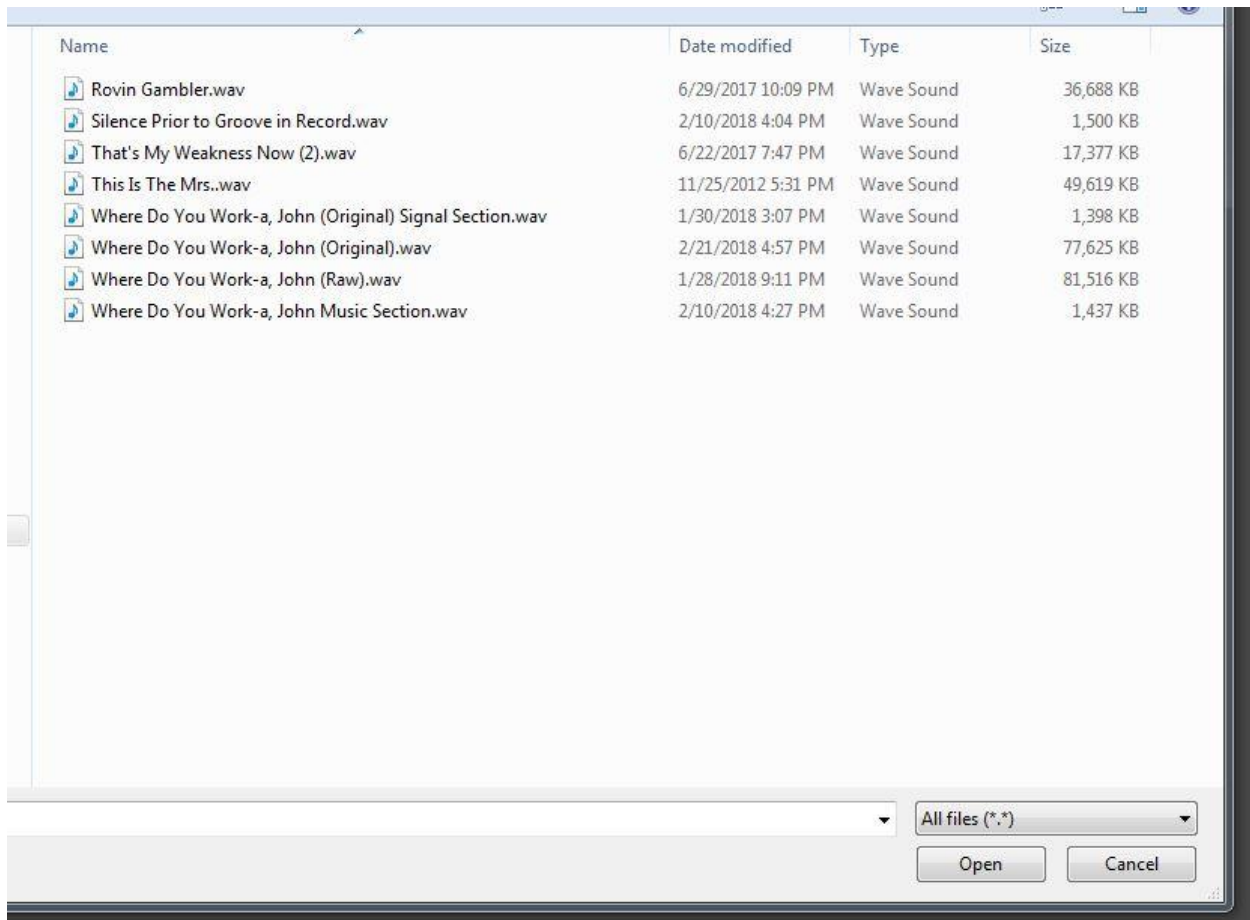


Figure 8-21 Open File Box before a pkf file is generated.

When a file is opened by the software a new file is generated that is used to create the waveform display for that file. The software will store this new file at the same location as the previously opened music file. Refer to figure 8-22 to see *this new file with the pkf extension* for the song Rovin' Gambler after the file was opened:

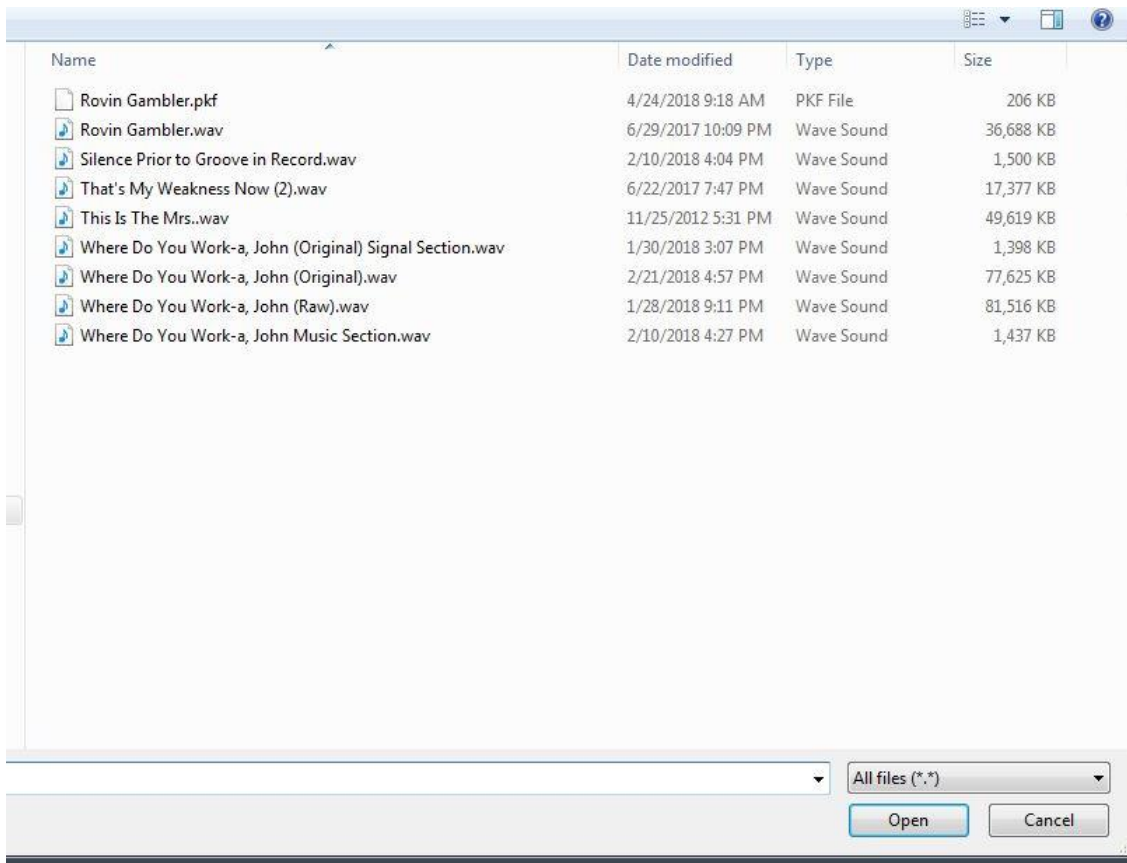


Figure 8-22 Open File Box after PKF has been generated

The DCart11 software will create a corresponding pkf file when a destination file is also saved. The software will not save a pkf file for the destination when it is displayed, *unless* you save the destination file.

When you have completed the noise removal on the original file and it is ready to be a Level One file the presence of the pkf file can remind you that this files work is done. The lack of a pkf file can be used to show you that the file needs noise removed since the pkf file is only created when DCart11 opens a file.

You can modify this method to keep track of which music files have been processed in later stages of your work. The pkf file can be deleted at any time because the DCart11 software will generate a new file, if needed, for a waveform display when you open any file.

DCart11 software will not remove any pkf type files. The software will overwrite a previous file, but will not remove them. You can remove them when you want.

8.11 Summary of Operations

An important sequence of *operations* must occur. Keep in mind that the Levels refer to a major operation while the Part One and Part Two refer to specific Multifilters that are used within the Levels.

This order of noise removal operations will be repeated for each type of record within the Level One and Level Two file generation sections shown in later chapters. The general order of operations for all record types is:

1. Copy the Original File (Flat Recording) and use it as the Working Source File. Check the channel balance and if needed change gain of a channel using the balance meter as shown in section 7.9.1.
2. Apply the Specific Multifilter Part One to the source to create a destination file starting with the values shown for the specific records. If many noise events remain in the destination file, increase the settings until the noise has been removed or the music starts to be degraded. Change the EQ settings (if used) for the desired sound.
3. If needed, manually remove noise that the Part One Multifilter could not remove from the source file, using the Sight and Sound Method.
4. The *Source* file in step 3 is now saved and this file is a new *Level One* file. The previously created destination file is not saved.
5. If noise removal for the working source file was not satisfactory, you can save the source *for another time* as a *Level One A* file.
6. Copy the *Level One* file and use this file as a working Source file.
7. Apply the Part One Specific Multifilter *again* to the Source.
8. Convert the resulting Destination file to a new Source file.
9. Change the 96 kHz sample rate to 44.1 kHz sample rate (or keep at 44.1 kHz if this was originally used for Level One).
10. Normalize the new file to have a maximum value of 0.0 dB.
11. Apply the Part Two Specific Multifilter to the new normalized Source from step 10. If needed, change the CNF settings for more noise removal while preserving the music.
12. Trim away the non-musical section of the music files beginning and end.
13. If needed, add additional fade down to the ending of the music.
14. Save the destination result of step 13 as a *Level Two* music file.

9.0 *78 Acoustic Noise Removal Methods*

This chapter covers the details for removing noise from 78 RPM records that used music power solely to record the record. The term 78 Acoustic refers to these records. The creation of the Level One and Level Two music files will follow the same workflow as described in the chapter General Noise Removal and will not be repeated in detail here. For the 78 Acoustic records the same multifilter for both 44.1 kHz and 96 kHz sampling can be used. An overall workflow is:

9.1 *Level One Creation*

1. Select classic view for the Diamond Cut Software.
2. Copy the desired music file from the original file folder and open the copy as a source file.
3. Check Channel Balance and Adjust if needed using Virtual Phono Preamp.
4. Run the 78 Acoustic Multifilter Part One.
5. If needed, increase the amplitude of the destination file by using the CD-Prep options on the destination file.
6. If many noise events remain slowly increase the EzImpulse settings to remove more noise while checking that music has not been removed.
7. Perform the Sight & Sound manual noise removal method for any remaining noise impulses in the source file.
8. When finished, store the source file as a Level One music file.

9.2 *Level Two Creation*

1. Copy the new Level One music file and open the copy as a source file.
2. Run the 78 Acoustic Multifilter Part One and make the destination into a source file.
3. Apply a Change Sample Rate operation if needed from the initial 96 kHz to 44.1kHz to the source file. Make the result a new source file.
4. Normalize the source file to have a maximum amplitude value of 0dB.
5. Apply the 78 Acoustic Multifilter Part Two to the source file, and make the resulting destination file the new source. If continuous noise remains, carefully increase the CNF settings while checking that desired music is not removed.

6. Trim away the remaining noise before and after the music from the source file. If desired, add extra fade down near the end of the music.
7. When finished, store the resulting file as a new Level Two music file.
8. After the Level Two music file has been created, you can choose to copy the file into the restoration music file location, add silence to the beginning or end, or continue to the Advance Section.

9.3 Details of the 78 Acoustic Multifilter Part One

The 78 Acoustic Multifilter Part One is shown in figure 9-1:

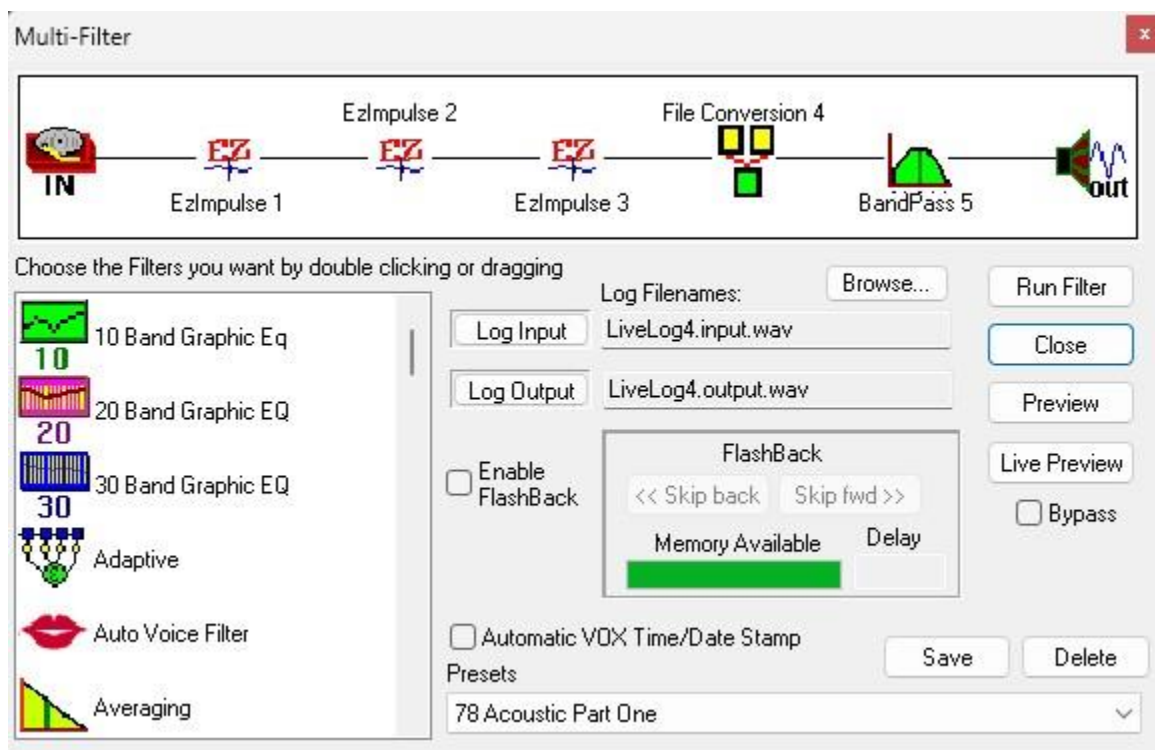


Figure 9-1 78 Acoustic Part One Multifilter

The filter is composed of five parts that are detailed below with their settings. A progression of Impulse Filters helps to find and remove noise.

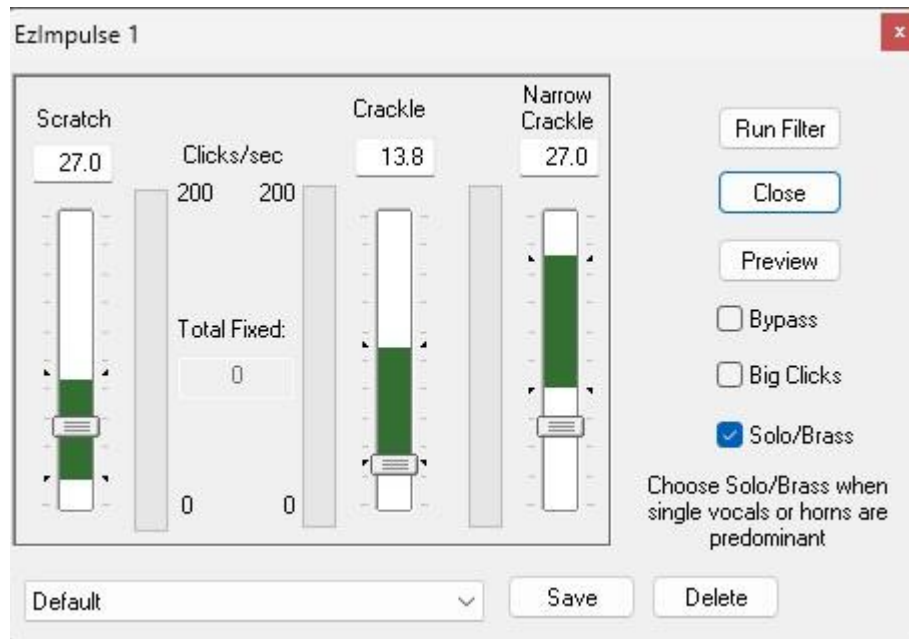


Figure 9-2 EzImpulse Filter (1) 78 Acoustic Multifilter Part One

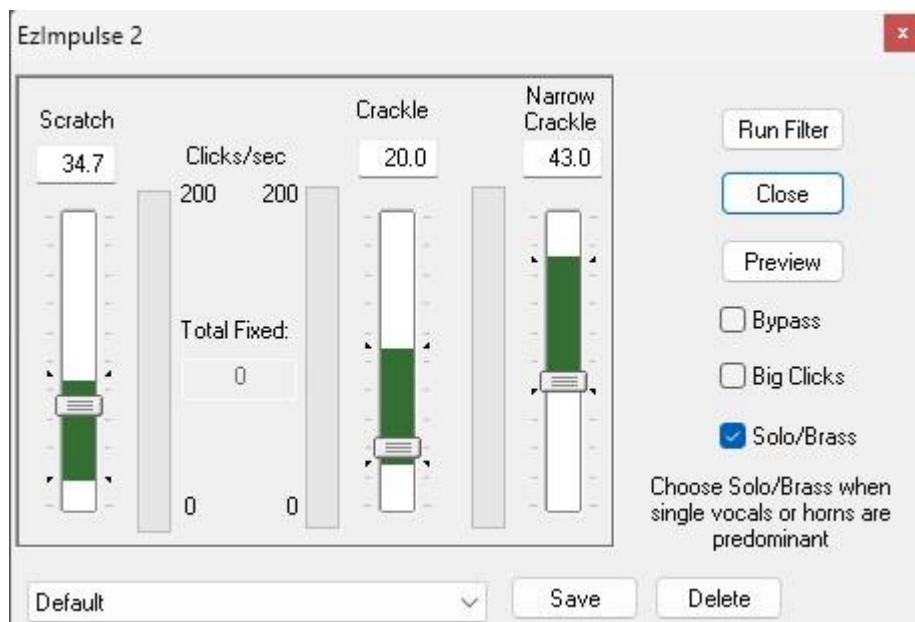


Figure 9-3 EzImpulse Filter (2) 78 Acoustic Multifilter Part One

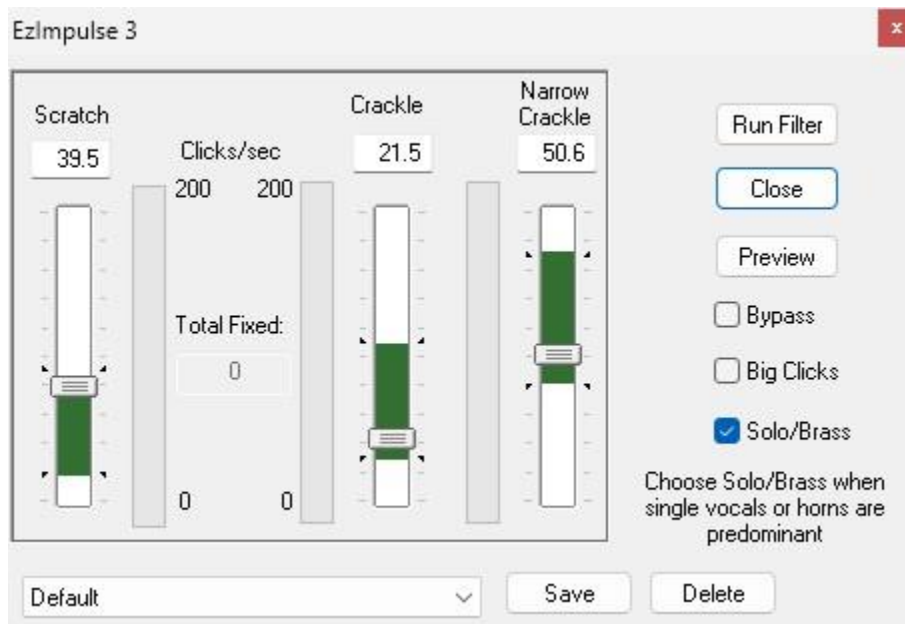


Figure 9-4 EzImpulse Filter (3) 78 Acoustic Multifilter Part One

Note that the various settings are on the low side and the Solo/Brass option is on.

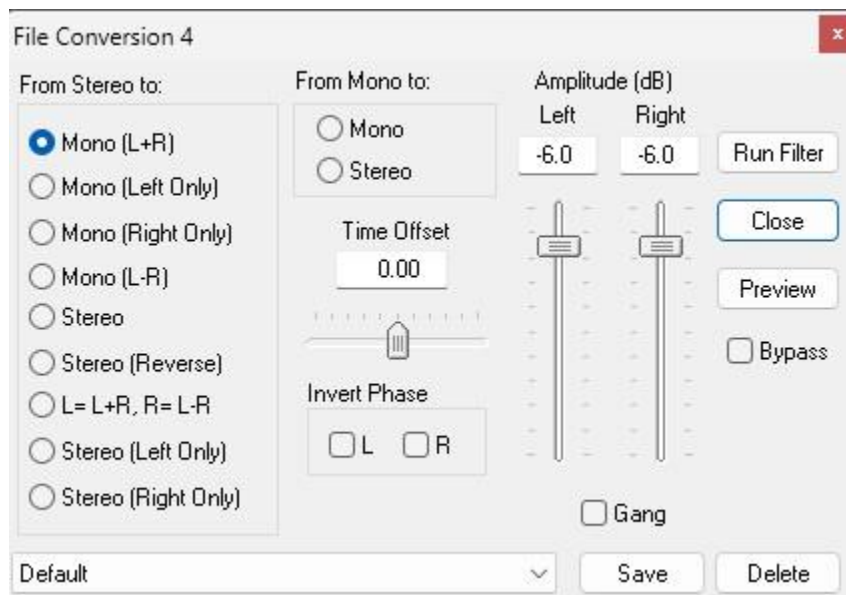


Figure 9-5 File Conversion Filter (4) 78 Acoustic Multifilter Part One

The two separate channels are converted to a monaural file at this point. When the two stereo files are converted to a monaural file the original stereo 45-degree motion that contains

horizontal and vertical information can be processed to remove either type. Thus, when the two channels are added together the vertical component is removed and the horizontal remains. The reason that the file conversion is performed after the initial noise is removed is that the shape of the noise is greatly modified after the two channels are combined and as such, the filters work better when they see the original shape of the noise as it directly comes from the record, rather than after it has been converted to a monaural signal.

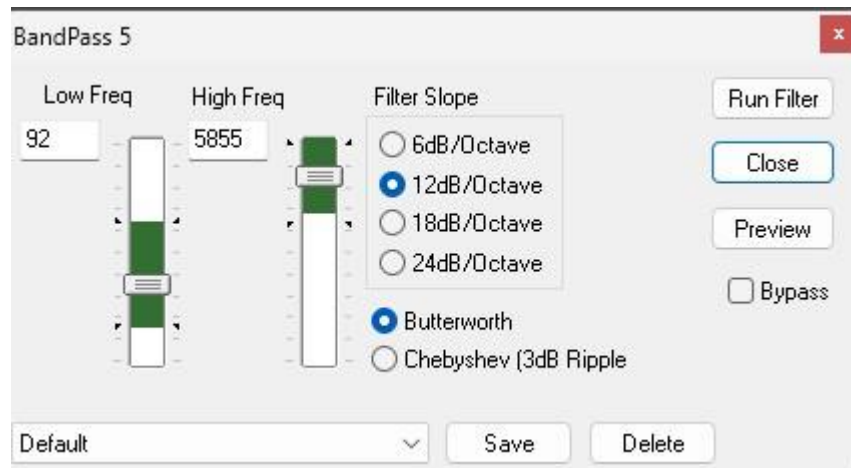


Figure 9-6 Bandpass Filter (5) 78 Acoustic Multifilter Part One

The last filter limits the overall bandwidth of the music. The limits are very broad at this point since additional limiting will be performed later during Part two. Some of the very early records have frequencies higher than often stated and a lower value may needlessly remove music.

9.4 *Details of the 78 Acoustic Multifilter Part Two*

The 78 Acoustic Multifilter Part Two is shown in figure 9-7:

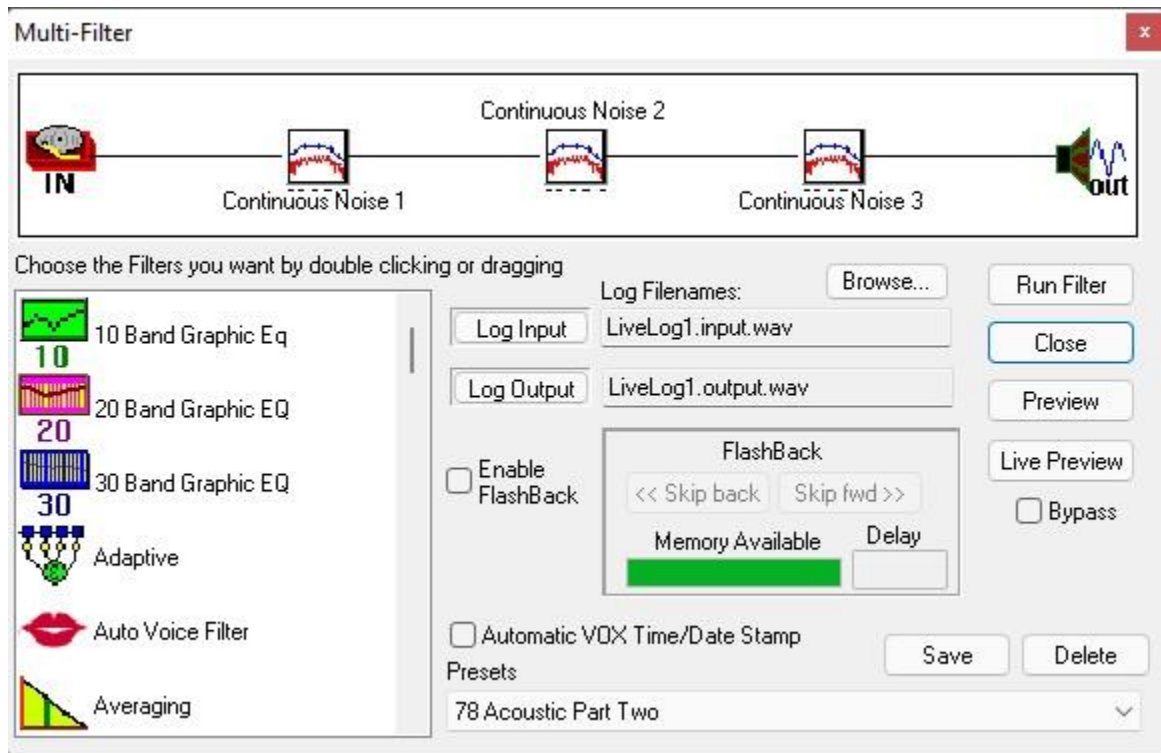


Figure 9-7 78 Acoustic Multifilter Part Two

The multifilter part two contains three CNF sections. The use of three CNF sections helps to remove some of the high frequency noise that the acoustic records have versus the electric recordings. The recording engineers did not remove some of the high and low frequency noise since the playback equipment at the time had a very limited response. With the current playback systems used today extra music frequencies can be heard and using three CNF sections helps to bring out the hidden music. The details of each CNF are shown in figures 9-8, 9-9, and 9-10 below:

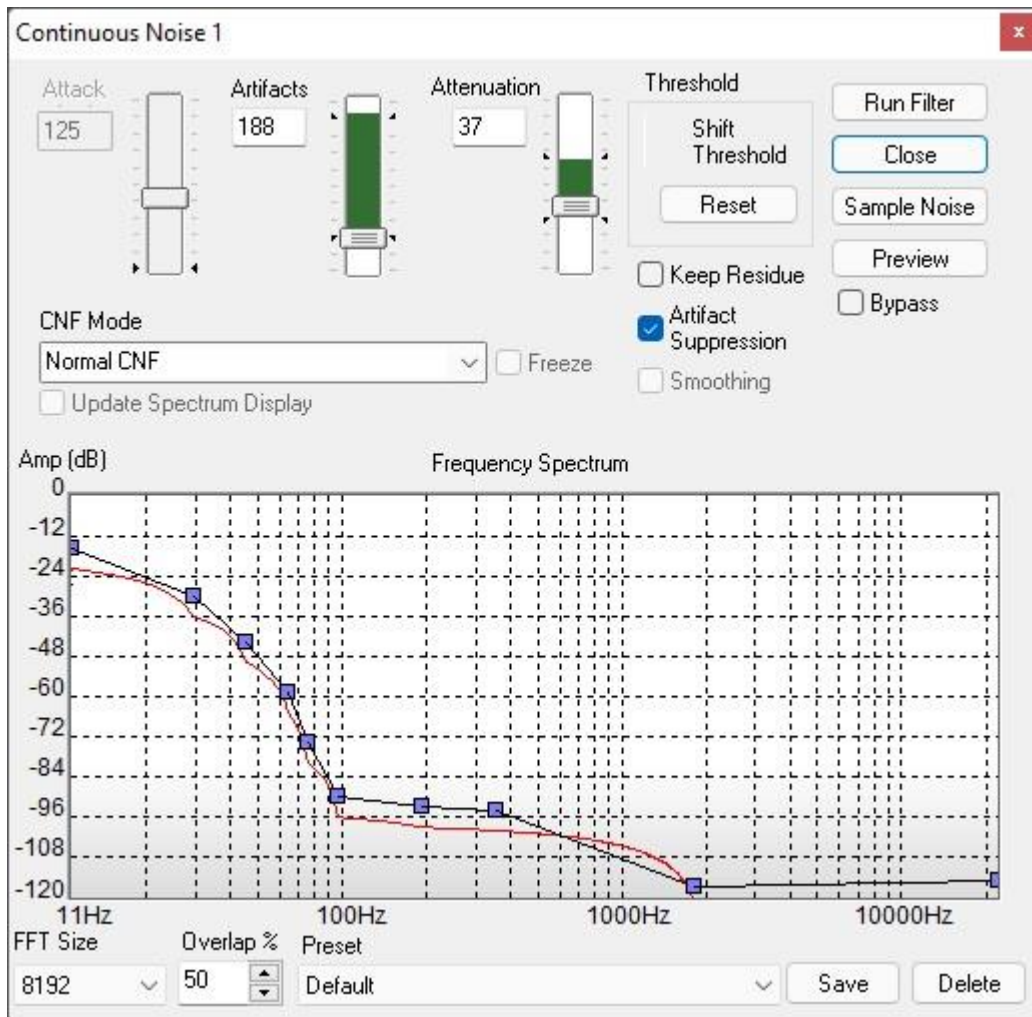


Figure 9-8 CNF (1) 78 Acoustic Multifilter Part Two

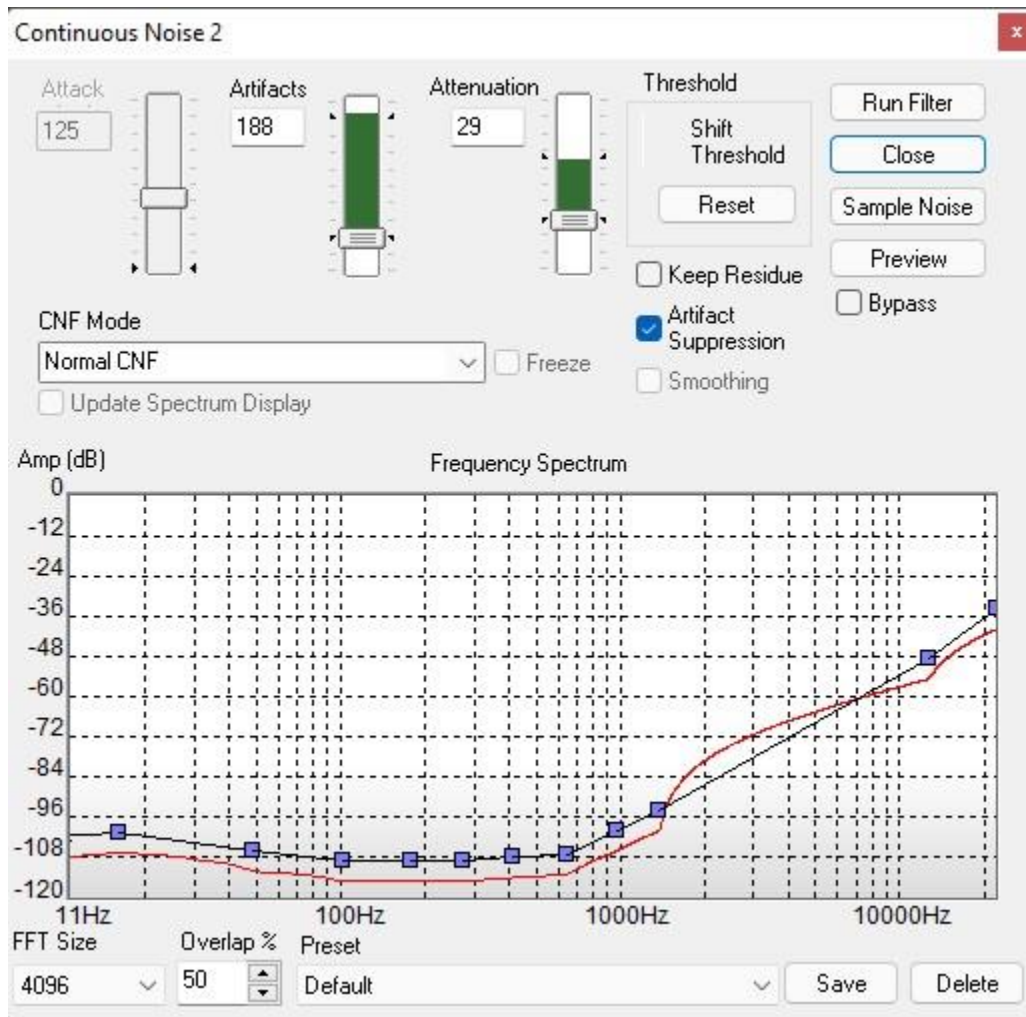


Figure 9-9 CNF (2) 78 Acoustic Multifilter Part Two

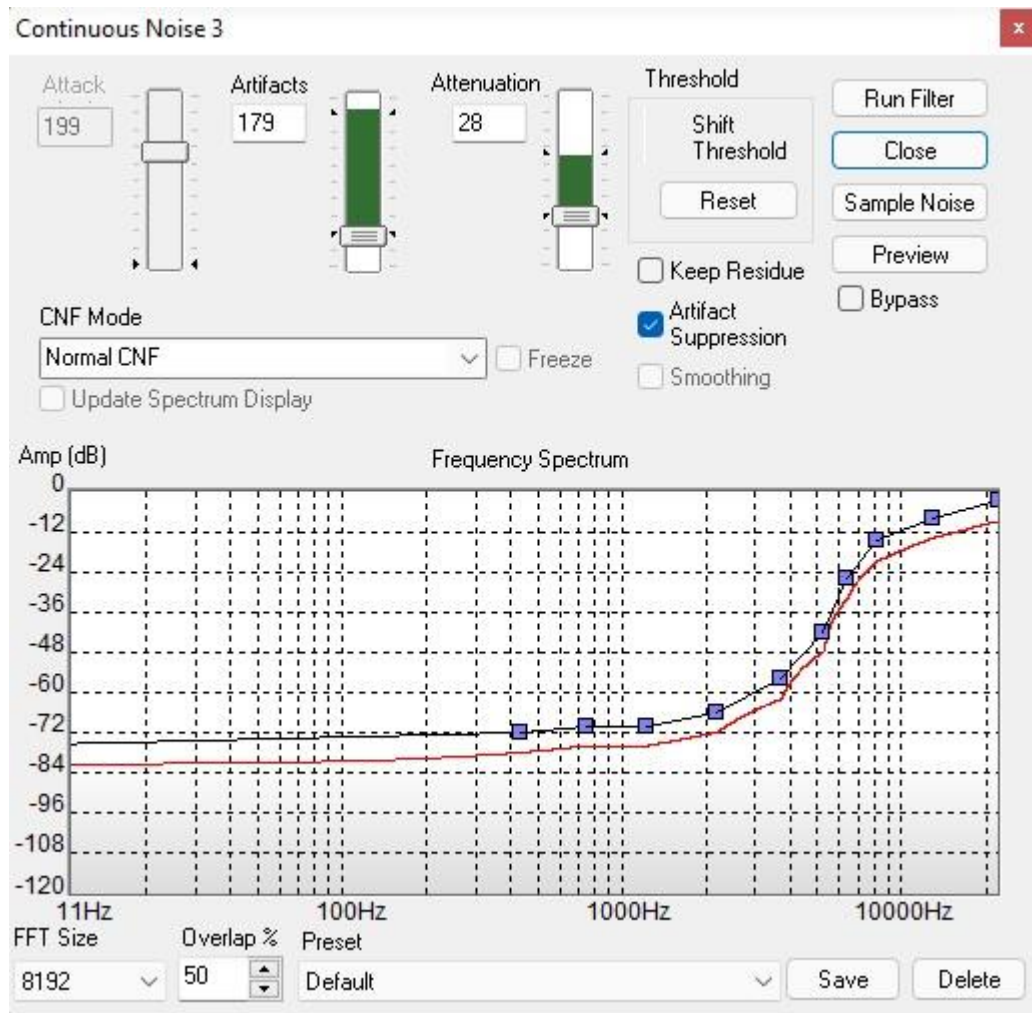


Figure 9-10 CNF (3) 78 Acoustic Multifilter Part Two

This completes the noise removal process for many 78 Acoustic Records that were in average condition. The CNF filters shown in figures 9-8, 9-9, and 9-10 have mild settings that remove mainly the higher frequencies and keep the important harmonics for many instruments.

10.0 78 Electric Noise Removal Methods

This chapter covers the details of removing the noise from 78 RPM records that used electrical recording methods to amplify the music energy. The term 78 Electric will also refer to these records.

10.1 Important Note about 78 RPM EQ Values

The DCart11 software can *remove* one specific type of record EQ, namely RIAA, while it can *apply* many types; the specific EQ that you use for a 78 RPM record can be a good estimate at this point in the restoration process and you will not be able to remove it after you create the Level Two file.

A good method for a starting point is to pick the Phono Preamp American 78 setting as the *default* setting. The 78 Electric Multifilter in the following text will show that default setting but if you want to change this to a specific one for the records, a change can be performed to this EQ filter in the Multifilter. The American 78 setting provides a turnover value of 500 Hz versus the European 78 setting that provides a turnover value of 250 Hz.

The 78 Electric records have a large variation in turnover values and some experimentation is often required.

Because of the file structure with progressive noise removal if the best EQ for a 78 RPM record must be changed after you have produced a Level Two file you can always return to the Level One Stage and modify the EQ you used to create a new Level Two file and save yourself any noise removal work that you did to produce the Level One file.

The Part One Multifilter will be used with either 96 kHz samples per second or 44.1 kHz samples per second.

The creation of the Level One and Level Two music files will follow the same workflow as described in the chapter General Noise Removal and will not be repeated in detail here.

10.2 Level One Creation

1. Select classic view for the Diamond Cut Software.
2. Copy the desired music file from the original file folder and open the copy as a source file.

3. Check Channel Balance and Adjust if needed using Virtual Phono Preamp.
4. Run the 78 Electric Multifilter Part One.
5. If needed, increase the amplitude of the destination file by using the CD-Prep options on the destination file.
6. If many noise events remain slowly increase the EzImpulse settings to remove more noise while checking that music has not been removed.
7. Perform the Sight & Sound manual noise removal method for any remaining noise impulses in the source file.
8. When finished, store the source file as a Level One music file.

10.3 Level Two Creation

1. Copy the new Level One music file and open the copy as a source file.
2. Run the 78 Electric Multifilter Part One and make the destination into a source file while changing, if needed, the EQ value in either the Paragraphic or Phono Preamp.
3. Apply a Change Sample Rate operation if needed from the initial 96 kHz to 44.1 kHz to the source file. Make the result a new source file.
4. Normalize the source file to have a maximum amplitude of 0dB.
5. Apply the 78 Electric Multifilter Part Two to the source file make the resulting destination file the new source. If continuous noise remains, carefully increase the CNF settings while checking that desired music is not removed.
6. Trim away the remaining noise before and after the music from the source file. If desired, add extra fade down near the end of the music.
7. When finished, store the resulting file as a new Level Two music file.

After the Level Two music file has been created, you can choose to copy the file into the restoration music file location, add one second of silence to the beginning and end, or continue to the Advance Section.

10.4 Details of the 78 Electric Multifilter Part One

The 78 Electric Multifilter Part One is shown in figure 10-1:

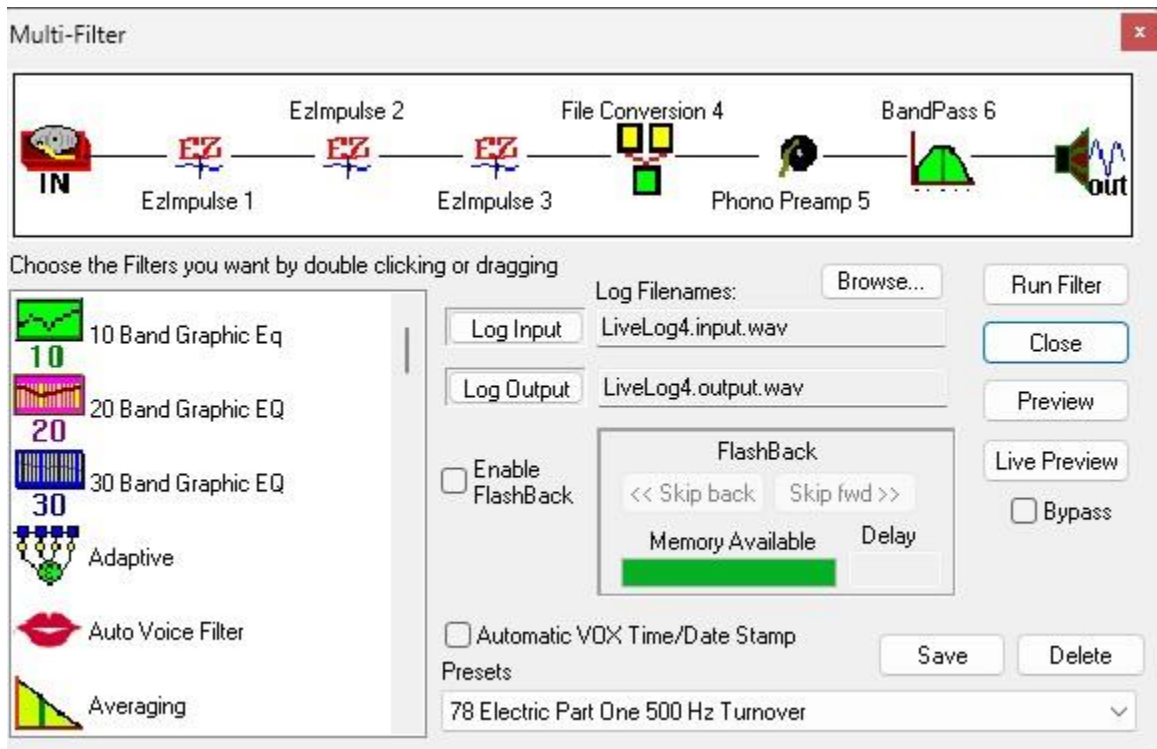


Figure 10-1 78 Electric Part One Multifilter

The filter is composed of six parts that are detailed below with their settings. A progression of Impulse Filters helps to find and remove noise.

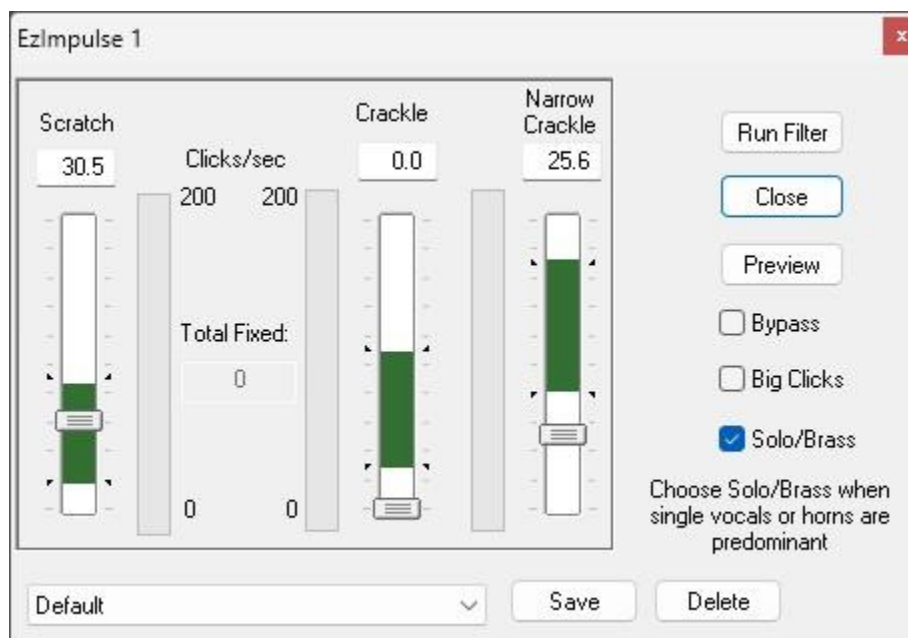


Figure 10-2 EzImpulse Filter (1) 78 Electric Multifilter Part One

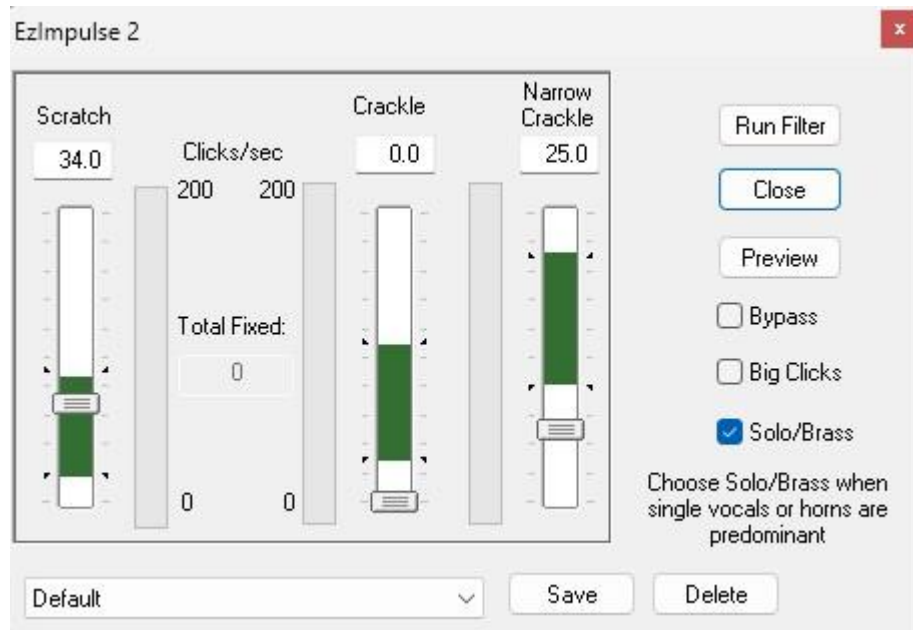


Figure 10-3 EzImpulse Filter (2) 78 Electric Multifilter Part One

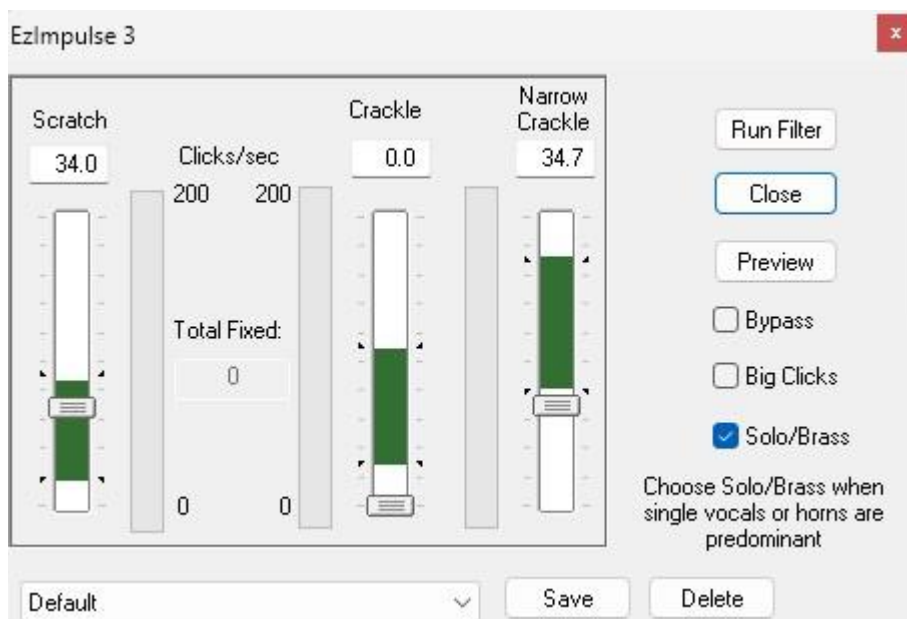


Figure 10-4 EzImpulse Filter (3) 78 Electric Multifilter Part One

Note that the various settings are on the low side and the Solo/Brass option is on. The Crackle setting is at the lowest value as this filter seems to detect brass music as noise with any setting.

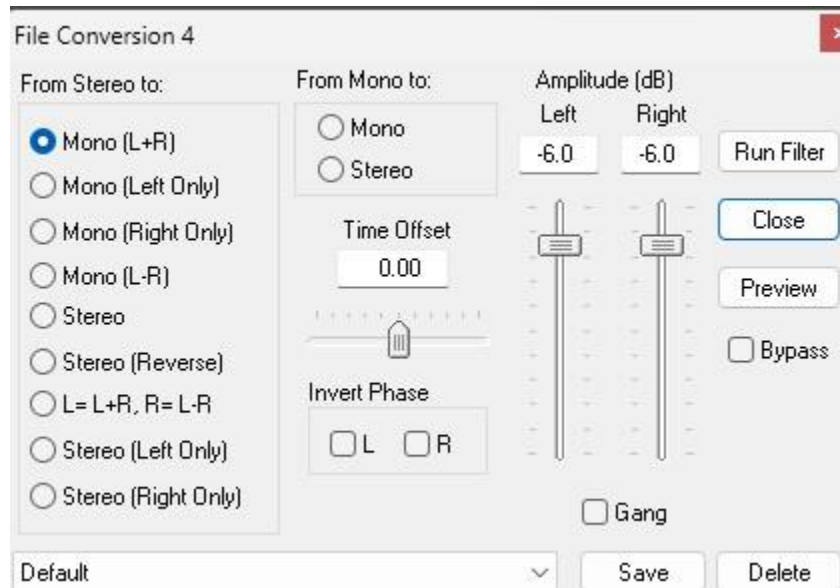


Figure 10-5 File Conversion Filter (4) 78 Electric Multiferter Part One

Note that the two separate channels are converted to a monaural file at this point. When the two stereo files are converted to a monaural file the original stereo 45-degree motion that contains horizontal and vertical information can be processed to remove either type. When the two channels are added together the vertical noise components are removed and the horizontal music remains. The reason that the file conversion is performed after the initial noise is removed, is that the shape of the noise is greatly modified after the two channels are combined and the filters seem to work better when they see the original shape of the noise as it directly comes from the record rather than after it has been converted to a monaural signal.

The next filter contains the conversion from a flat recording to a proper EQ. If you want to use a different EQ filter than shown this would be the place to use it in figure 10-6, by either changing the settings in the Phono Preamp or removing the Phono Preamp Filter and replacing it with the desired setting in the Paragraphic EQ filter.

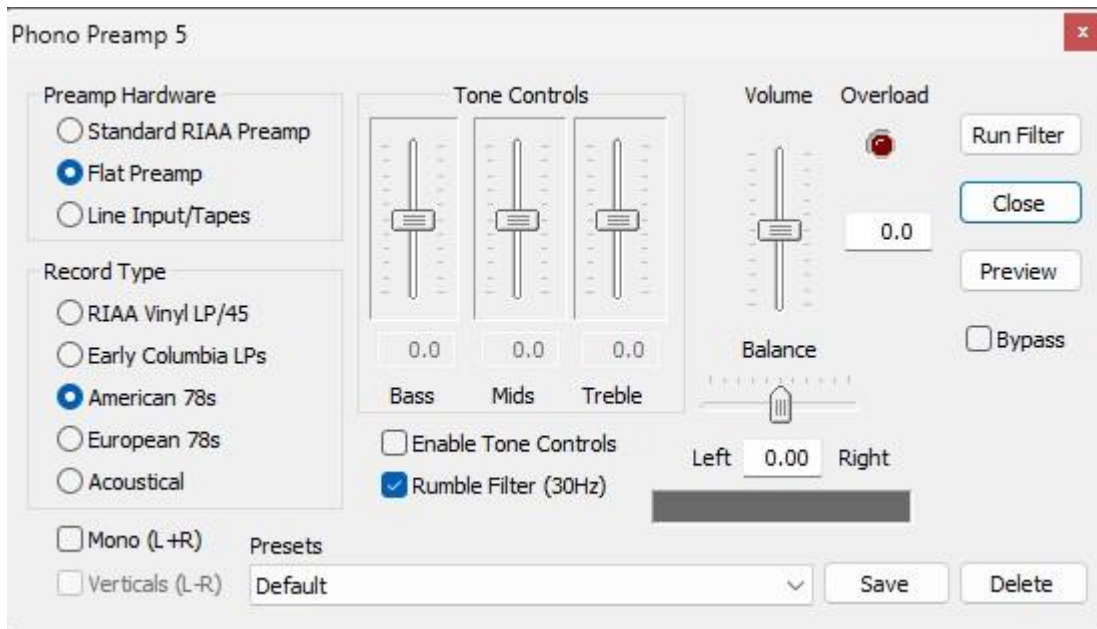


Figure 10-6 Phono Preamp Filter (5) 78 Electric Multifilter Part One

The last filter limits the overall bandwidth of the music. The limits are very broad at this point since additional limiting will be performed later during Part two. Some of the early records have frequencies much higher than often stated and a lower value may needlessly remove music. This filter is in figure 10-7:

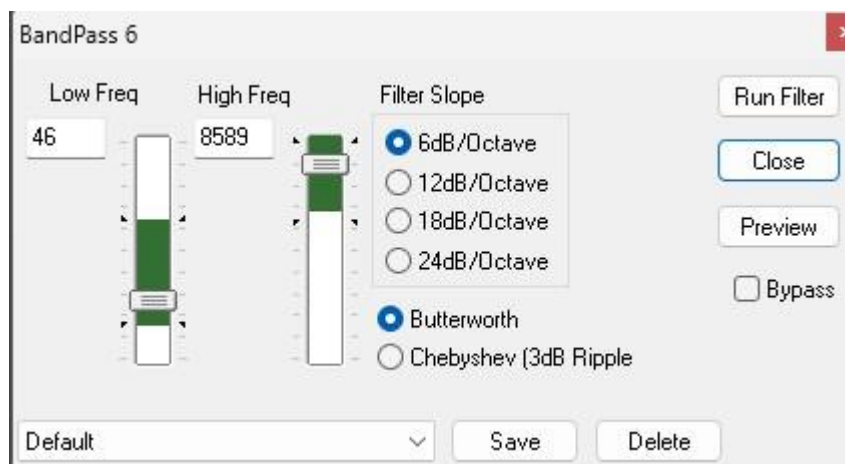


Figure 10-7 Bandpass Filter (6) Multifilter Part One

10.5 Details of the 78 Electric Multifilter Part Two

The 78 Electric Multifilter Part Two is shown in figure 10-5:

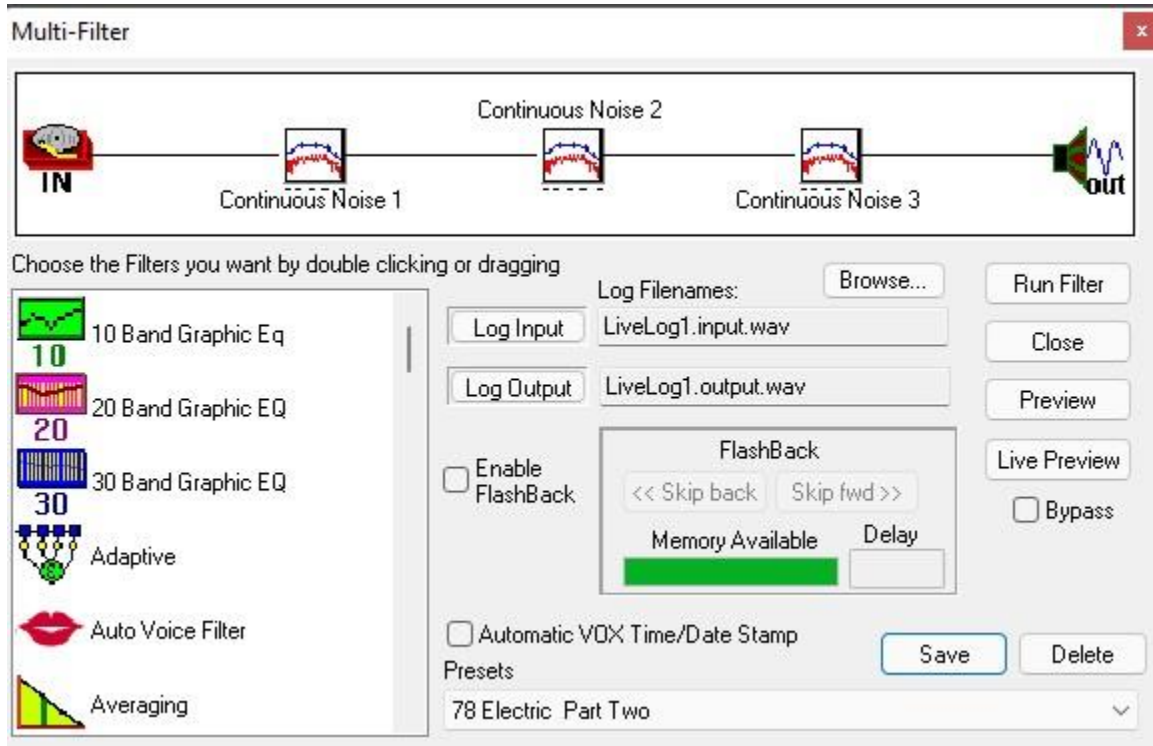


Figure 10-8 78 Electric Multifilter Part Two

The multifilter part two contains three CNF sections. The details of the CNFs are shown in figure 10-9, 10-10, and 10-11:

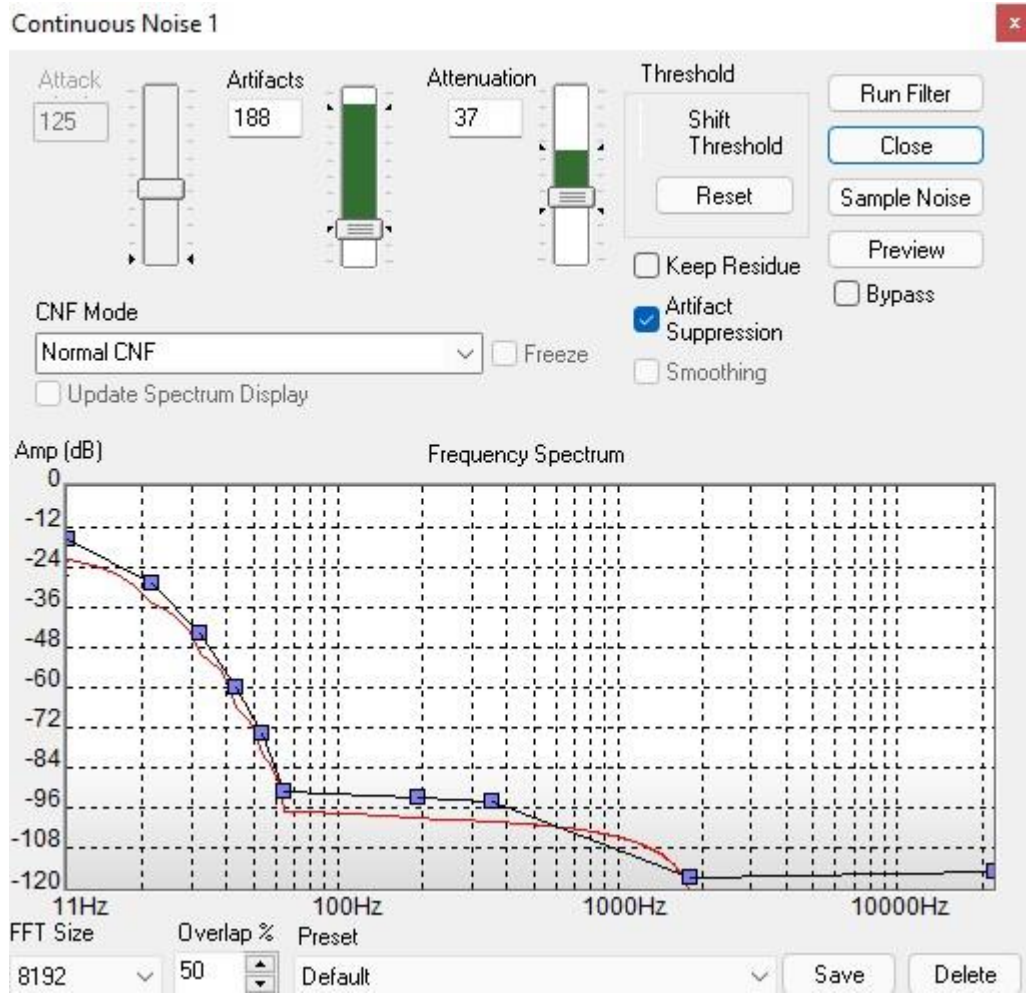


Figure 10-9 CNF (1) 78 Electric Multifilter Part Two

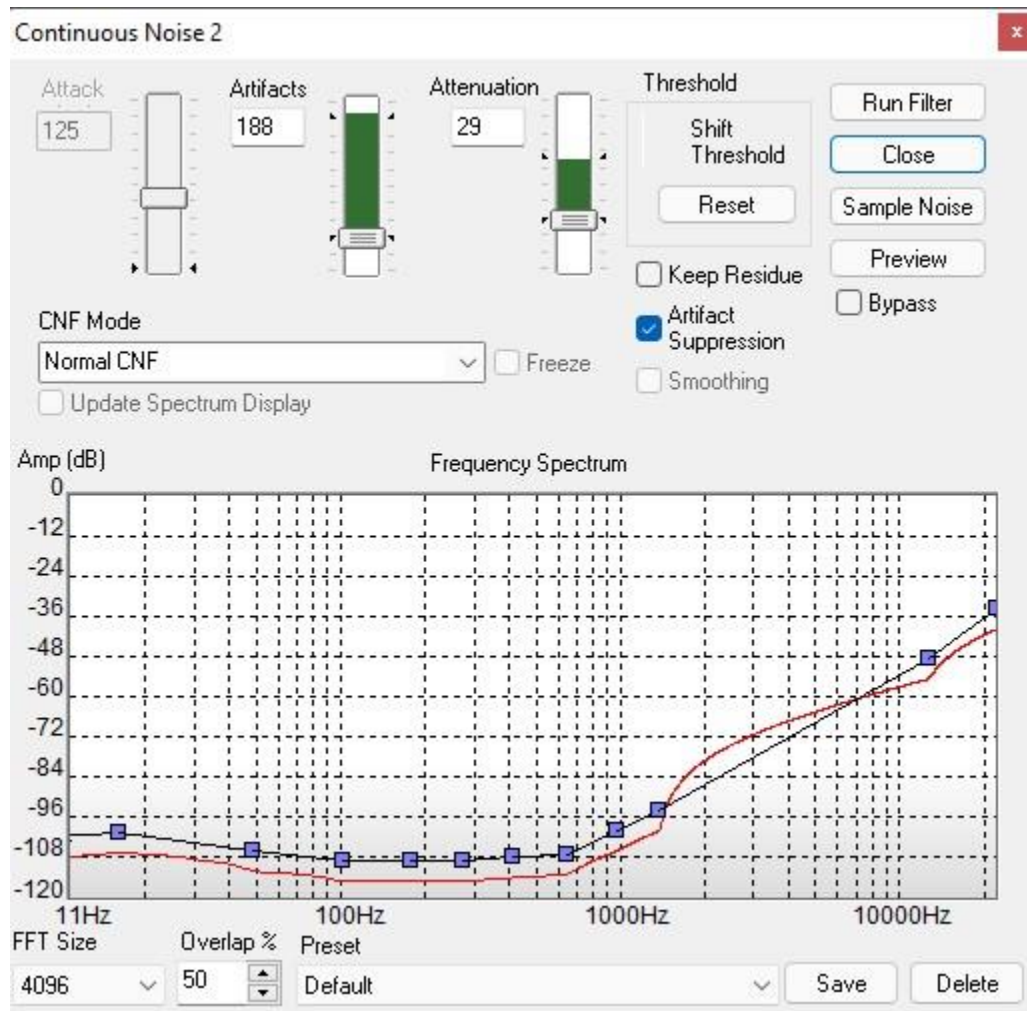


Figure 10-10 CNF (2) 78 Electric Multifilter Part Two

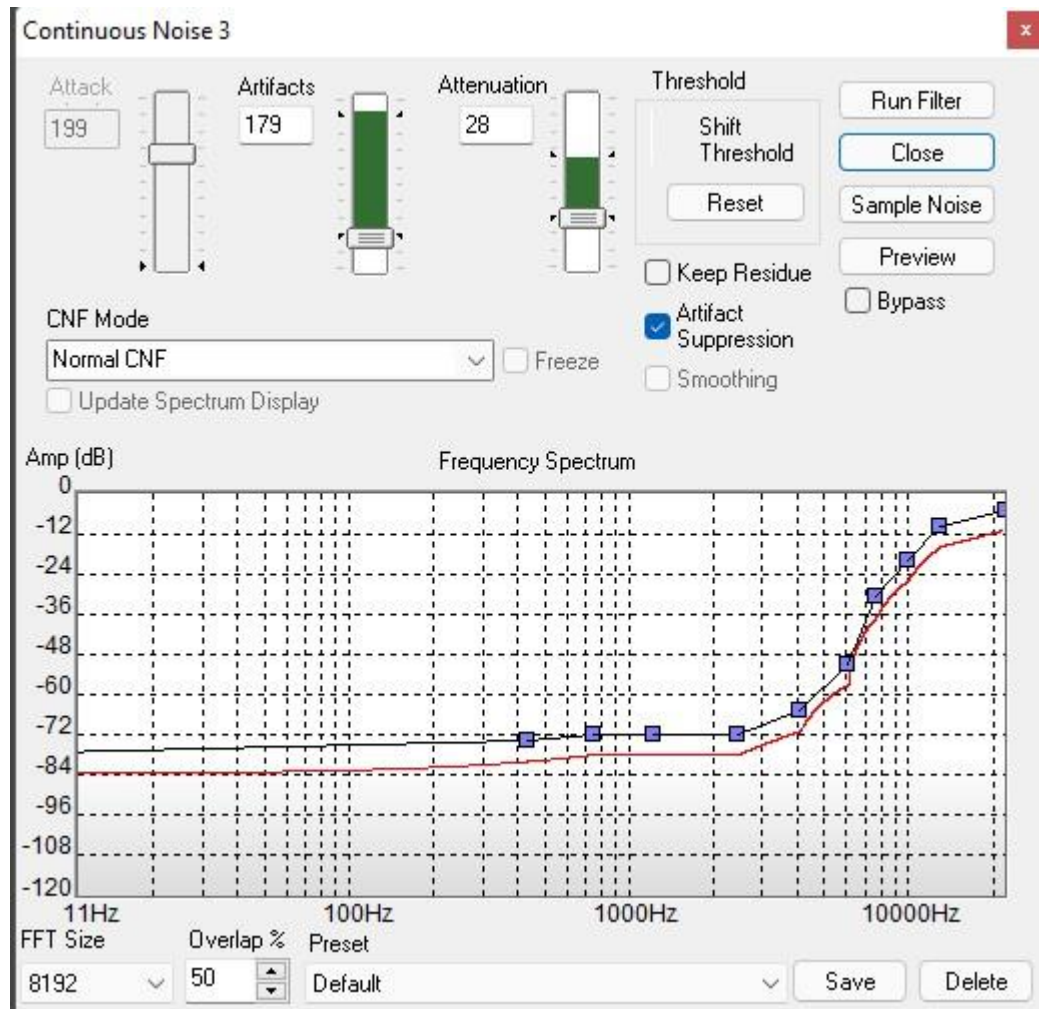


Figure 10-11 CNF (3) 78 Electric Multifigure Part Two

This completes the noise removal process for many 78 Electric Records that were in average condition. The CNF filters have mild settings that remove the noise frequencies and keep the important harmonics for many instruments.

11.0 *Diamond Disc Noise Removal Methods*

This chapter will cover the details of removing the noise from Edison Diamond Disc records. This method will be used for both Acoustic and Electric recorded records. For Electric recorded records, frequency modification can be used as described in the Enhancement Chapter.

For the Diamond Disc Records one Multifilter for Part one will be used for either a 44.1 kHz or 96 kHz sampling rate. The general design of the Multifilter will have the two channels combined prior to the noise removal as this method works best for vertical recording with a small, 0.7 mil radius stylus. The location of the conversion to monaural may benefit from *after* the Impulse filters *if a larger stylus is used*.

The creation of the Level One and Level Two music files will follow the same workflow as described in the chapter General Noise Removal and will not be repeated in detail here. An overall workflow is:

11.1 *Level One Creation*

1. Select classic view for the Diamond Cut Software.
2. Copy the desired music file from the original file folder and open the copy as a source file.
3. Check Channel Balance and Adjust if needed using Virtual Phono Preamp.
4. Run the Diamond Disc Acoustic Multifilter Part One.
5. If needed, increase the amplitude of the destination file by using the CD-Prep options on the destination file.
6. If many noise events remain slowly increase the EzImpulse settings to remove more noise while checking that music has not been removed.
7. Perform the Sight & Sound manual noise removal method for any remaining noise impulses in the source file.
8. When finished, store the source file as a Level One music file.

11.2 Level Two Creation

1. Copy the new Level One music file and open the copy as a source file.
2. Run the Diamond Disc Multifilter Part One and make the destination into a new source file.
3. Apply a Change Sample Rate operation if needed from the initial 96 kHz to 44.1 kHz to the source file. Make the result a new source file
4. Normalize the source file to have the maximum amplitude a value of 0dB.
5. Apply the Diamond Disc Acoustic Multifilter Part Two to the source file, and make the resulting destination file the new source. If continuous noise remains, carefully increase the CNF settings while checking that desired music is not removed.
6. Trim away the remaining noise before and after the music from the source file. If desired, add extra fade down near the end of the music.
7. When finished, store the resulting file as a new Level Two music file.

After the Level Two music file has been created, you can choose to copy the file into the restoration music file location, add one second of silence to the beginning and end, or continue to the enhancement chapter.

11.3 Details of the Diamond Disc Multifilter Part One.

The Diamond Disc Part One is shown in figure 11-1:

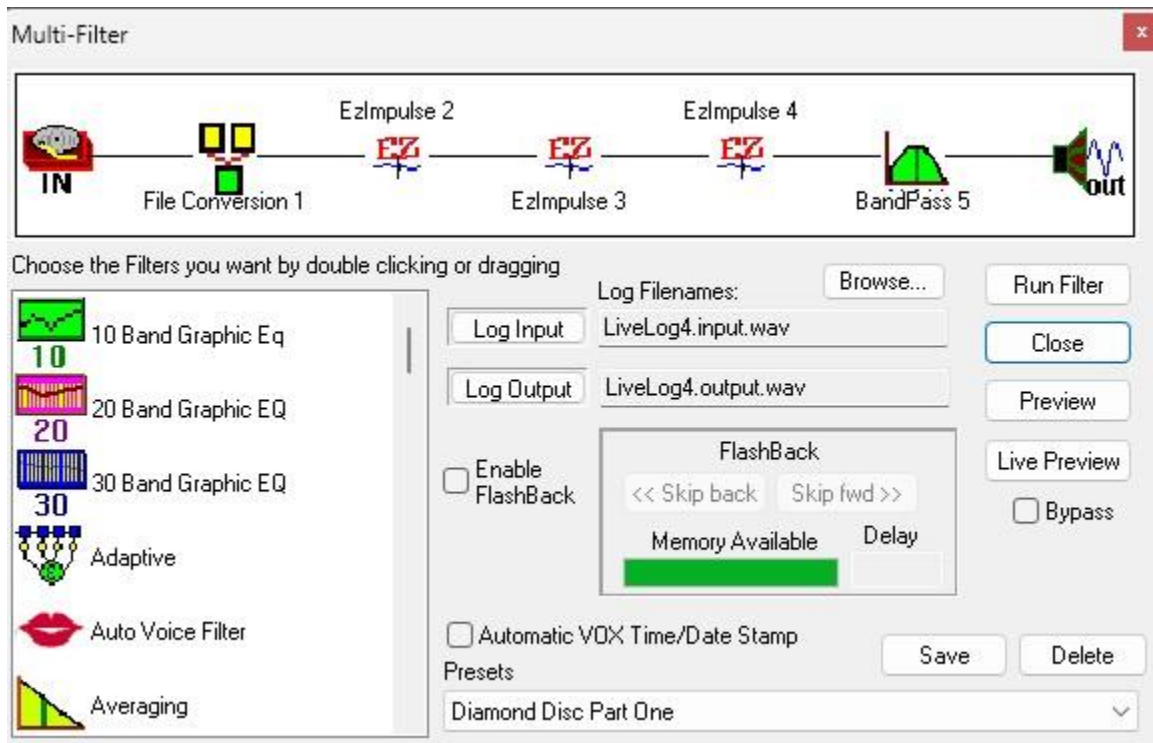


Figure 11-1 Diamond Disc Multifilter Part One

The filter is composed of five parts that are detailed below with their settings. A progression of Impulse Filters helps to find and remove noise.

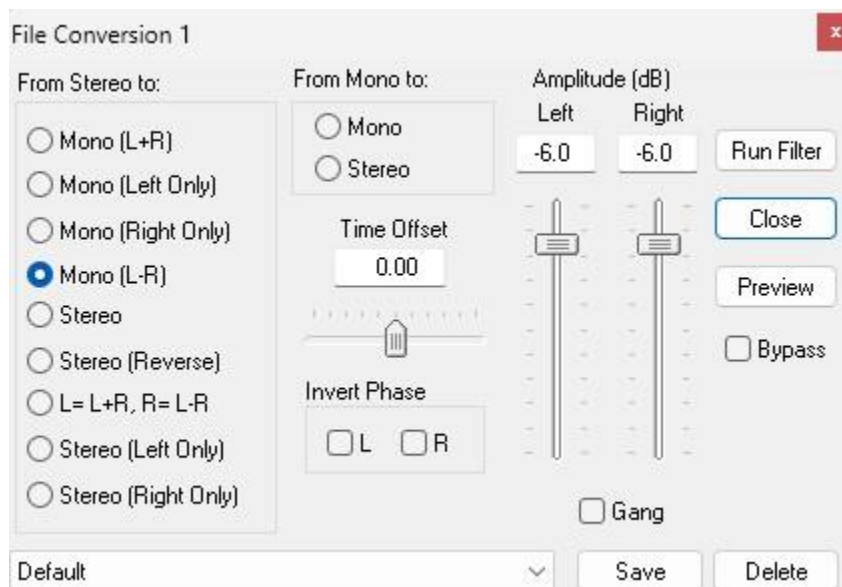


Figure 11-2 File Conversion (1) Diamond Disc Multifilter Part One

Note that the two separate channels are converted to a monaural file at this point. When the two stereo files are converted to a monaural file the original stereo 45-degree motion that contains horizontal and vertical information can be processed to remove either type. When the right channel is subtracted from the left channel the vertical component remains and the horizontal is removed.

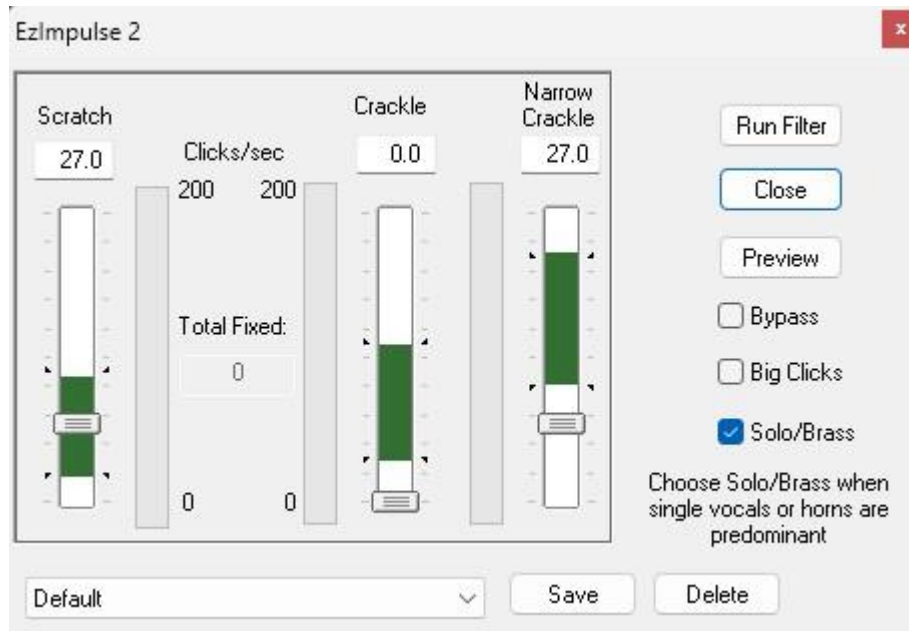


Figure 11-3 EzImpulse Filter (2) Diamond Disc Multifilter Part One

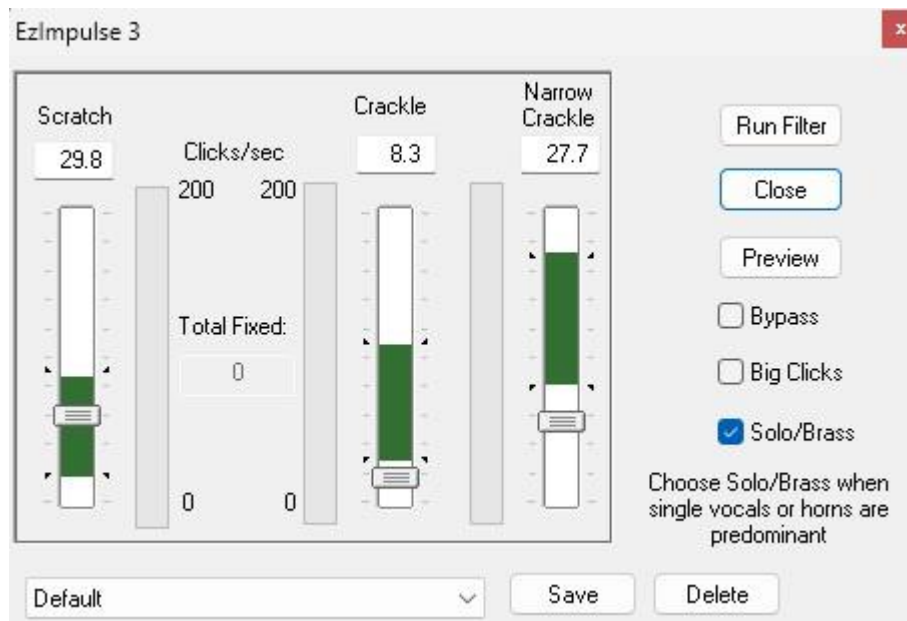


Figure 11-4 EzImpulse Filter (3) Diamond Disc Multifilter Part One

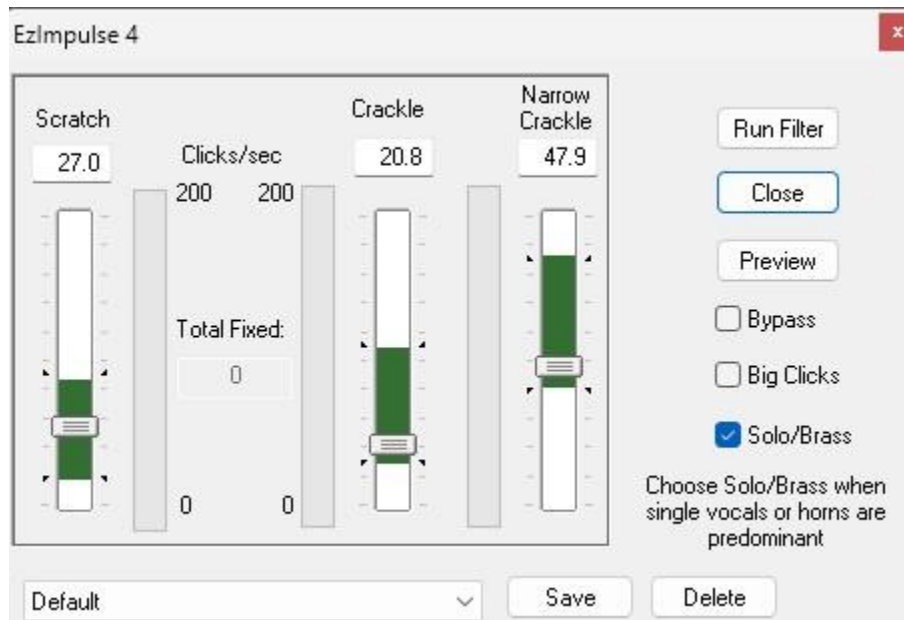


Figure 11-5 EzImpulse Filter (4) Diamond Disc Multifilter Part One

Note that the various settings are on the low side and the Solo/Brass option is on.

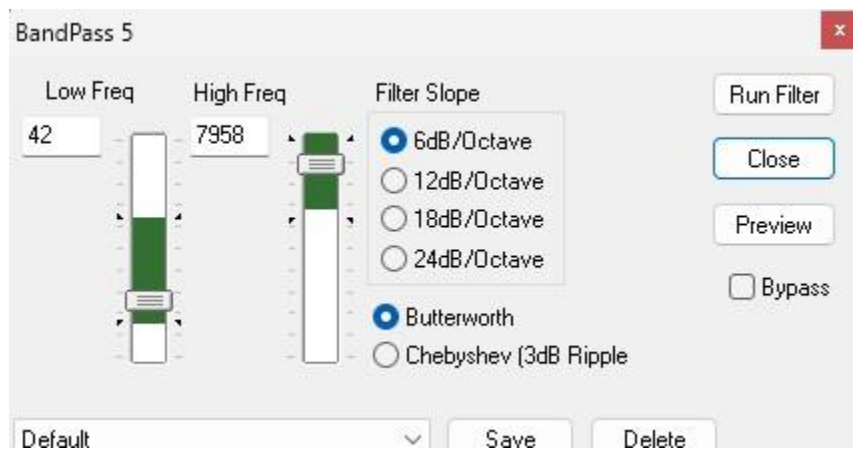


Figure 11-6 Bandpass (5) Diamond Disc Multifilter Part One

The last filter limits the overall bandwidth of the music. The limits are very broad at this point since additional limiting can be performed during Part two. Some of the early records have frequencies much higher than often stated and a lower value may needlessly remove music.

11.4 Details of the Diamond Disc Multifilter Part Two

The Diamond Disc Part Two is shown in figure 11-8 and is the same for either 96 kHz or 44.1 kHz sample rate since all Level One files are converted to 44.1 kHz:

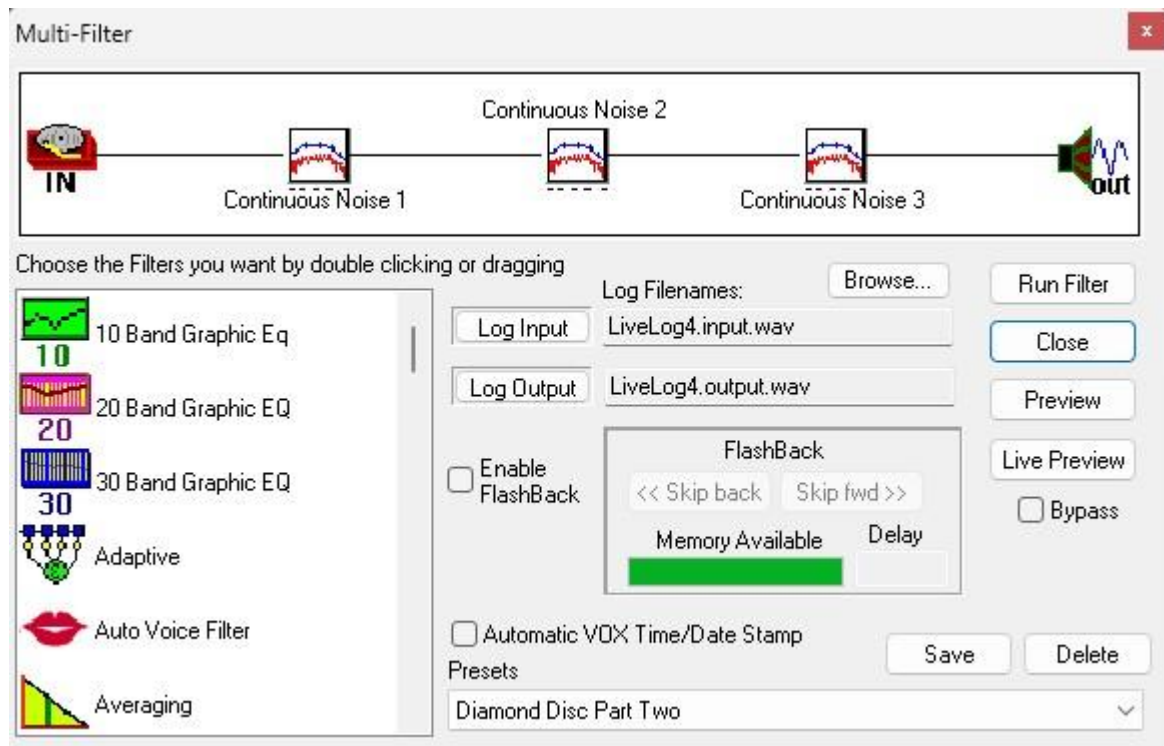


Figure 11-8 Diamond Disc Part Two

The multifilter part two contains three CNF sections. The use of three CNF sections helps to remove some of the high frequency noise that the acoustic records have versus the electric recordings. The recording engineers did not remove some of the high and low frequency noise since the playback equipment at the time had a very limited response. With the current playback systems today extra music frequencies can be heard and using three CNF sections helps to bring out the hidden music. Refer to figures 11-9, 11-10, and 11-11 for the details.

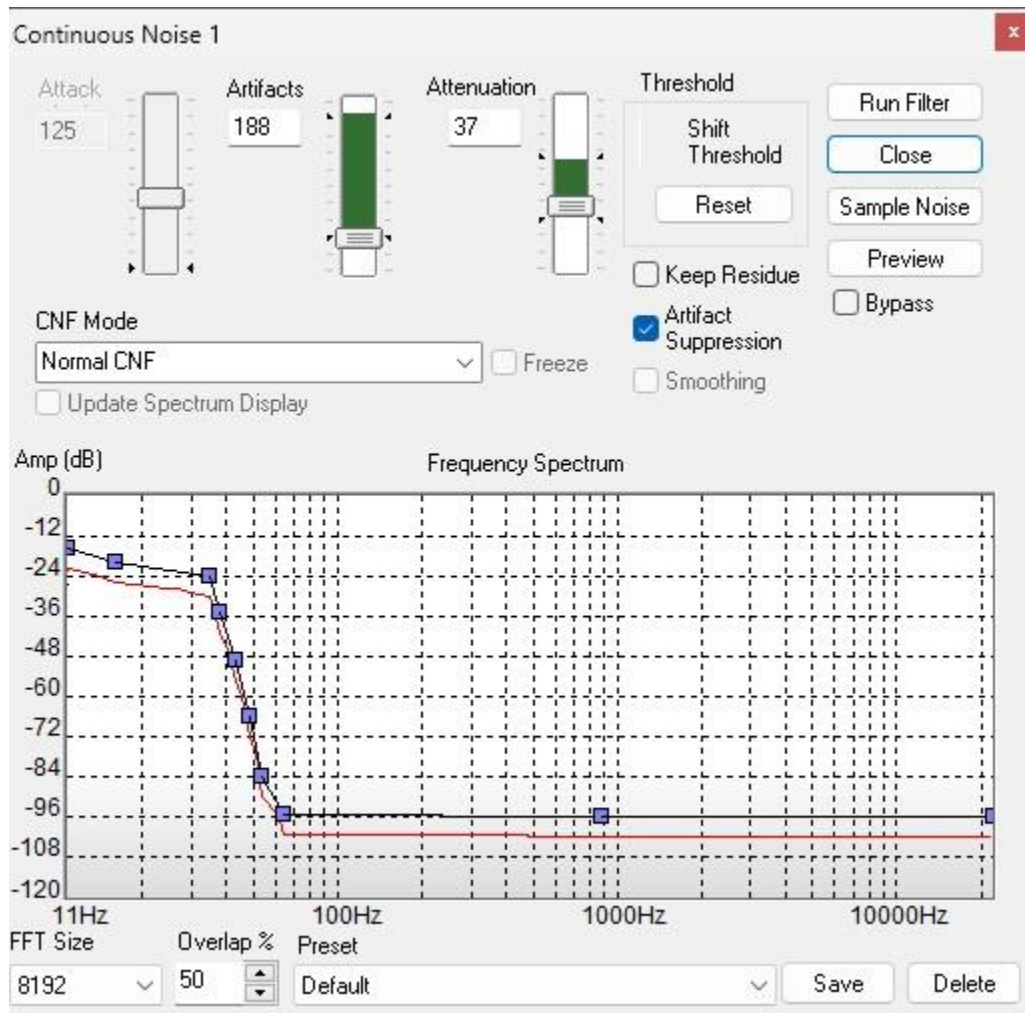


Figure 11-9 CNF (1) Diamond Disc Multifilter Part two

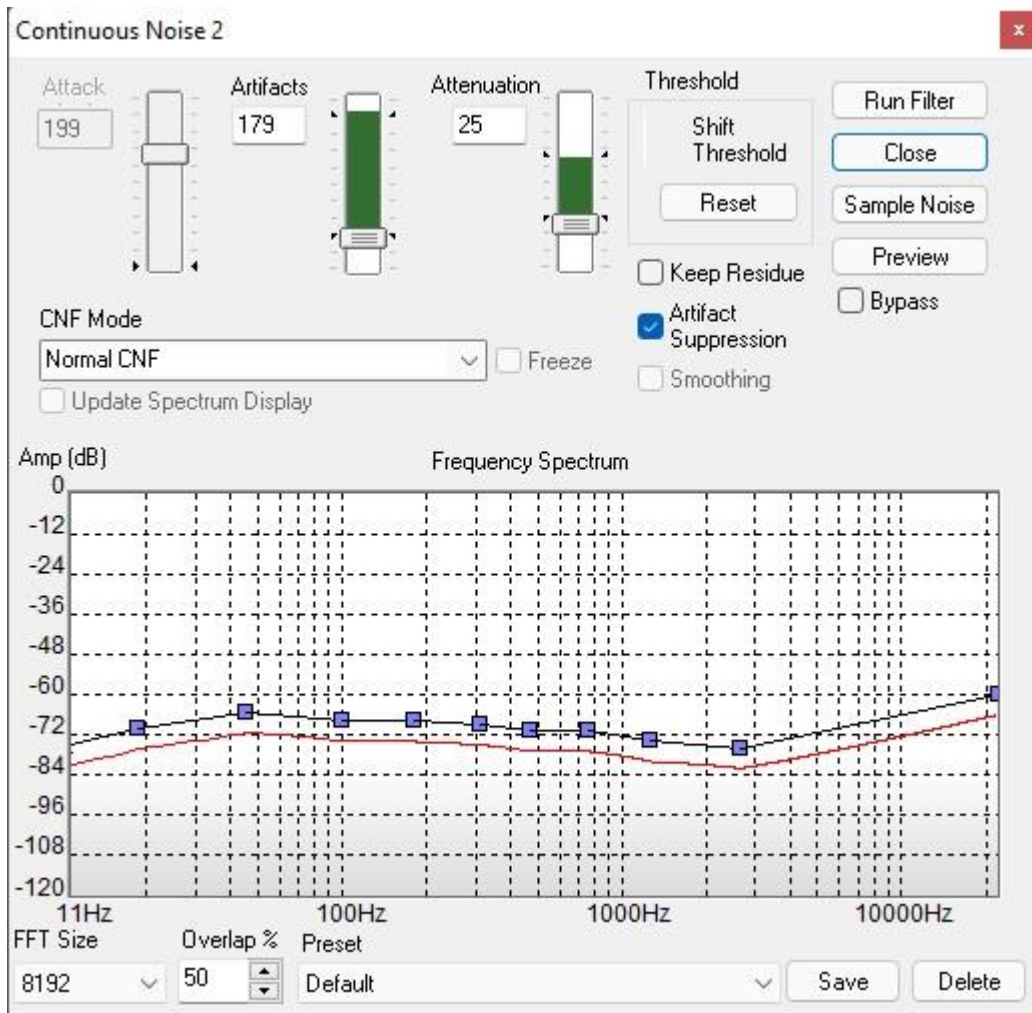


Figure 11-10 CNF (2) Diamond Disc Multifilter Part Two

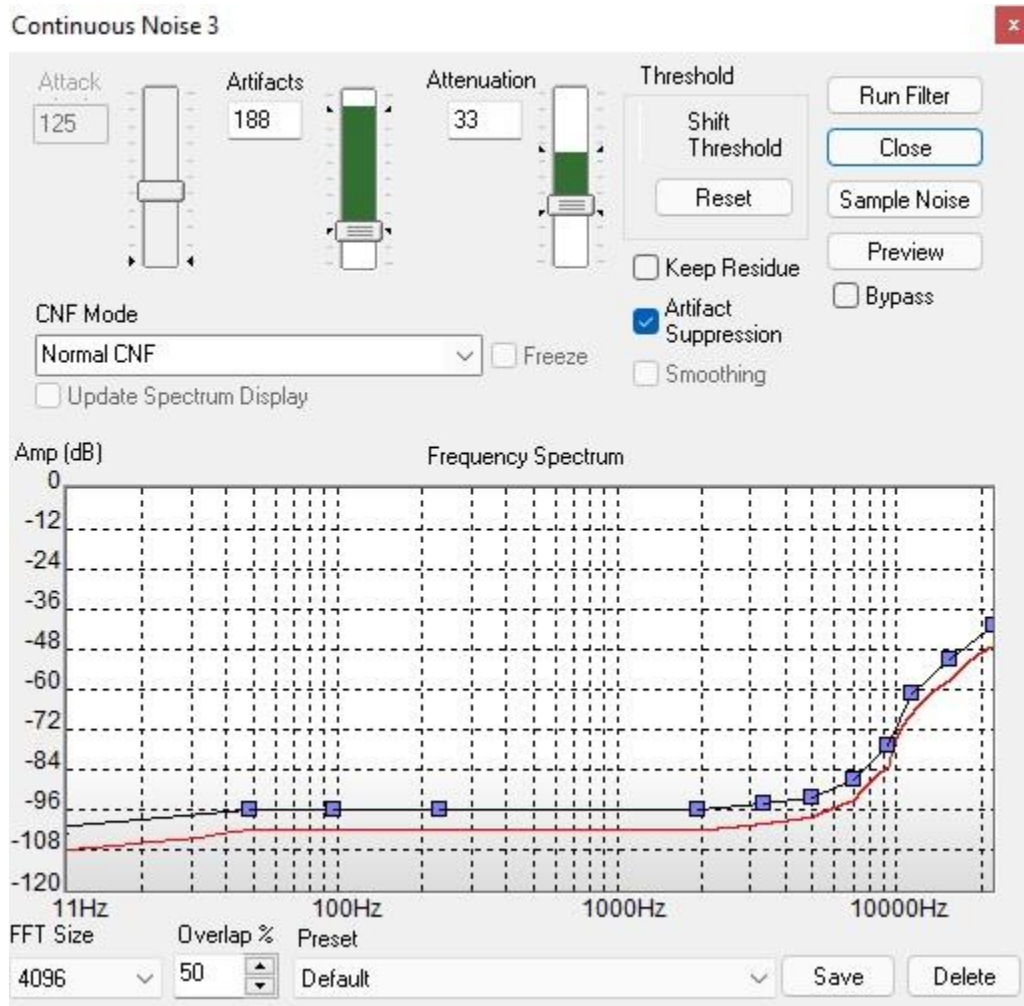


Figure 11-11 CNF (3) Diamond Disc Multifilter Part Two

This completes the noise removal process for many Diamond Disc records that were in average condition. Because of Low frequency Noise most of these Records will need extra filtering in the Advanced Section for these Records.

12.0 LP/45 Noise Removal Methods

This chapter covers removing noise from LP (Long Playing) and 45 records. These records were introduced around 1950 and originally produced with a monaural recording. After 1956 stereo recordings were gradual introduced. Noise removal for the Vinyl LP/45 records is difficult for these reasons:

1. The type of noise that occurs with the vinyl material has a shape that can be difficult for the impulse filters to find without using aggressive settings which in turn remove some of the music.
2. The surface noise of the vinyl records is low so that the quiet music passages allow the impulse noise to be heard. This contrasts with the surface noise of the 78 RPM records which can mask some impulse noise.

12.1 LP/45 Monaural or Stereo Records?

The first LP/45 records used single channel (monaural) recording technology. The stereo concept was later introduced in 1956 by one record producer and gradually became the standard for all records. Because the change to stereo was gradual and backwards compatible with monaural the record companies would sometimes produce a record that was labeled as monaural but could be a stereo record (since the record companies would manufacture one type of record for both markets). Some LP records that were labeled as monaural, were stereo.

For records made prior to the introduction of the stereo LP/45s, the conversion to horizontal movement to recover the music can also remove groove noise since the (L+R) operation (to create monaural) tends to cancel common channel noise. Because the early LP monaural records would only have horizontal movement for music the first step would be to use the (L+R) file conversion however, this is not the best choice *unless* you have *clear* information that the record is really a monaural recording. Since many records do have a stereo presence even though the label may say monaural, the file conversion (L+R) is not recommended, and the music file will be processed in all cases as a stereo recording. Use extra care to create a monaural recording from LP and 45 RPM records.

12.2 LP Stylus Size

There is another important factor about these early LPs regarding the stylus size. The first LPs used a stylus with a radius of 1.0 mils for the monaural recording. The changeover to stereo reduced the stylus size to 0.7mils and using a 0.7mils stylus in a 1.0mils application can introduce extra noise. However, the 1.0mils stylus in a stereo groove made for 0.7mils can cause damage to the groove. Thus, the best approach is to use the current stereo stylus on all LP records. Along with the Stylus radius change other changes to the shape occurred (conical to elliptical) and it was not always clear when these changes occurred. The best method is to use a current LP Stereo elliptical Stylus.

12.3 Modification Possible to RIAA Curve for Normal and Worn Condition

The LP records used the standard RIAA curve when they were made, and the Diamond Cut Software Virtual Phono Preamp should be the correct one to use for *all* LP and 45 RPM records during the Level Two file creation. However, many of these records can be improved if the Paragraphic EQ software tool (RIAA Curve or Improved) is used instead of the Virtual Phono Preamp. The Paragraphic EQ RIAA version has slightly less low-end boost than the Virtual Phono Preamp and somewhat greater high-end boost. The reason for this difference in the frequency curve is related to the specific method to implement the RIAA curve within the Virtual Phono Preamp software and the Paragraphic filter. The Virtual Phono Preamp is an exact rendering of the curve while the Paragraphic approximates the curve.

The reason that the Paragraphic RIAA can improve the sound over using the exact RIAA curve can be for two reasons. First, as records wear the high frequency rapid groove motions tend to degrade greater than the lower frequency slow motions. Secondly, the original recording may have been mastered on equipment that did not follow closely the RIAA curve. The Paragraphic EQ provides some frequency improvement that can compensate for the high-end loss.

The best judge of which RIAA software filter to use is made while you listen. You can try either one during the Level One creation when you are listening and removing noise. Then you can pick a specific one to use for the Level Two creation. Later, you can return to the Level One File and pick the other RIAA software filter to make a new Level Two file if you want. It is best to not use the *reverse* feature in the different filters as a method to undo your work on a Level Two file. Just re-do the Multifilter Part One with the other RIAA filter.

The Level One filters with different EQ settings will be shown as two separate Part One Multifilters. To keep track of which EQ is used, the phrase *Normal* will be used for the regular Virtual Phono Preamp with the exact RIAA curve and the phrase *Worn* will be used

when the other RIAA curves are used. The choice will depend on the condition of the song and your preference.

12.4 Sample Rate for LP & 45RPM Records

Because the EzImpulse filters need all the help they can get for Vinyl music, 96 kHz sampling filters should be used. If you do not have the ability to transcribe the music at this rate, then 44.1 kHz can be made to work, but the results will need more manual noise removal and lower settings on the EzImpulse filters than as shown.

The creation of the Level One and Level Two music files will follow the same workflow as described in the chapter General Noise Removal and will not be repeated in detail here. An overall workflow is:

12.5 Level One Creation

1. Select classic view for the Diamond Cut Software.
2. Copy the desired music file from the original file folder and open the copy as a source file.
3. Check Channel Balance and Adjust if needed using Virtual Phono Preamp.
4. Run the LP 45 Part One Multifilters selecting either Normal or Worn. If needed, increase the amplitude of the destination by using the CD-Prep gain increase on the destination file.
5. If many noise events remain slowly increase the EzImpulse settings to remove more noise while checking that music has not been removed.
6. Perform the Sight & Sound manual noise removal method for any remaining noise impulses in the source file.
7. When finished, store the source file as a Level One music file.

12.6 Level Two Creation

1. Copy the new Level One music file and open the copy as a source file.
2. Run the LP 45 Part One Multifilter (with normal or worn as previously selected) and make the destination into a source file.

3. Apply a Change Sample Rate operation, if needed, from the initial 96 kHz to 44.1 kHz to the source file. Make the result a new source file.
4. Normalize the source file to have a maximum amplitude of 0dB.
5. Apply the LP 45 Part Two Multifilter to the source file and make the resulting destination file the new source.
6. Trim away the remaining noise before and after the music from the source file. If desired, add extra fade down near the end of the music.
7. When finished, store the resulting file as a new Level Two music file.

After the Level Two music file has been created you can trim the start and end then copy the file into the restoration music file location, add one second of silence to the beginning and end, or continue to the enhancement chapter.

12.7 Details of the LP 45 Multifilter Part One Normal

The LP 45 Normal Multifilter Part One is shown in figure 12-1. In this figure, the Virtual Phone Preamp is used. A progression of Impulse files helps to find and remove noise.

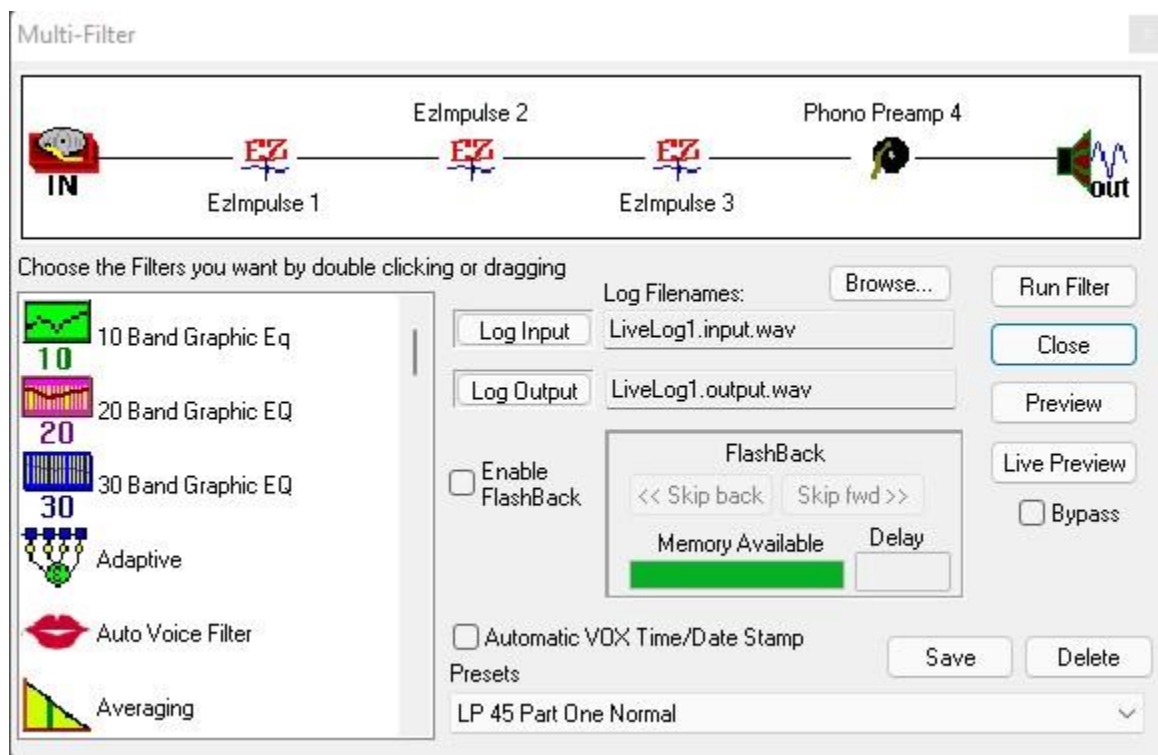


Figure 12-1 LP 45 Part One Normal Multifilter

The filter is composed of four parts that are shown below with their settings. Note that three noise impulse filters are used to help remove noise.

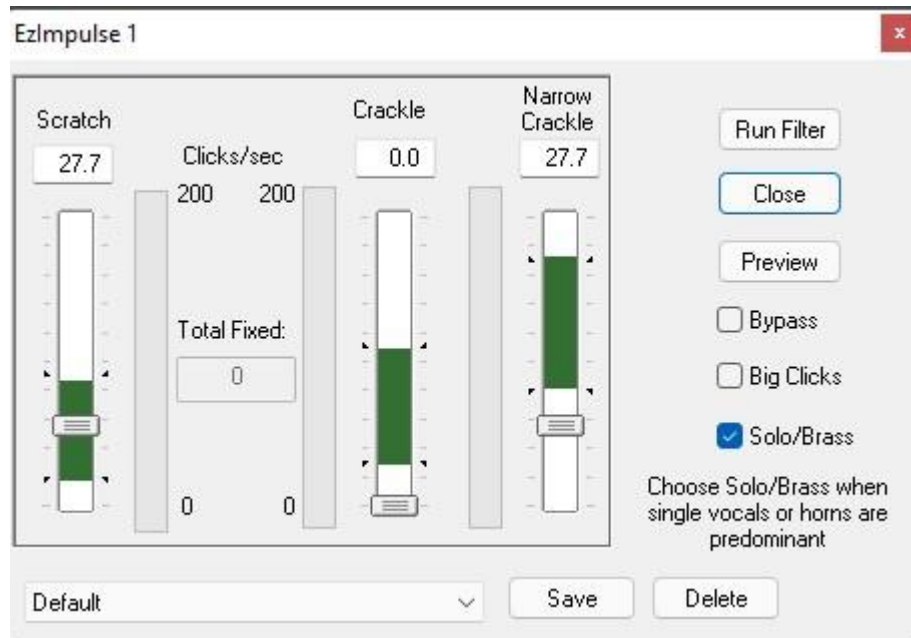


Figure 12-2 EzImpulse (1) Filter LP 45 Multifilter Part One

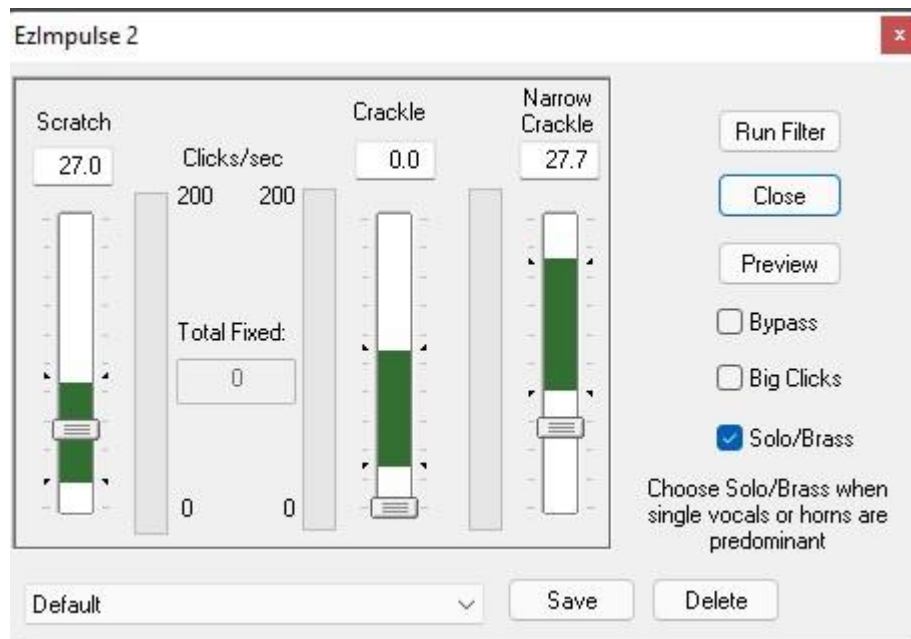


Figure 12-3 EzImpulse (2) Filter LP 45 Multifilter Part One

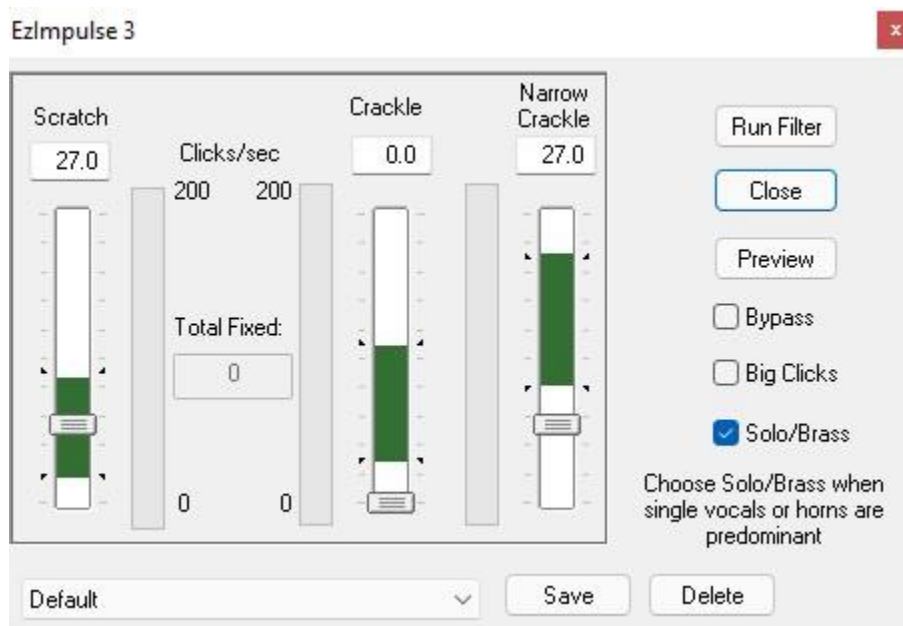


Figure 12-4 EzImpulse (3) Filter LP 45 Multifilter Part One

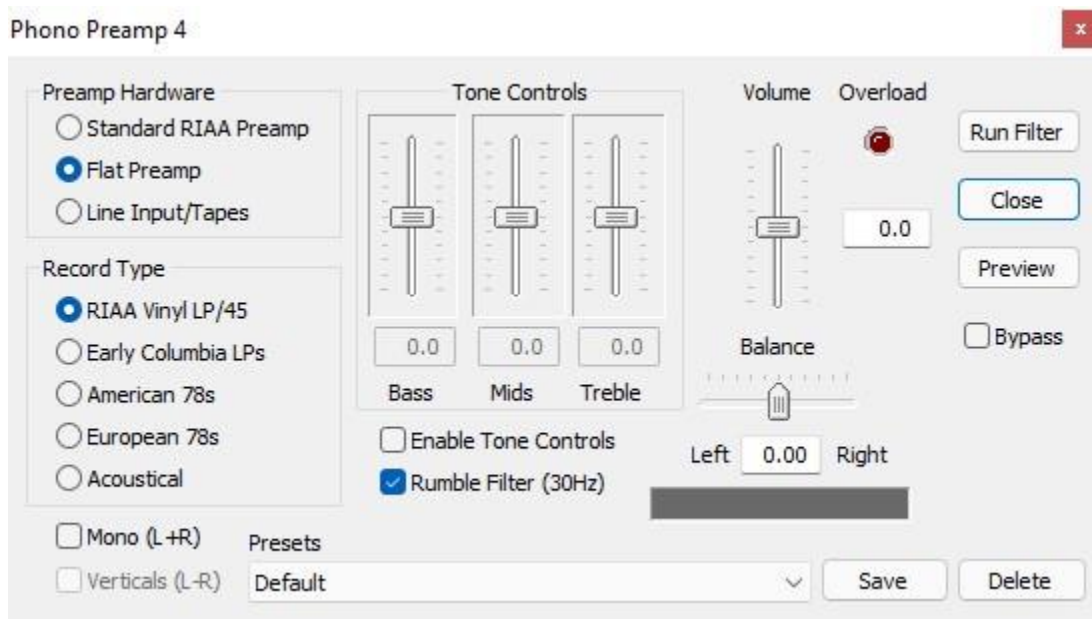


Figure 12-5 Phono Preamp (4) LP 45 Multifilter Part One

The rumble filter has been turned on to help with low frequency noise.

12.8 Details of the LP 45 Multifilter Part One Worn

The LP 45 Worn Multifilter Part One is shown in figure 12-6.

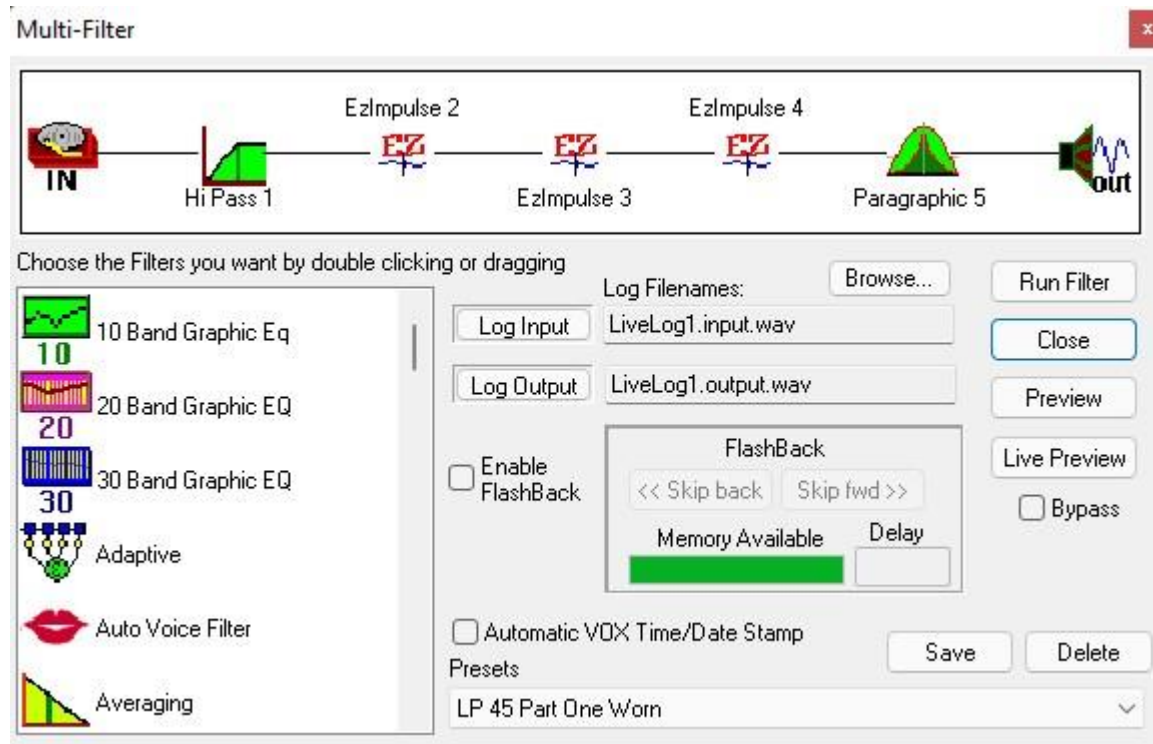


Figure 12-6 LP 45 Part One Worn Multifilter

The filter is composed of five parts that are shown below with their settings. A progression of Impulse files helps to find and remove noise.

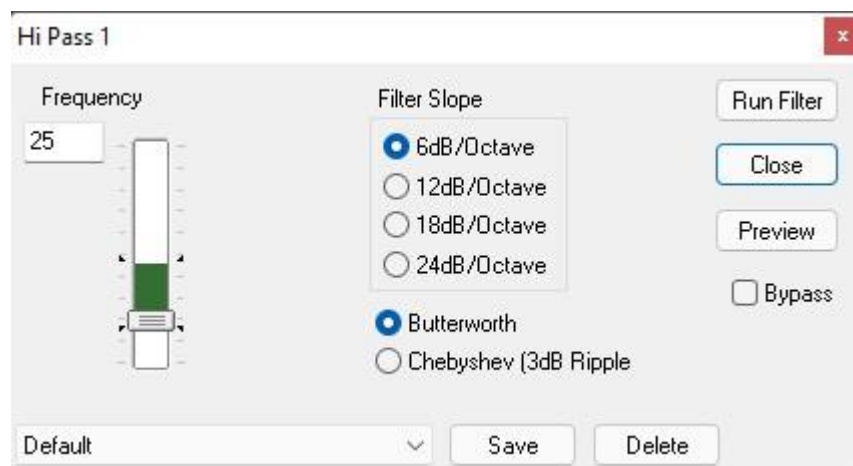


Figure 12-7 High Pass (3) LP 45 Multifilter Part One Worn

This filter removes any rumble from entering the filters since this EQ stage does not have a rumble filter.

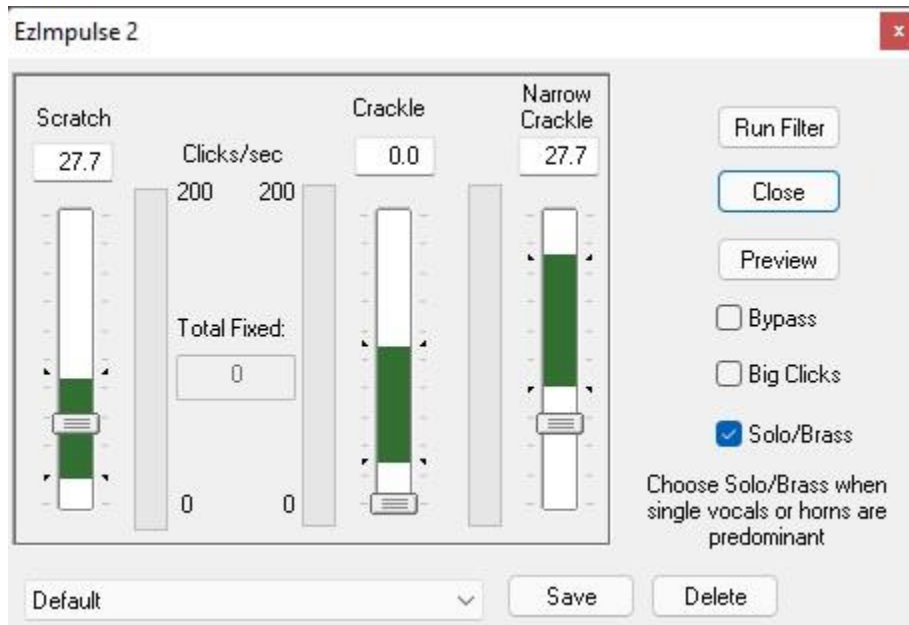


Figure 12-8 EzImpulse (2) LP 45 Multifilter Part One Worn

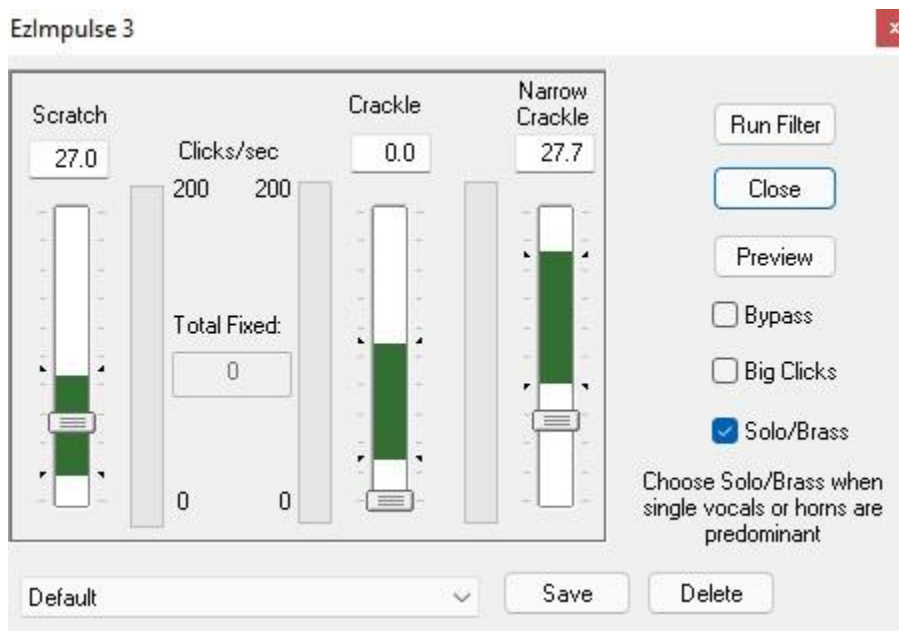


Figure 12-9 EzImpulse (3) LP 45 Multifilter Part One

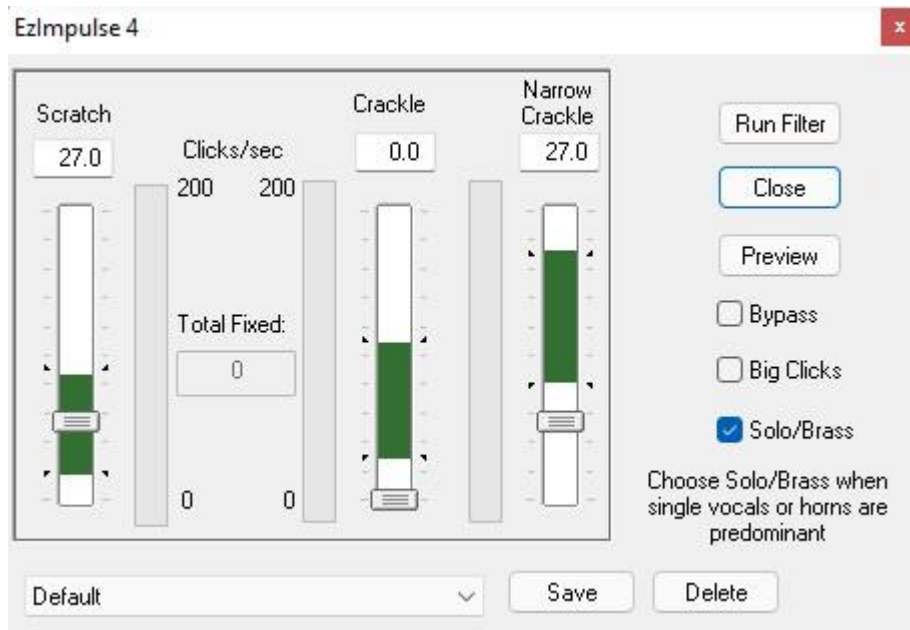


Figure 12-10 EzImpulse (4) LP 45 Multifilter Part One Worn

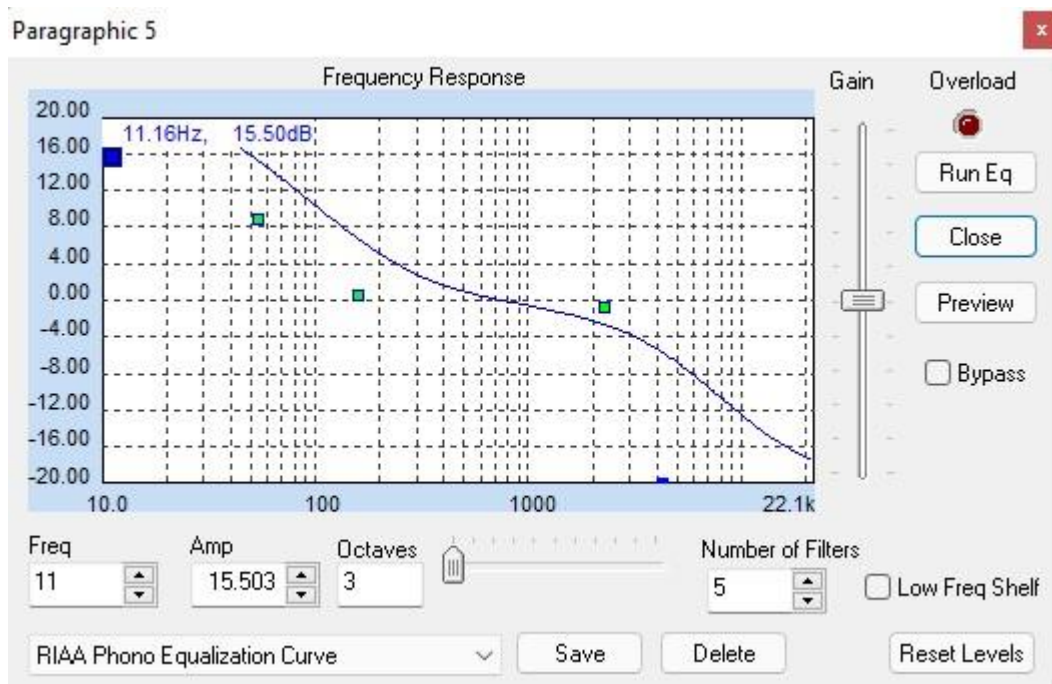


Figure 12-11 Paragraphic (5) LP 45 Multifilter Part One Worn

The Curve in this example used the RIAA EQ curve or the RIAA Phono Equalization Curve – Improved Performance.

12.9 Details of the LP 45 Multifilter Part Two

The LP 45 Multifilter Part Two is shown in figure 12-17:

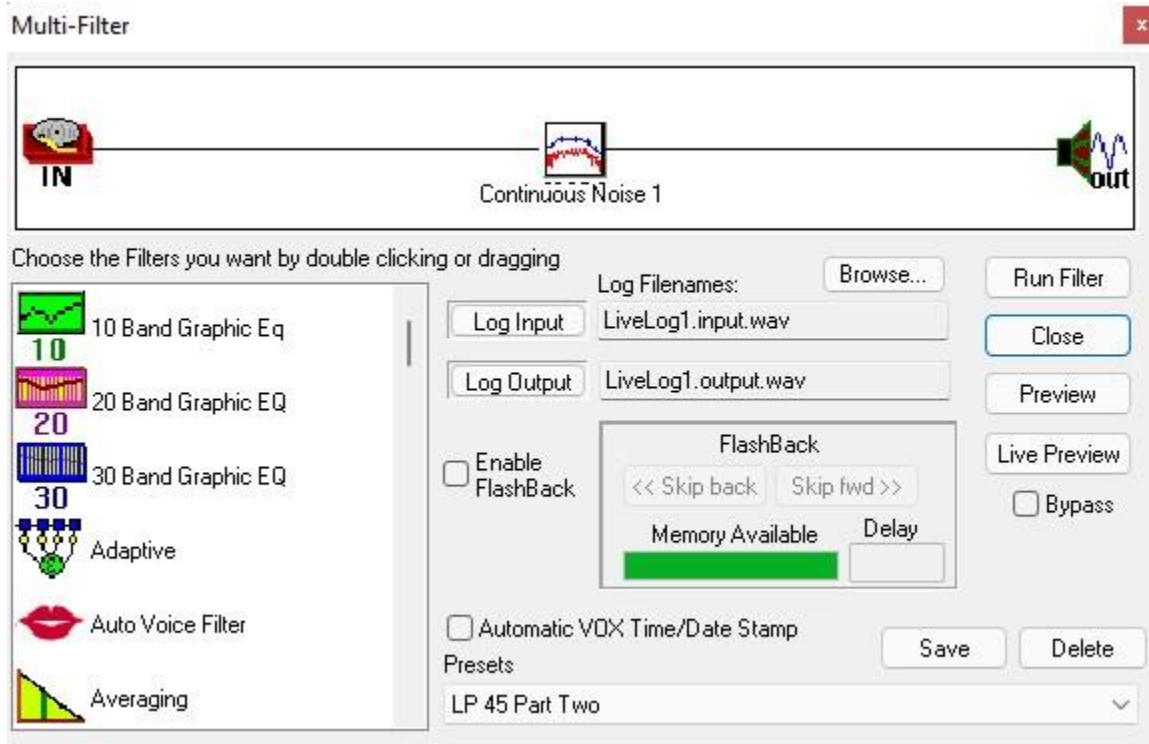


Figure 12-12 LP 45 Multifilter Part Two

The multifilter part two contains one CNF section. The details of the CNF are shown in figure 12-13:

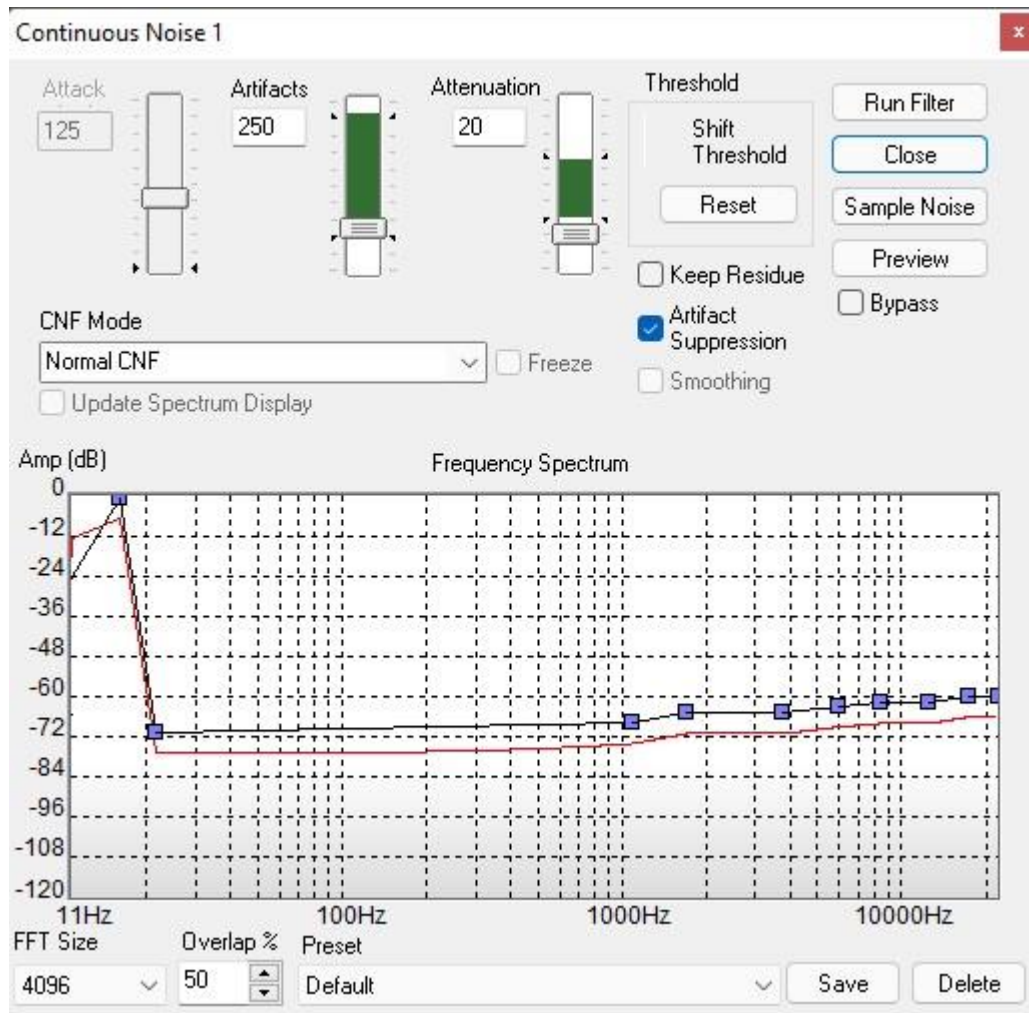


Figure 12-13 CNF (1) LP 45 Multifilter part Two

This completes the LP and 45 noise removal section.

13.0 *Tune Library and Creating Your Music Playlists*

The DCart11 software provides a very useful feature that can help you to enjoy the results of your restoration work. As your restoration work progresses you will create music files in the Restored folder and the Tune Library provides a way to find them and create playlists of music to be enjoyed on your audio equipment.

13.1 *DC Tune Library Basics*

The DC Tunes software does *not* modify the music content or file location on your system in any manner. When you create a music entry or delete a music entry in the library, the original music file location on your system is not changed. The software in the library only *points* to the file location in your computer system.

The structure of your music folders and the method that the software uses to *load* the information must be in sync with each other. This *loading of the file information* requires that you use a specific way to set up the music folders in the library layout. The help file for DC Tunes Library describes a method of loading the music to the library when the files contain *Tags* that provide needed information regarding artist, title, and more. For the Music that you have restored using the methods in this Handbook, the information about song title and artist will be provided by the file structure, not by Tags on the music file. The *loading in the music* using the file system described in this Handbook will be used instead of the *Tags* feature. If wanted, you can try the Tag method.

The Tune library's preferences window shows the method used by the software to bring in the music for the library when not using Tags. Refer to Figure 13-1:

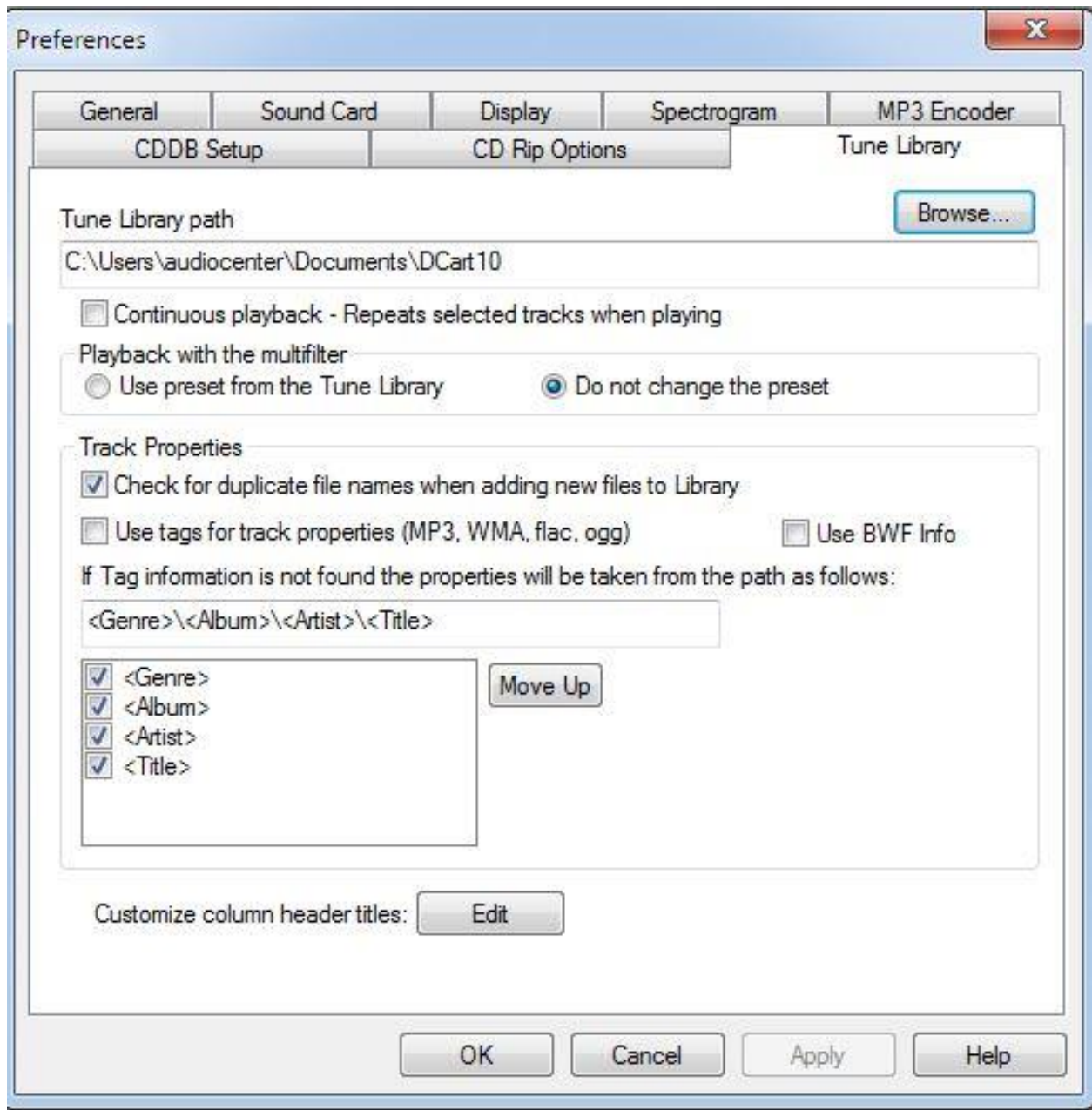


Figure 13-1 Tune Library Options

The box to use tags is not checked. The order of the loading in of the file information can be modified by clicking the feature (Genre, Album, Artist, and Title) and then selecting the move up button.

The library also provides a useful way to locate and use music that you may have from other sources, for example music CD's you have purchased. This feature for CD's will not be covered in the Handbook.

You can change the column names that show on your Library Window; however, their default names will always appear within the software in the Tune Library Tab under Preferences.

13.2 Music Folder Layout on your System

The source of the music for the library will come from the *Restored Music Files* on your computer system. If you use the methods described in Chapter 5 (Music File Organization), then the DC Tune Library will *pull in* the file's *information* correctly after some modifications to the preferences section.

Under the Preferences Tab is the library path that shows where, the actual file information for the library is located. The DCart11 help files describe these files and it is possible to delete some or all if you want to *start over* with a new library set-up. These files will be found in the Documents location using Windows Explorer.

If you have two different Diamond Cut Software products (DCart11 and DC Forensics) you can use a common library path for each program so that the libraries are in sync. The ability to find a specific song with this database is very useful.

13.3 Library Headings

The library software has four columns that have default names labeled: Genre, Artist, Album, and Title. When the library is used, additional column information is shown to the right of these names providing more information about the files.

The column names used are shown below and have been changed from the default names using the edit command in the Tune Library Preferences section. Figure 13-2 shows the changes:

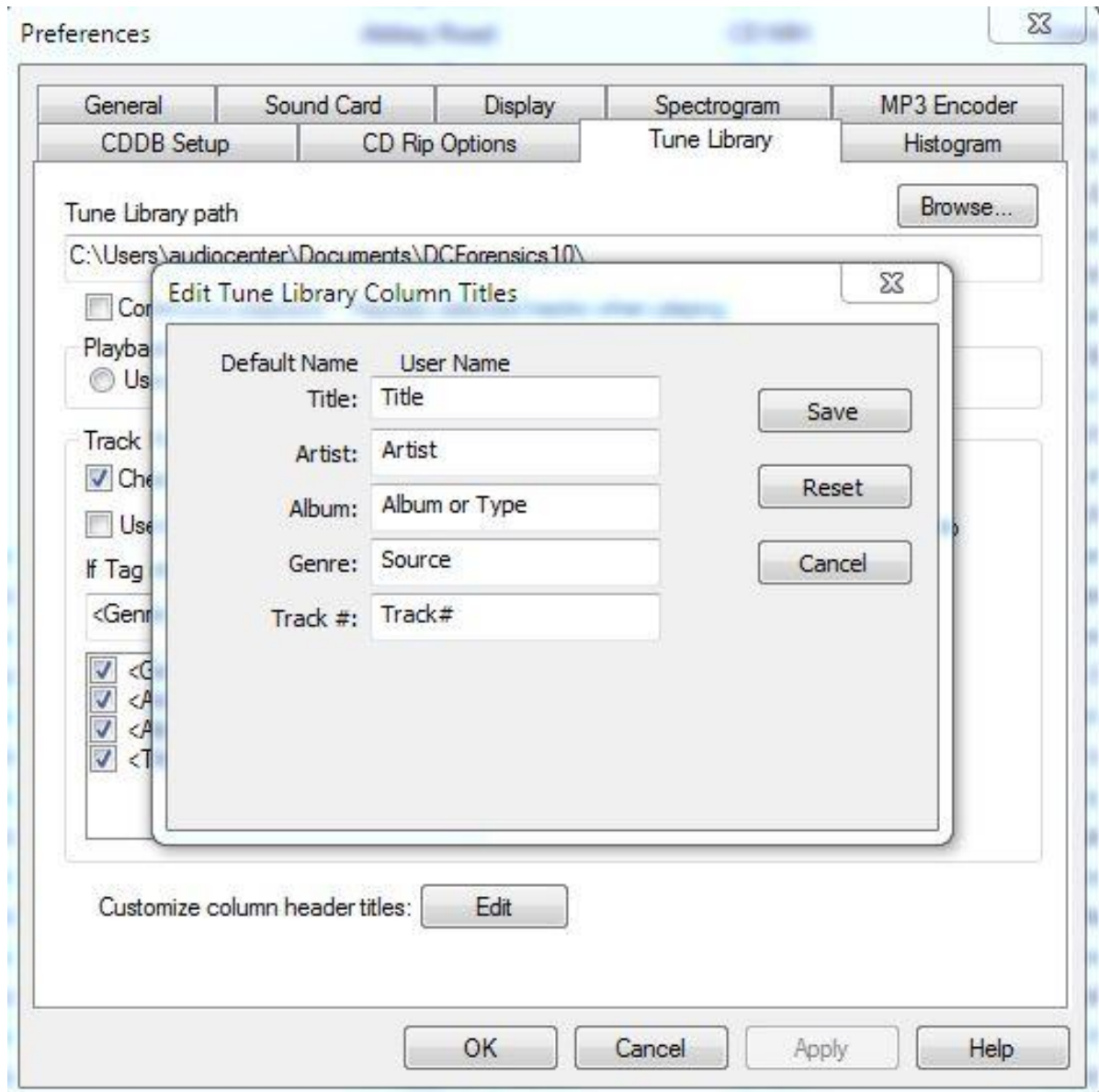


Figure 13-2 Library Column New Titles

When this command is performed, the Tune Page will look like this:

Title	Artist	Album or Type	Source

Figure 13-3 Library Columns

This figure shows only the beginning of the library columns as they continue further to the right. Now that the library columns are labeled, music from the restoration folders can occur. The importing of the music will be different for LP Albums versus individual records.

13.4 Importing the Music File Information into the Library

The first step will be to set the library preferences to the correct values unique for each of the two layouts, Individual Records or LP Albums.

When the music file information is imported, the various details of Title, Artist, etc. are entered in a specific manner by the Tune software. When you have imported a folder, review the results in the library to understand the details of *how* the information came from the folder headings with the specific and Column Title preference settings.

13.4.1 Individual Record Setting

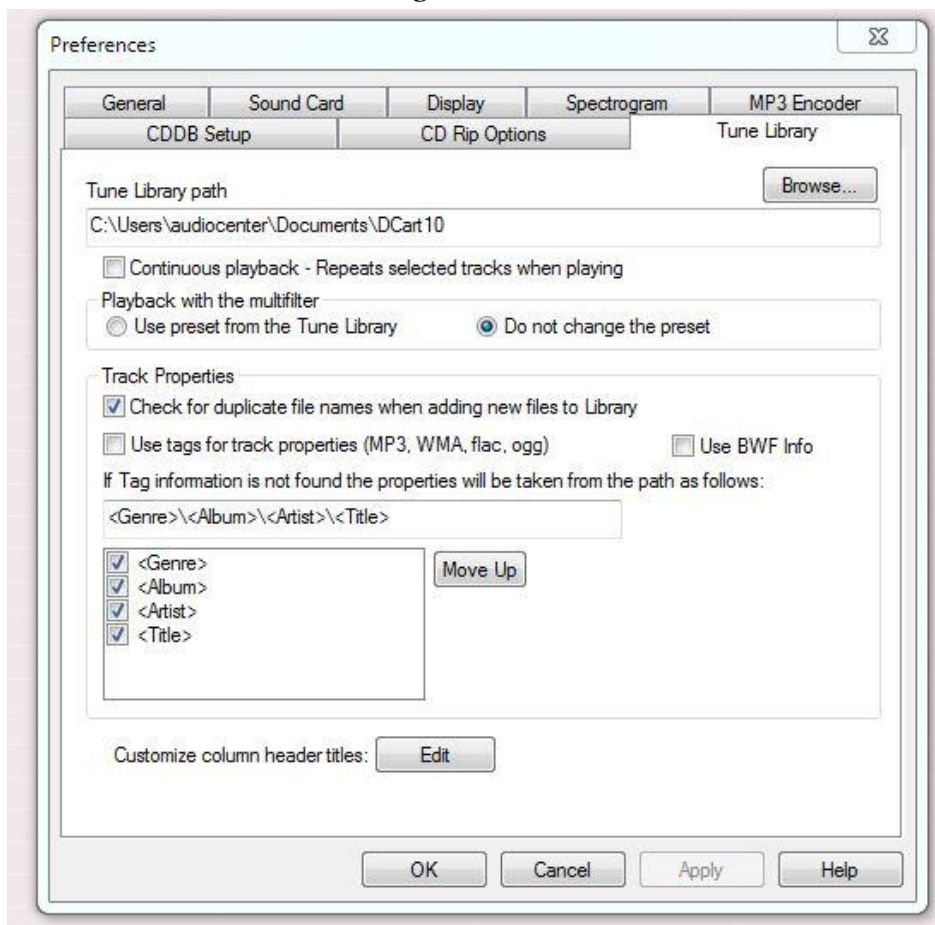


Figure 13-4 Preferences for Individual Records

Next, we select the import folder command as shown in figure 13-5:

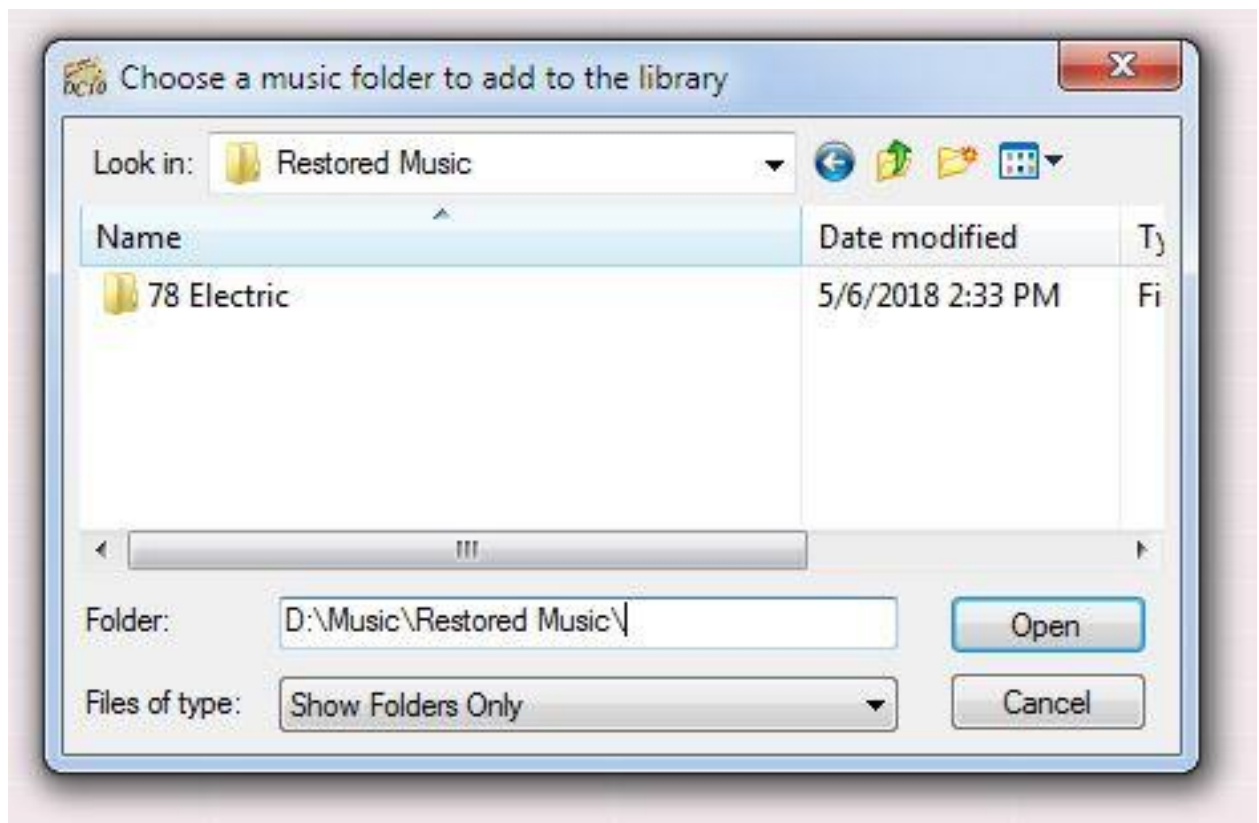


Figure 13-5 Importing Folder

After the command is opened the Tune Library will contain all the files in the 78 Electric Folders (Al Jolson for this example). If you have other individual records that are Edison Diamond Disc records or 78 Acoustic, you can perform a similar operation.

13.4.2 *LP Records*

The preference settings are changed to the values shown in figure 13-6:

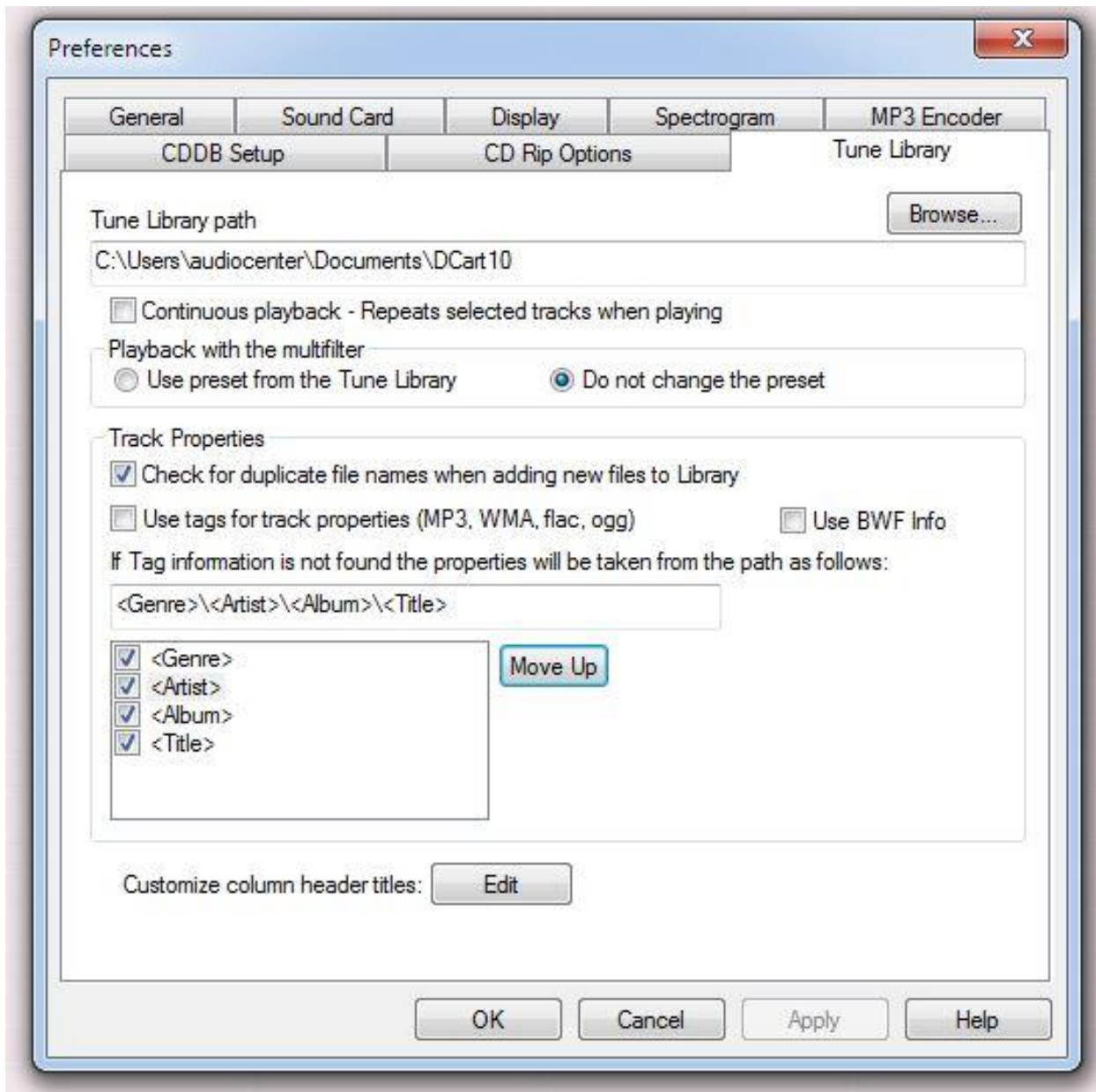


Figure 13-6 Preferences for LP Albums

Use the move up command to change the order. Now we will select the import folder command as shown in figure 13-7:

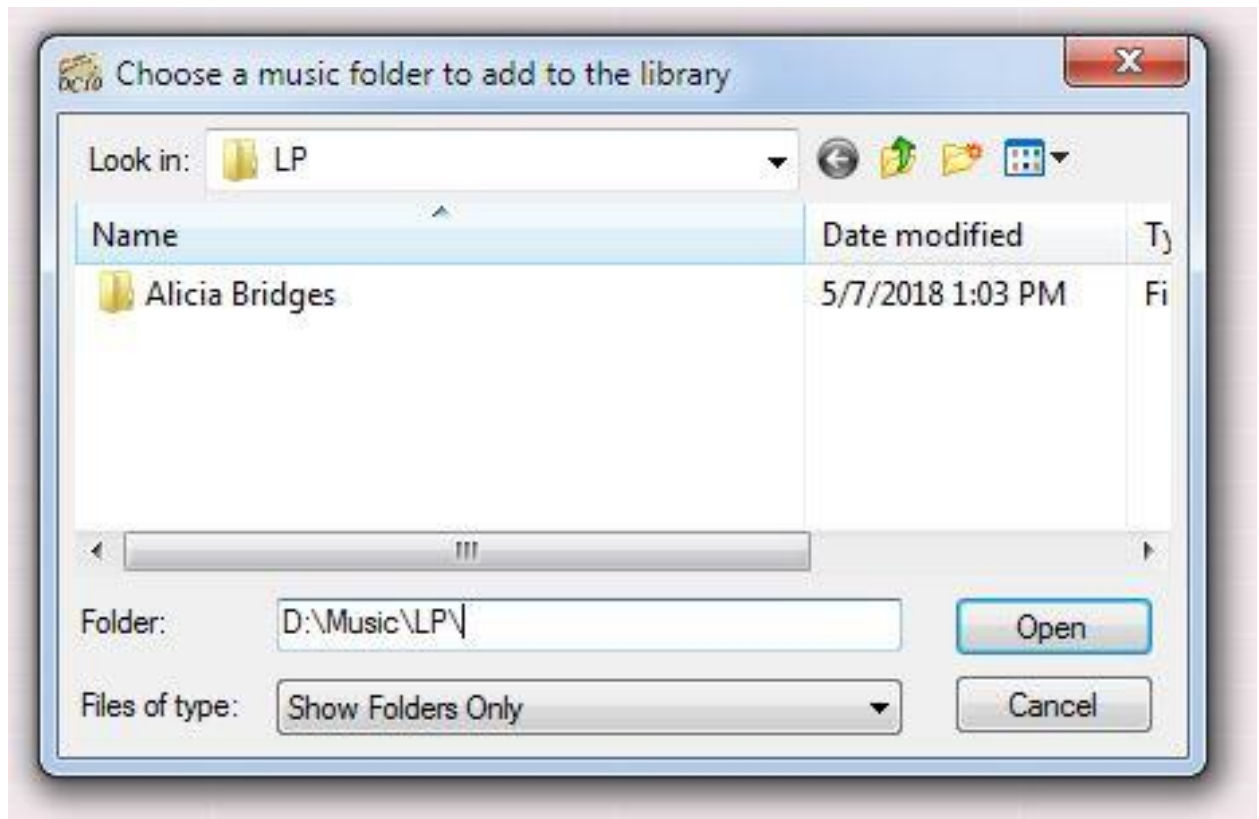


Figure 13-7 Importing LP Folders

The results of these two operations are shown in Figure 13-8:

Title	Artist	Album or Type	Source
Carolina In The Morning	Al Jolson	78 Electric	Restored Music
I Love The Night Life (Disco Round)	Alicia Bridges	Disco Round	LP

Figure 13-8 Tune Library Results

The results for this Tune Library example can be modified for your own titles and customized the way that you want. Playlists of music can be created for a CD or other digital storage device in the Tune Library.

The Source listing above will vary between LP Albums and Individual due to the needed folders to hold the music.

Advanced Enhancement Methods

14.0 General Concepts for the Enhancement Methods

The previous Basic Restoration Section concludes with music files that have the record noise removed and located in the Level Two Folders.

This chapter describes some concepts to enhance the recording from your previous restoration work. Chapters will address specific methods for different types of recordings. A special chapter describes new methods developed for restoring the Acoustical Recorded Records back to their original studio sound.

14.1 Starting Condition of Files

All Enhancement Work starts with a Level Two file that represents the best noise removal and equalization that you could achieve for the record.

14.2 Types of Enhancements

The previous work returned the music to the condition that the recording producer wanted for the record. For many music files this will be the music that you will place into the final Restored location. The use of the word *Enhancement* means that some type of *modification to the frequency content of the original desired recording will be made*. The use of RIAA or 78 EQ by themselves is not an enhancement as it was required for the correct playback of the music.

There are Five types of Enhancement

1. Removal of Noise Frequencies that were retained on the master recording due to the limited playback technology at the time of recording.
2. Modifications to the recording signal that were placed on the record due to the limited state of the art for both recording and playback technologies.
3. Increase or decrease in the frequency range of the music to counter-act remaining high order distortion in the music.
4. Modifications to the recording frequencies that you believe will improve the sound of the music for their own taste in music.
5. Generation of new music frequencies that were missing or very much reduced from the original studio recording during the Acoustical Recording Era.

14.3 Details for the Different Types

14.3.1 First Type of Enhancement

The first enhancement removes frequencies that were recorded in the music but were not a concern to the producer at the time since these frequencies could not be reproduced during the playback *when using the equipment at the time*. Consider an acoustic recording that had significant frequencies present above or below the limited playback range from about 300 Hz to 3000 Hz. These noise frequencies can now be heard with today's audio equipment and can be removed. When the music was recorded, the audio engineers, did not need to avoid recording this noise if the playback equipment could not reproduce the noise.

An acoustic recording may have recorded low bass frequencies in the same location as turntable rumble noise. If these low bass music frequencies are simply removed along with the turntable rumble noise, then a possible improvement to the acoustic recording has been lost. As the enhancement noise filters are used, a balance will often be needed between removing real noise and real music.

The high frequencies contain surface noise from the material used in these early records and can easily dominate the desired music. Many acoustic and even later electric playback phonographs did not reproduce frequencies above 5,000 Hz. Today we hear the scratchy sound and it degrades the music.

The Edison Diamond Disc Records are an example of this type of recording that can be enhanced by removal of low frequency rumble that occurs in many of these records when played back on today's Turntables.

14.3.2 Second Type of Enhancement

The second enhancement will use your own hearing to determine the amount of music frequency modification that maybe needed.

Early electric music customers were preconditioned to expect a certain sound when a record was played back due to their experience hearing music on the new radio technology. The wide spread use of radios predated the introduction of electric recorded records (many early radios used battery power as widespread utility home power was not available). When a new electric recording of a song was first heard the expectation was to hear a similar range of music frequencies as heard from the radios. Later into the early 1930s, many listeners of

records were hearing records with deep bass due to the introduction of *jukeboxes* in public spaces. The record producers would often boost highs and lows to counteract the playback systems own frequency modifications to give the sound they believed the public wanted to hear. Today with our hi-fi audio systems that can respond from 20-20,000 Hz the same record that sounded good in 1932 will often sound wrong with a modern system. Converting the recording to sound natural with today's audio systems will be a second type of enhancement.

The use of *Pre-Emphasis* is a method to extend the upper frequency range of the music by reducing the effect of surface noise on the music. This concept was first discussed in the 1925 paper by the Western Electric Laboratory regarding electric recording, however it first gained commercial use much later in FM (Frequency Modulation) radio. This boosting of the higher frequencies was applied to some of the 78 RPM records; however, this practice was not uniform with the record companies and was not well documented.

For the 78 Electric Level Two results, the equalization consists of using a single turnover value and does not decrease the frequencies above the turnover starting value. There will be some records that had an increase in the higher frequencies and this *extra boost* during recording may need to be reduced for playback with today's audio systems.

Another type of record that fits in this section is the Edison Diamond Disc Electric recorded records. These records were developed by Edison independent from the 78 RPM Electric records and have their own unique EQ. Your own hearing will be the best method to adjust the finished restored product.

14.3.3 *Third Type of Enhancement*

The general optimum range of frequencies to reproduce music was studied in detail from the mid 1920's through the 1950's when many pioneering scientists started in-depth studies understanding how we hear speech and music. The cost of many audio devices from radio receivers to broadcasting stations are price sensitive for the range of frequencies that are reproduced which in turn helped to sponsor this research. The Audio Engineering work performed by Harry Olson in his book *Elements of Acoustical Engineering* is one of many excellent sources that details important findings regarding the relationships between the frequency range of the music, the distortion in the music, and the perception by listeners on the quality of the music. Distortion in the music is defined as frequency modifications to the notes in the music. Musical instruments create harmonic frequencies from the base fundamental note and any distortion added to these higher order harmonics will degrade the

sound. An example of frequency modifications would be a tearing sound in a brass instrument. Some of the key findings from the research were:

1. If the music has very little distortion, then a wide frequency range during playback will enhance the quality of the music as heard by a listener.
2. For music with some noticeable distortion, a reduced frequency range will help to reduce the effect of the distortion on the music as heard during the playback to the listener. The reduced range is for both the low and high frequencies.
3. The range of music should be such that the *arithmetic mean* of the frequency range is centered around 800 Hz to produce a balanced overall sound. Therefore, if you reduce the upper frequency limit you need to reduce the lowest frequency by a corresponding amount to keep the mean around 800 Hz. Another similar method is to have the product of the lowest and highest frequency be approximately 500,000 (Hz Squared) for a balanced sound.

14.3.4 *Fourth Type of Enhancement*

The fourth type of enhancement provides an opportunity for you to perform some *music creativity*. The Diamond Cut Software provides many ways to modify the sound from the recording to satisfy your desires *independent of the original studio sound*. For example, a pseudo-stereo effect from the original monaural can be achieved through time offsets and selective right and left channel filtering. Extra reverberation or echo can be introduced if want to add a lively sound to the music. Some examples will be provided.

14.3.5 *Fifth Type of Enhancement*

The fifth type of enhancement is a new area of research I have developed. Many of the early Acoustical Records used musical instruments that produced frequencies over which their total range was not recorded. The lack of recording low frequencies, that were often the *fundamental* tones, meant that the music had a *tinny* sound that was not heard in the recording studio. Many of the higher harmonics to the instrument's sound did get recorded on these early records which provides a new technique to add back to the music; the missing fundamental tones.

While this concept appears to be difficult, the idea of replacing the missing fundamental note is performed by our brain when we hear certain types of music. The general study of

Psychoacoustics describes this and other aspects to hearing and should be consulted for more information. The encoding of music to the various standards (MP3) takes advantage of psychoacoustics and removes some of the music frequencies to minimize the file size with the expectation that the human brain will create the missing values.

14.4 Examples of Enhancement Multifilters

Later sections will provide enhancement examples that include a combination of types one, two, three, four, and five that are tailored for various types of recordings. The specific settings shown in these Multifilters are *starting* points for your enhancement work. The final setting that you use will often be unique for each record. Although the examples are specific for a certain type of record, they can be applied to other types of records.

14.5 Saving the Results of Your Enhancement

The music that you use for the enhancement work comes from the Level Two location. The results of the enhancement can be placed in the Restored File location. You will want to keep the original Level Two file before the enhancement operation so that you can try other enhancements on the Level Two music.

15.0 First Type Enhancement

This example will use an Edison Diamond Disc Acoustic Record, but the methods can be used on other Acoustically recorded songs. The Edison Diamond Disc Acoustical Records have many songs that contain extra low and high frequency distortion. Because there is some music present with the noise the various filters used in the Level Two restoration section were set to wide frequency ranges. When rumble and hiss are present more removal may be needed from the Level Two results. First step is to remove low frequency noise rumble.

Rumble present on Acoustic Diamond Discs was caused by manufacturing defects in the disc surface that allows the stylus to have a vertical motion not due to the music. Examination of these disc surface shows bumps and depressions that usually cause a *thump* sound once per revolution. A close examination of this type of noise has shown that most of these frequencies are below 150 Hz. When the same record that has a thump sound is played on a period Edison Phonograph this noise is not heard since the Edison Phonographs low end

is around 200 Hz. A series of figures will show this in detail and the solution to the noise will be to remove all frequencies below 150 Hz for a noisily record.

The following is a frequency response of an Acoustic Recording as heard from a period Edison Diamond Disc Phonograph. The Song that was played was *The Charleston* by the Golden Gate Orchestra. Refer to figure 15-1:

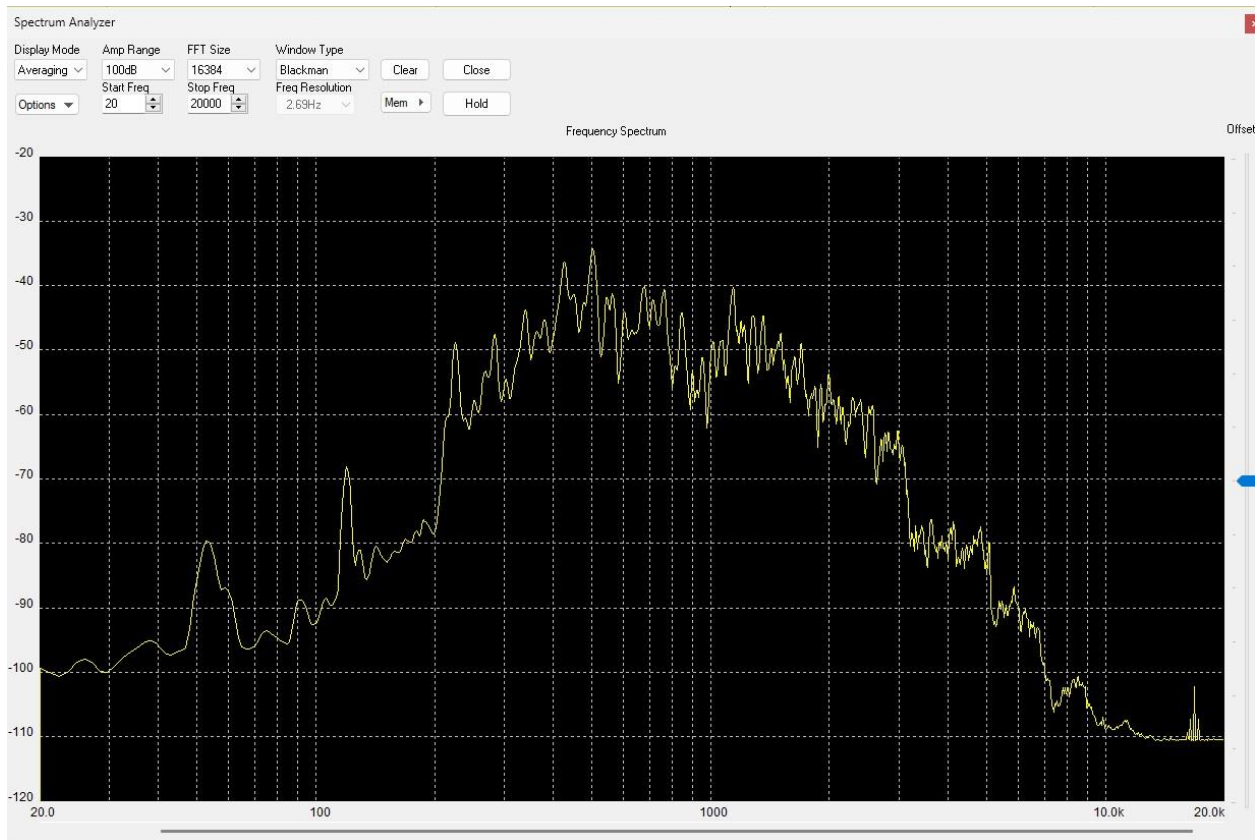


Figure 15-1 Charleston Song as heard played from an Edison Record Player

The frequency output is about 200-3000 Hz and no low-end distortion or *thumps*, or *hiss* were heard. The noise spike at 120 Hz was related to an environmental sound during recording and not from the record. The Level Two frequency response of this song is in figure 15-2:

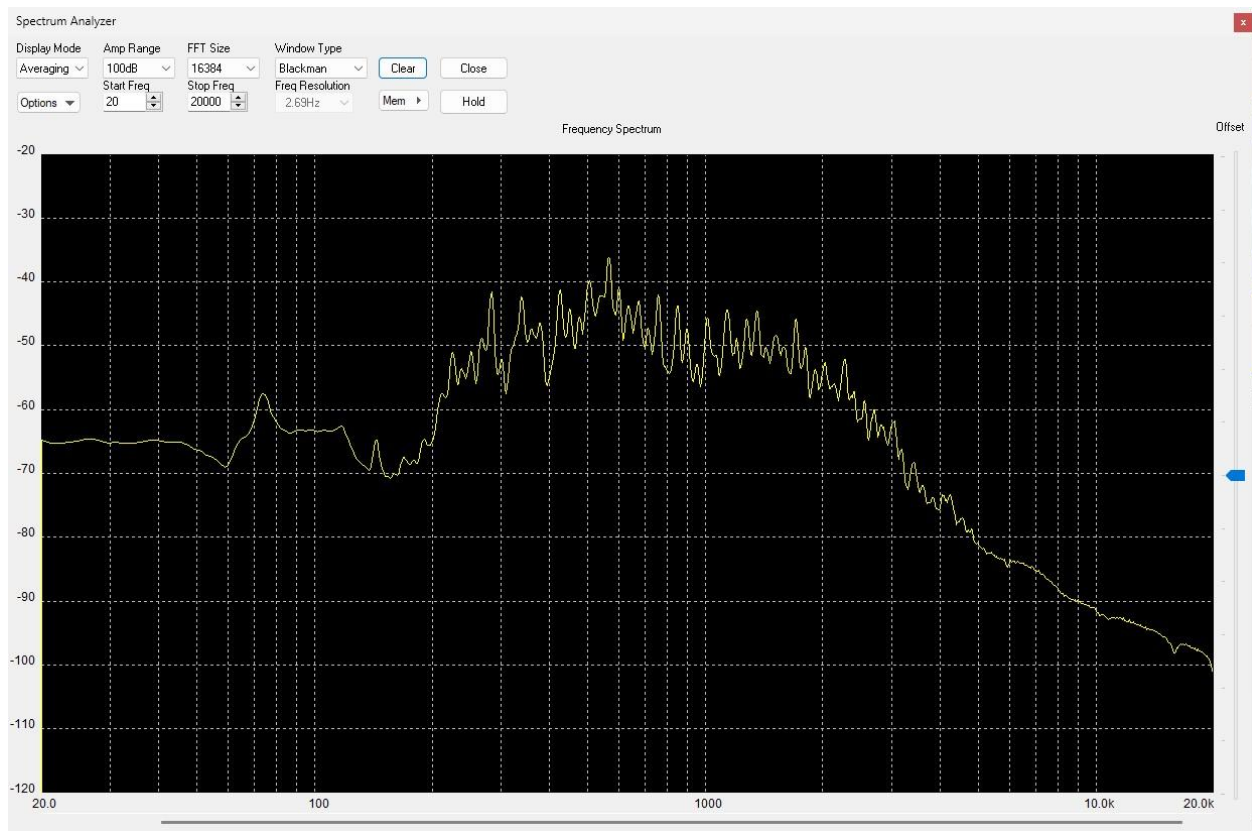


Figure 15-2 Charleston from Level 2 Restoration

Notice the large amount of low frequency energy below 150 Hz that appears to be noise rather than music. The high end shows the result of the Level 2 filtering with the high-end drop-off. Both Low and High frequencies will need some modification.

15.1 Rumble Removal Acoustic Diamond Discs

The frequencies from 150 Hz and lower contain noise with a very small amount of music that is *masked* or covered over by the rumble. The frequencies and noise that will remain above 150 Hz after filtering are a compromise between hearing the rumble and keeping some music. After many tests on these Diamond Disc Records, a Continuous Noise Filter at 150 Hz Cut-Off was found to remove the rumble. The Multifilter used for this rumble is shown in figure 15-3:

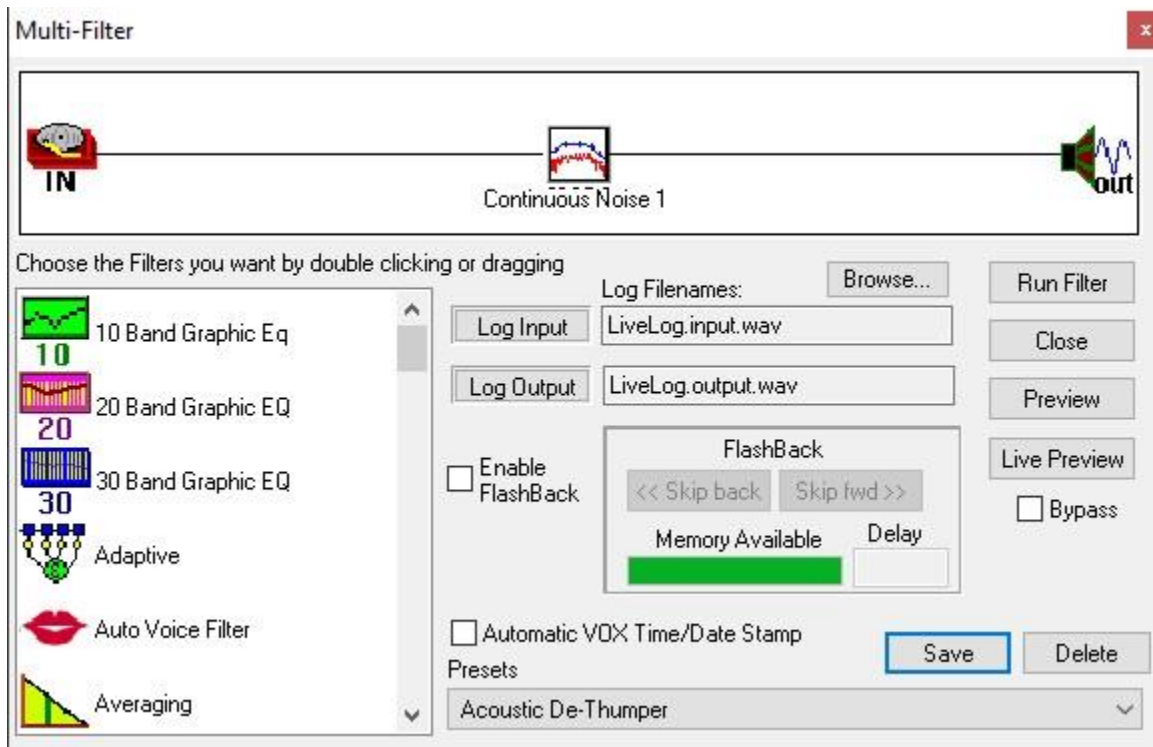


Figure 15-3 Multifilter for Rumble Removal

The details of the continuous Noise Filter are in figure 15-4:

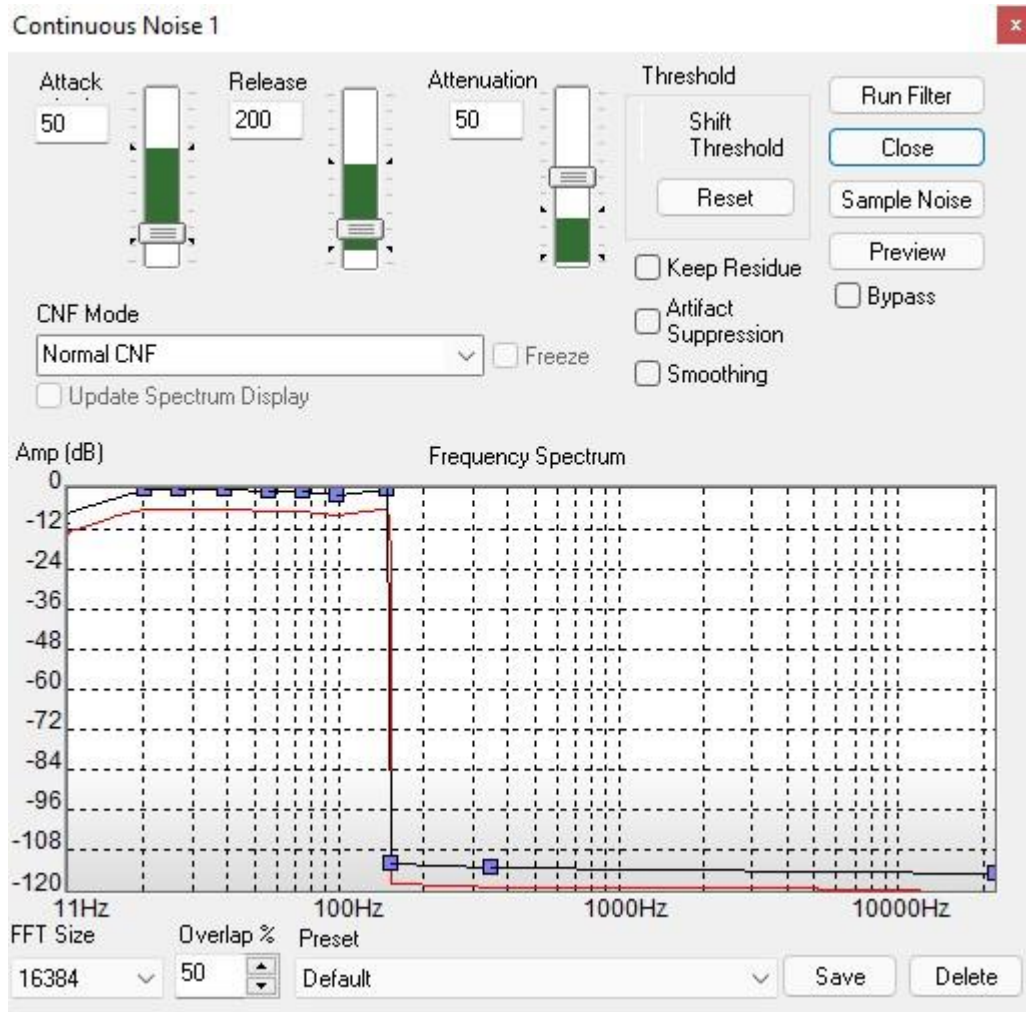


Figure 15-4 CNF (1) for Multifilter

The resulting frequency spectrum from using this multifilter on the Charleston Level Two Song is in figure 15-5:

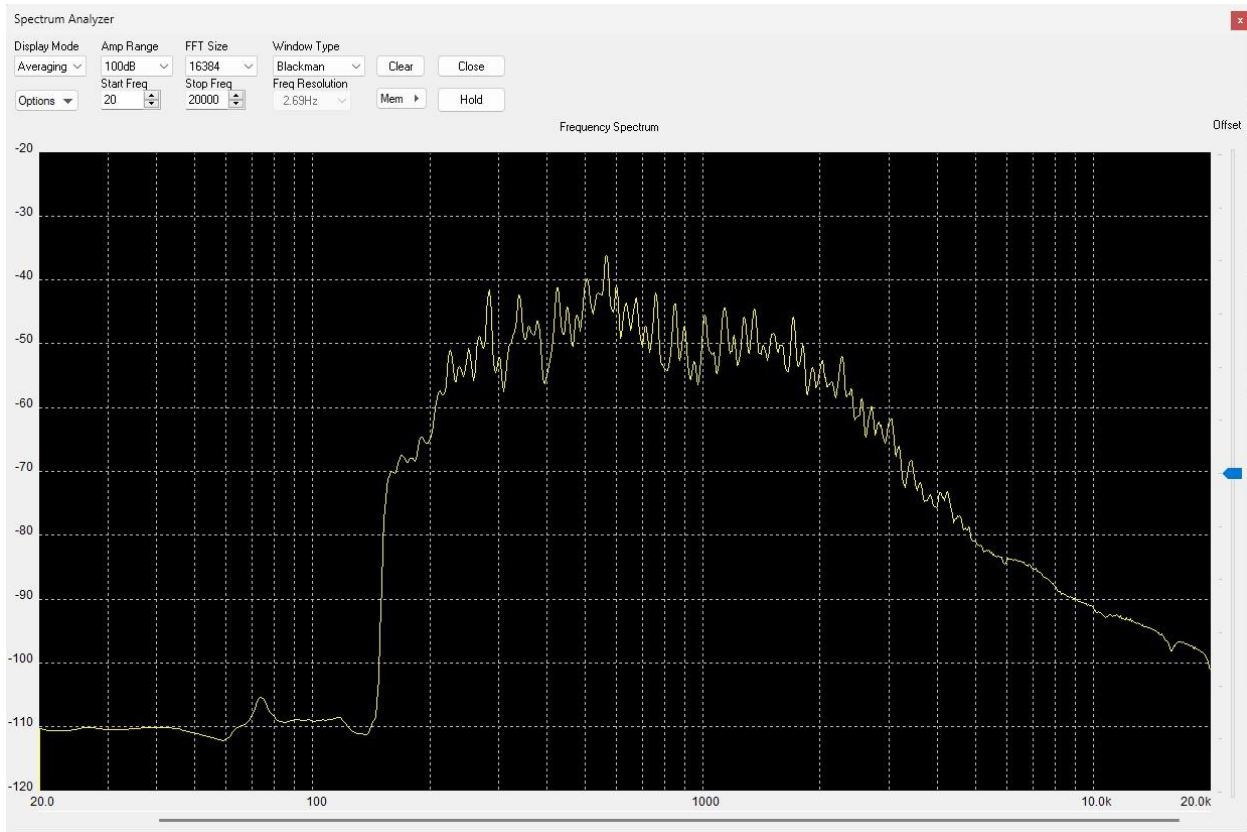


Figure 15-5 Charleston after De-Thump Filter

Compare 15-5 to 15-2 to see how the low frequency noise is removed. When this song is played back the rumble is gone with a much better low-end sound. The high-end noise will now be removed.

15.2 Surface Noise Removal and Final Cleanup Acoustic Diamond Discs

The high-end frequencies will often contain music and record surface noise. When the high-end surface noise frequencies are simply removed by a sharp filter the music can have a somewhat dull or lack of sparkle to the sound. It is easy to cut off music frequencies along with surface noise at this point so an alternative to a low-pass filter is to boost the frequencies somewhat just before the high frequency cut-off and then use a somewhat gradual CNF to limit the scratch sounds. By slightly boosting the higher frequencies just before the cut-off you are helping your ear to *replace* some of the lost music to the surface noise. Note that this method may need adjustments for a specific record as the surface noise varies from record to record.

Figure 15-6 shows the Multifilter used with the name Diamond Disc Acoustic Cleanup Multifilter. Since this filter is the last operation that is used on the file some low frequency boosting is also present. However, the low frequencies are missing from much of the music and many of them cannot be boosted much at this point.

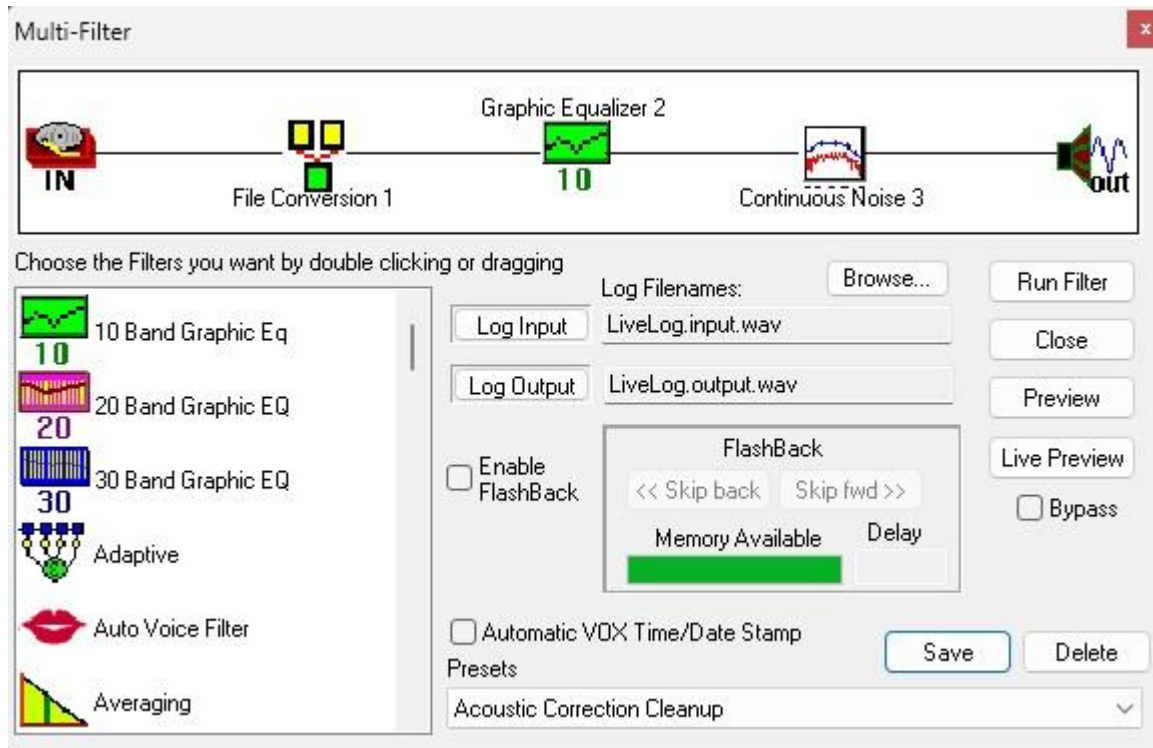


Figure 15-6 Diamond Disc Acoustic Cleanup Multifilter

Figure 15-7 shows the first block which is used to slightly lower the amplitude so that any later boosting will not saturate the music file.

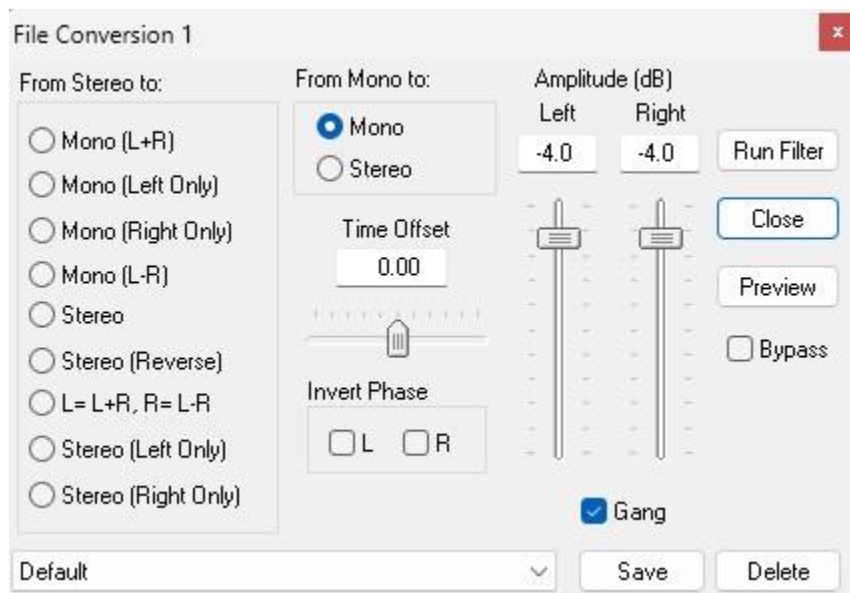


Figure 15-7 Gain Block (1) Diamond Disc Acoustic Cleanup Multifilter

The next filter boosts some of the frequencies in the music. Refer to figure 15-8:

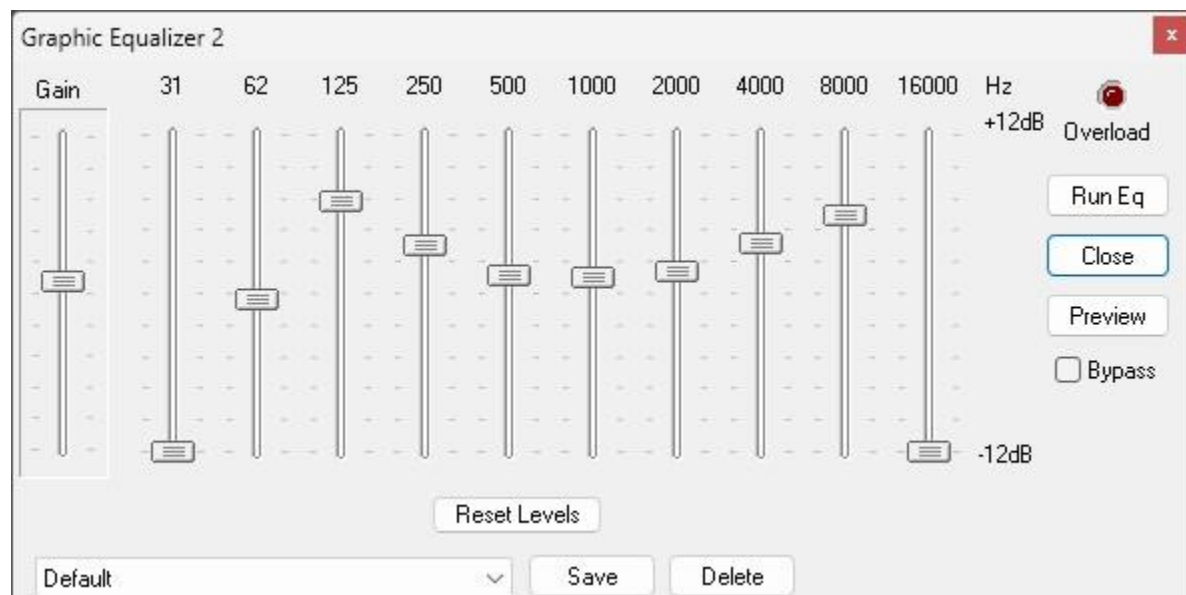


Figure 15-8 Frequency Boosting EQ (2) Diamond Disc Acoustic Cleanup Multifilter

The last filter cleans up surface noise that can be heard. Refer to figure 15-9:

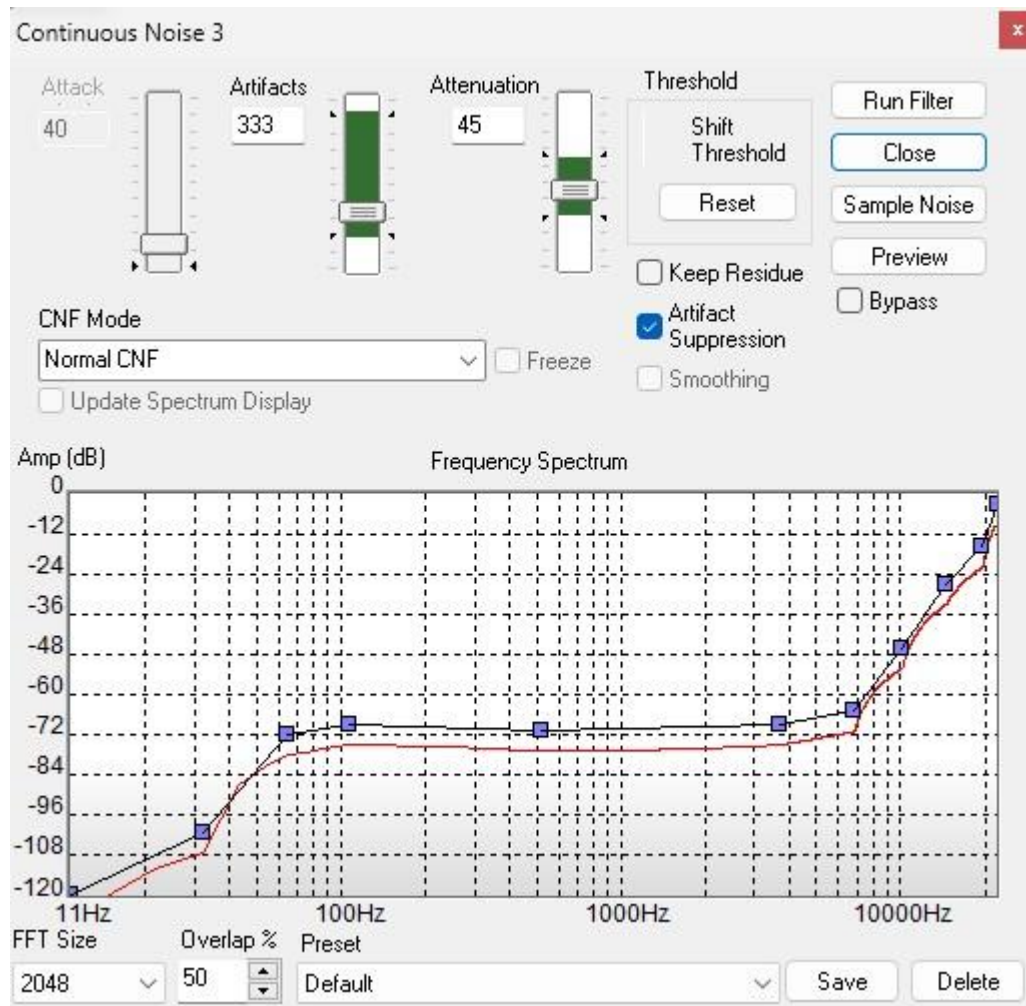


Figure 15-9 CNF (3) Diamond Disc Acoustic Cleanup Multifilter

This concludes the section on the *First type of Music Enhancement* for a Diamond Disc Acoustic Record. The same general approach can be used for 78 RPM Acoustic Records or other early Cylinder Acoustic recordings.

16.0 Second Type of Enhancement for Various Record Types

The second method of enhancement modifies the frequency of the Level Two music to sound *correct* using today's audio playback equipment; rather than what was available in the past. Music frequencies can be boosted or reduced to achieve a normal sound, as if you were in the recording studio.

16.1 78 RPM & LP/45 Records

This enhancement shares some of the techniques used for EQ selection for both 78 RPM Electric and LP/45 RPM records. When the Level Two recordings were made for these records, you had an opportunity to use the *standard* EQ values or a modification. For example, the *Worn* EQ for LP or different turnover values for 78 RPM records. A further modification can be made using the 10 or 20 Band Graphic EQ filter under the Filter Tab however, the 10 Band version is adequate and is a major change from tradition Bass and Treble Shelf type of controls. A picture of the filter to reduce high frequency from added pre-emphasis in the recording and slightly boost the low end is in figure 16-1:

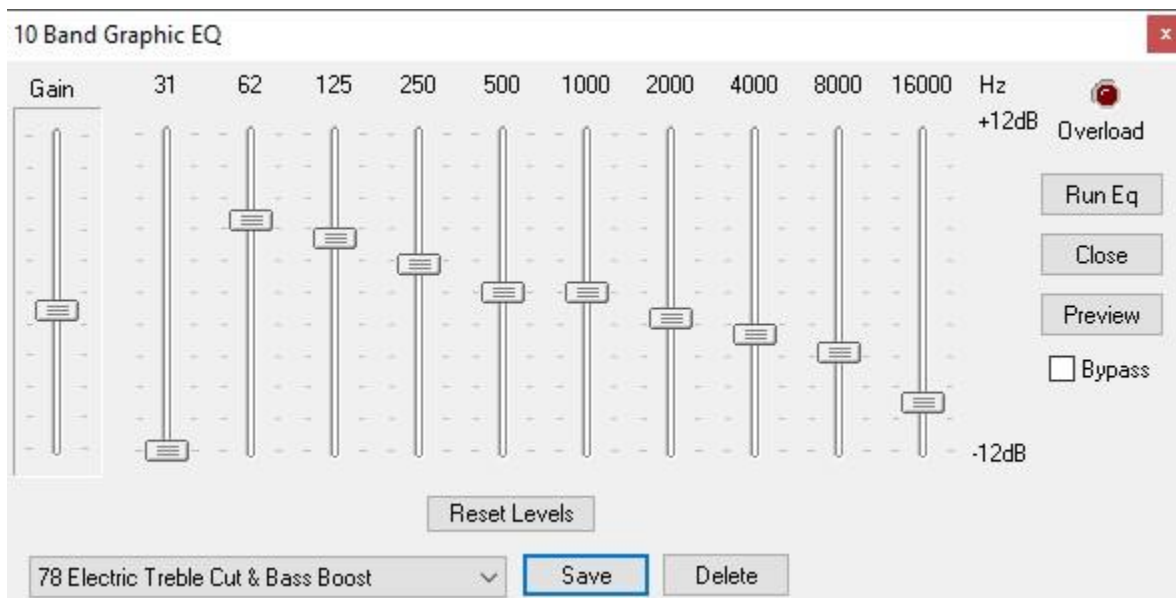


Figure 16-1 Type Two Enhancement Example

The settings reflect a sound that is correct to your hearing. Note that instead of decreasing the higher frequencies, you could boost them for dull sounding LP/45 Records.

16.2 *Diamond Disc Electric Enhancement*

The Edison Diamond Discs that used electric recording, benefited from a wider recording frequency range than the Diamond Disc Acoustic recorded records. The resulting enhancement multfilter has settings that will reflect this wide range.

The Edison Electric Recorded Diamond Disc technology used a different recording method than the new Western Electric technique. The Western Electric recording method was specific for the lateral or horizontal stylus motion in contrast to the use of a vertical motion for the Edison Diamond Discs records. Little information is known about the technical details that Edison used for these records. The multfilter here was developed using *trial and error* methods to achieve a natural sound *when played back on today's equipment*.

During the brief time that Edison produced these records the quality of the recordings varied with some having a strong bass while others were thin sounding. Many of the records contained large amounts of low frequency rumble that were close to the desired music frequencies since the phonographs used for playback were only mechanical and could not reproduce these low frequencies. Edison did develop an electrical playback phonograph with EQ at the very end of the 1920's, but shut down all music production shortly afterwards. Because of the large variation in sound each record may need unique settings.

16.2.1 *Diamond Disc Electric Enhancement Method*

The Edison Electric Recordings reproduce new low and high frequencies from the previous Acoustic Recordings so the method described in 15.0 will be modified for use with these records.

The Edison music engineers knew that the Edison phonographs at the time could not reproduce many low frequencies below 200 Hz so that the presence of low frequency noise was not a problem for them at this time in the 1920's. However, since we can reproduce the inherent noise today this noise will be heard. An additional complication comes from the fact that music has many frequencies at the same location as these rumble frequencies so that a simple removal, as was done with the Acoustic Diamond Disc records of all the music frequencies below 150 Hz, would degrade the quality of the song. The first step before enhancement is to remove any *specific noise frequencies* that were in the original recordings.

16.2.2 Check for Strong Discrete Rumble Frequencies

The first step is to playback a Level Two recording and use the spectrum analyzer to check for any significant low-end rumble frequencies. The following figure 16-2 shows noise at 87 Hz and 123 Hz that was found on this Level 2 recording. This noise was reduced some in the previous Part Two Multifilter for the Electric Diamond Disc records. However, there is music information in the same low-end area so we cannot remove these rumble frequencies during the Level Two file creation.

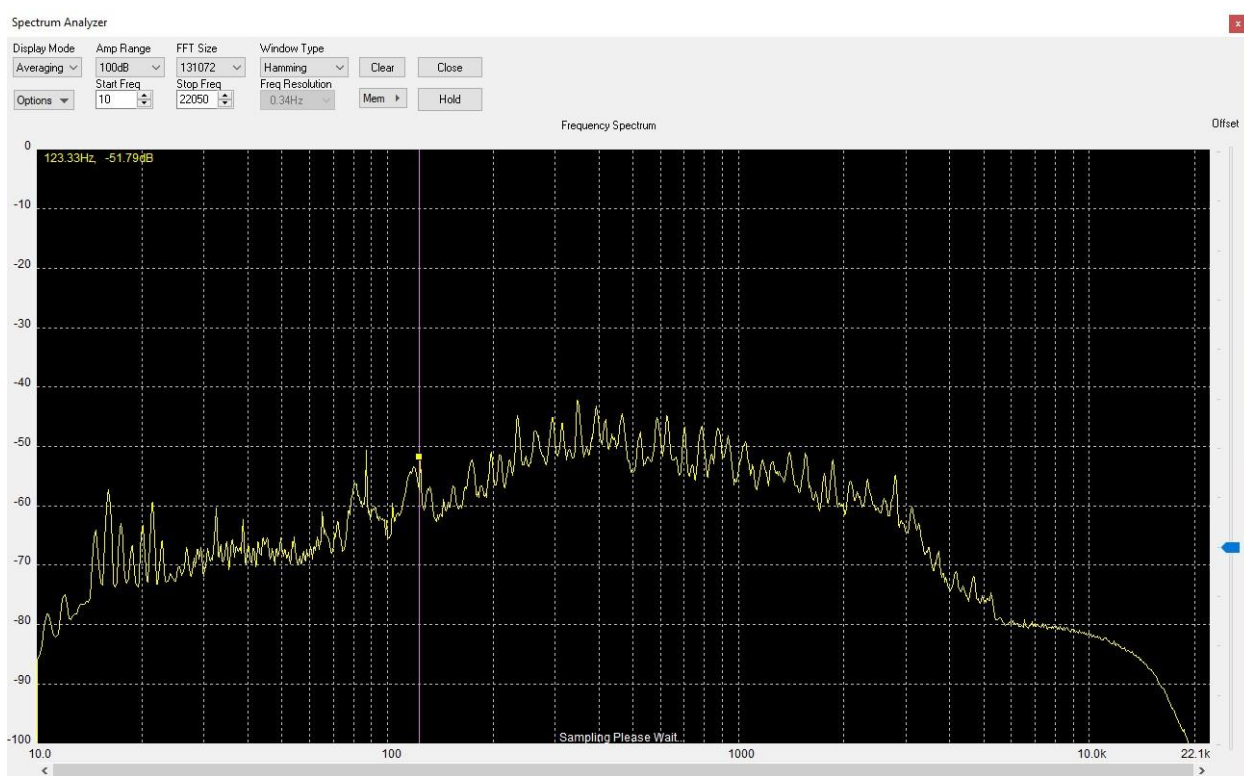


Figure 16-2 Specific Rumble noises.

The marker is located at 123 Hz and another peak at 87 Hz can be seen. The noise at 123 Hz could have been introduced from the initial transfer made at 78 RPM since common 120 Hz noise, when multiplied by the speed increase, 78.26 RPM to 80 RPM, is about 123 Hz. The noise at 87 Hz is from the original manufacturing process. Notice that the general frequencies below 100 Hz have quite a bit of energy that is not from the music.

If we did not remove the rumble noise, then the improved sound from the enhancement process would have a strong steady drone in the song. These rumble frequencies can be removed by using individual notch filters on the Level Two file for any noise that was found.

The Edison Diamond Disc Records at the time that electric recording was used have a wide variation in the noise. It is equally possible to find records that are very *clean* as to find records that have *noise*.

Figure 16-3 shows a notch filter for 123 Hz. The method is to remove the noise and not close music frequencies. The notch filter is the right tool for this.

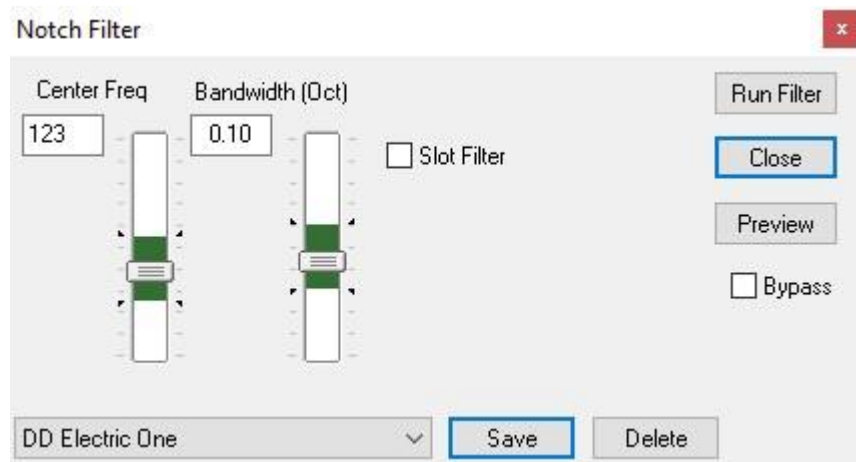


Figure 16-3 123 Hz Notch Filter.

Figure 16-4 87 Hz Notch Filter.

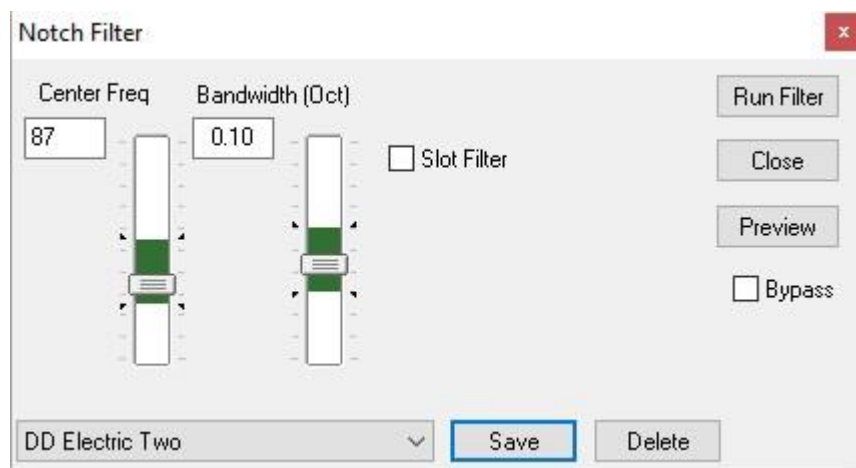


Figure 16-4 87 Hz Notch Filter

The spectrum of the same Level Two music after the application of the two Notch filters is in figure 16-5.

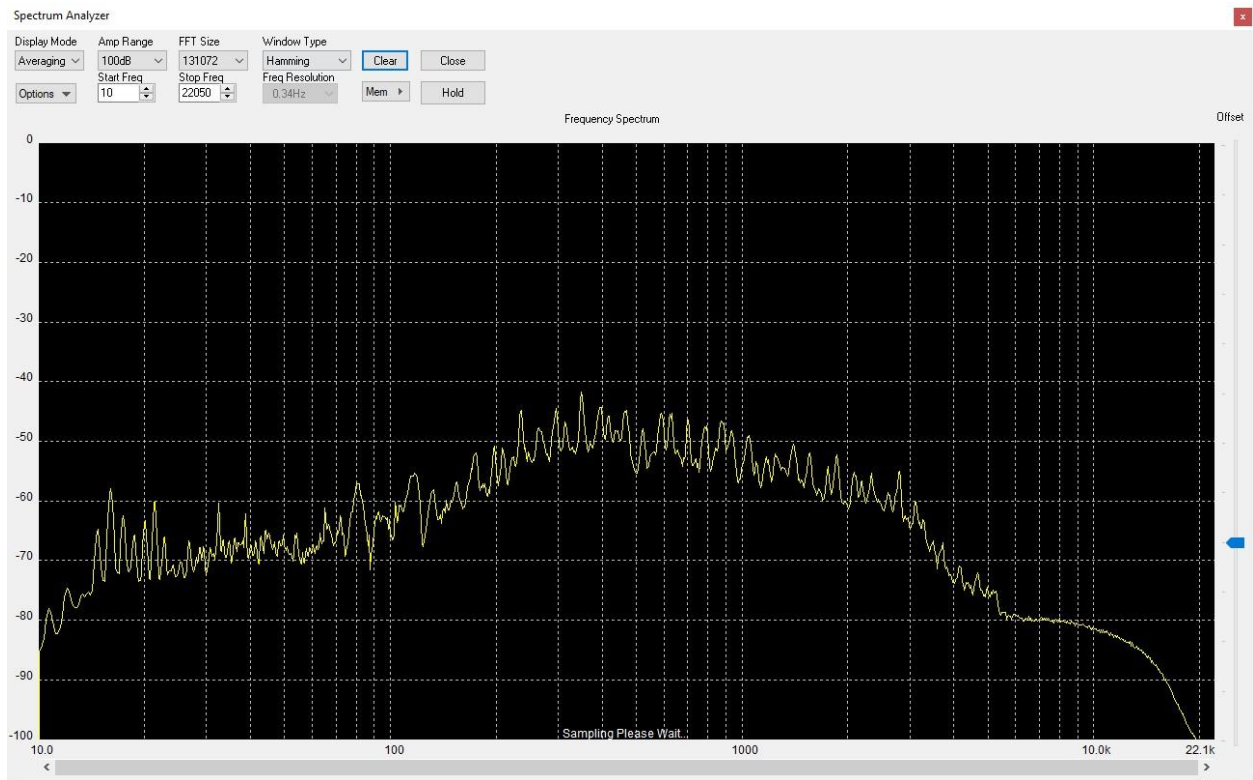


Figure 16-5 Notch Filter Results

The rumble at the previous frequencies has been removed. Any low and moderate level rumble frequencies will be further removed in the enhancement multifilter.

16.2.3 *Electric Diamond Disc Enhancement Multifilter*

Now that strong rumble discrete frequencies have been removed, a Multifilter can be applied to the music. Since a specific EQ curve for these records is not known, various turnover values were tried until a natural sound was found. Refer to figure 16-6 for this multifilter:

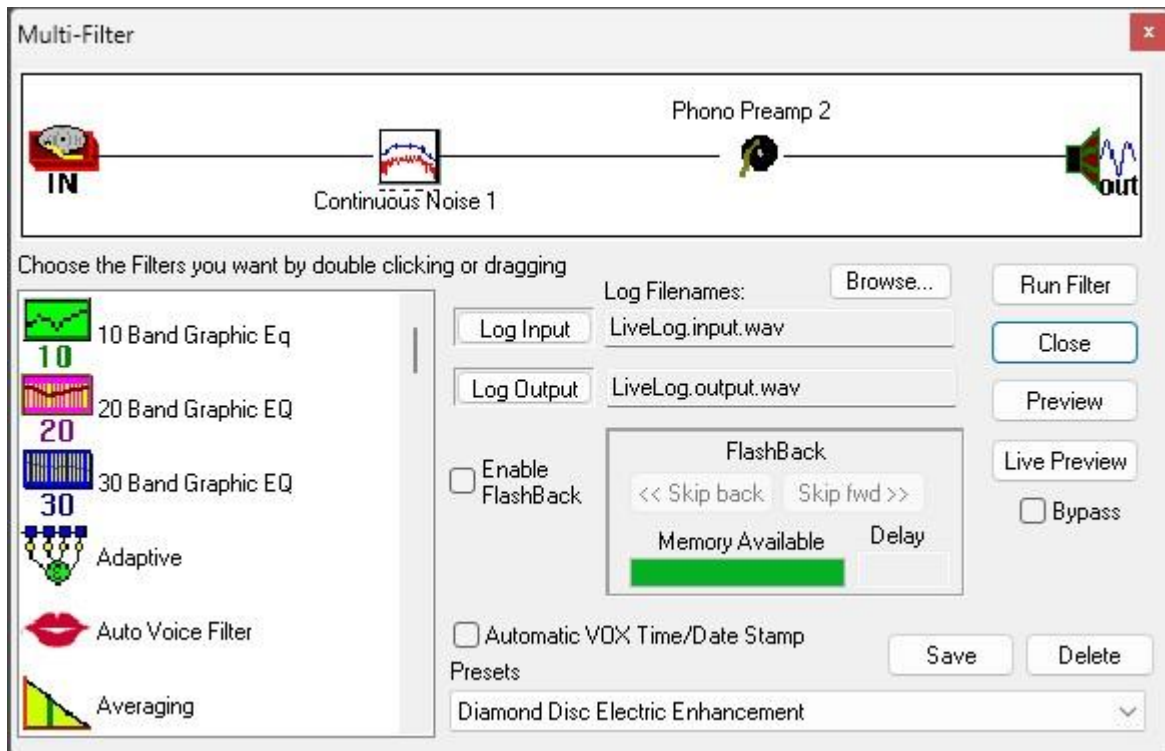


Figure 16-6 DD Electric Enhancement Multifilter

The filter details are in figures 16-7 & 16-7.

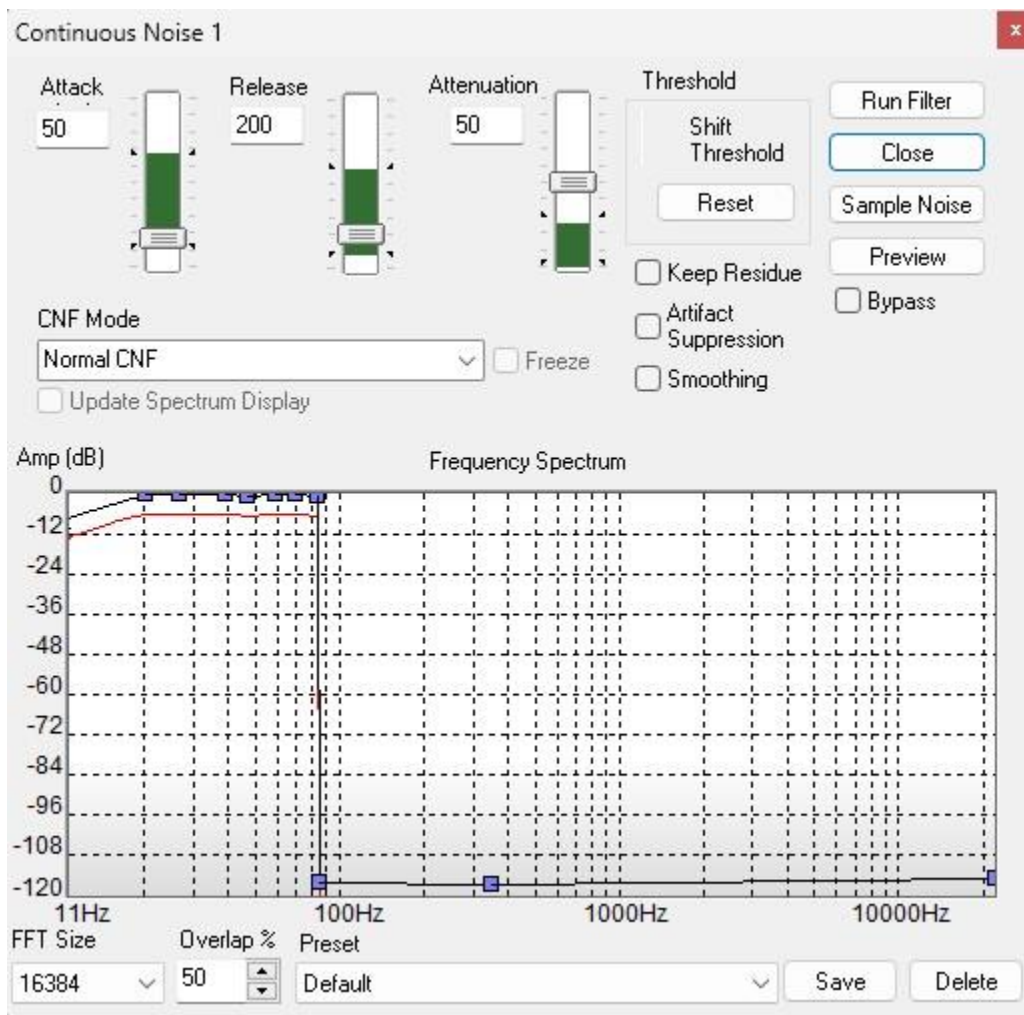


Figure 16-7 Continuous Noise Filter (1) DD Electric Enhancement Filter

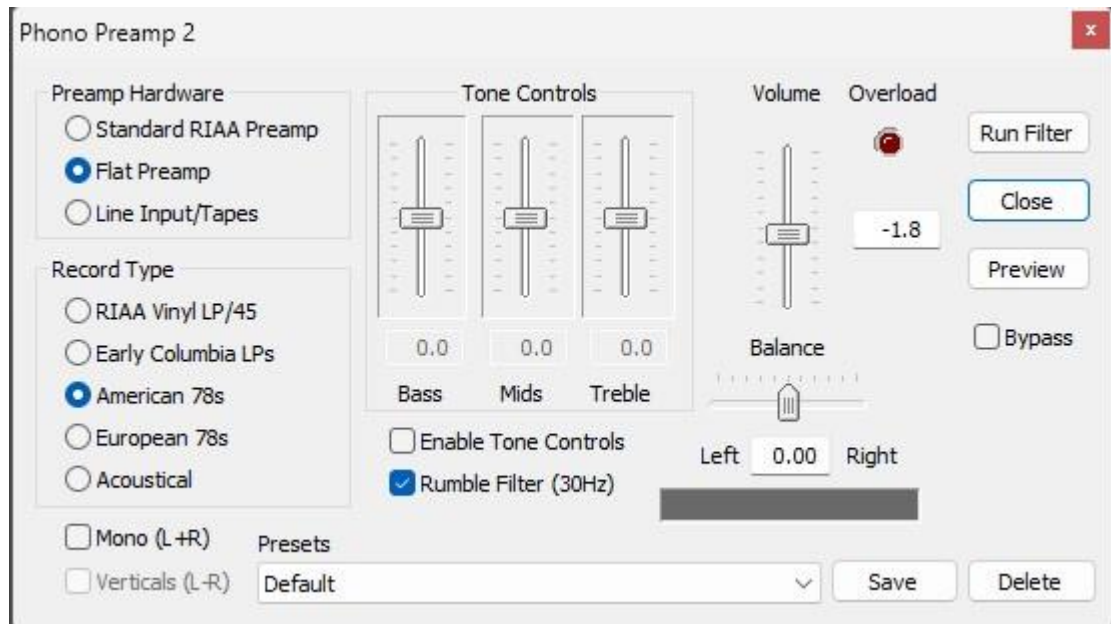


Figure 16-8 Phono Preamp (2) DD Electric Enhancement Filter

The values for all filters can be optimized for each record due to the large recording variation in these Electric Recorded Diamond Discs.

17.0 Third Type Enhancement

This method builds on research work developed by Audio Scientists regarding the best frequency range for the playback of music. One excellent source is contained in the book *Elements of Acoustical Engineering* by Harry F. Olson. Internet searches for his book will find both used Books and *pdf* versions available and should be part of your reference library.

The key points discovered were **First** that the presence of high-order distortion products in the higher frequencies will causes the music to sound un-natural and irritating to the human ear to the point where the music will sound better by lowering the upper frequency limit to the song. For our restoration and enhancement work our playback equipment (modern hi-fi & large speakers) should not introduce any additional distortion products. Therefore, any distortion that we hear in the Level Two music is present in the recording and removal at this point cannot be performed. What is possible is to reduce the high end of the music as much as needed to remove the distortion while keeping a natural sound to the music. If a change to the high-end recording is needed, then a further improvement can be made by the **Second** key point found.

The second key point was that the playback of music sounded best to the human ear if either the product of the range of music frequencies was around 500,000 or the geometric mean of the music's frequencies was close to 800. The overall result should sound balanced between the lows and highs with the balance point around 800 Hz. Some early music, while having low frequencies, does not have the correct amplitude or strength for these low frequencies due to the recording technology used at the time. Check both sides of the frequency spectrum to create a balance point around 800 Hz between equal amplitudes of the low and high frequencies.

These findings were developed after electrical recording had been established and should be used with care when you have acoustical recordings for enhancement. The acoustical recordings were often made using a reduction in the number of instruments in the recording studio to try and make the best recording possible with the technology available. For these records the balance point may be different than that for electrical recordings.

This enhancement can be implemented by using the CNF (Continuous Noise Filter). The CNF filter is an ideal method to control the range of the frequencies that you want to be heard by adjusting the total range via the movable frequency levels on the low and high end. For an example refer to figures 17-1 and 17-2:

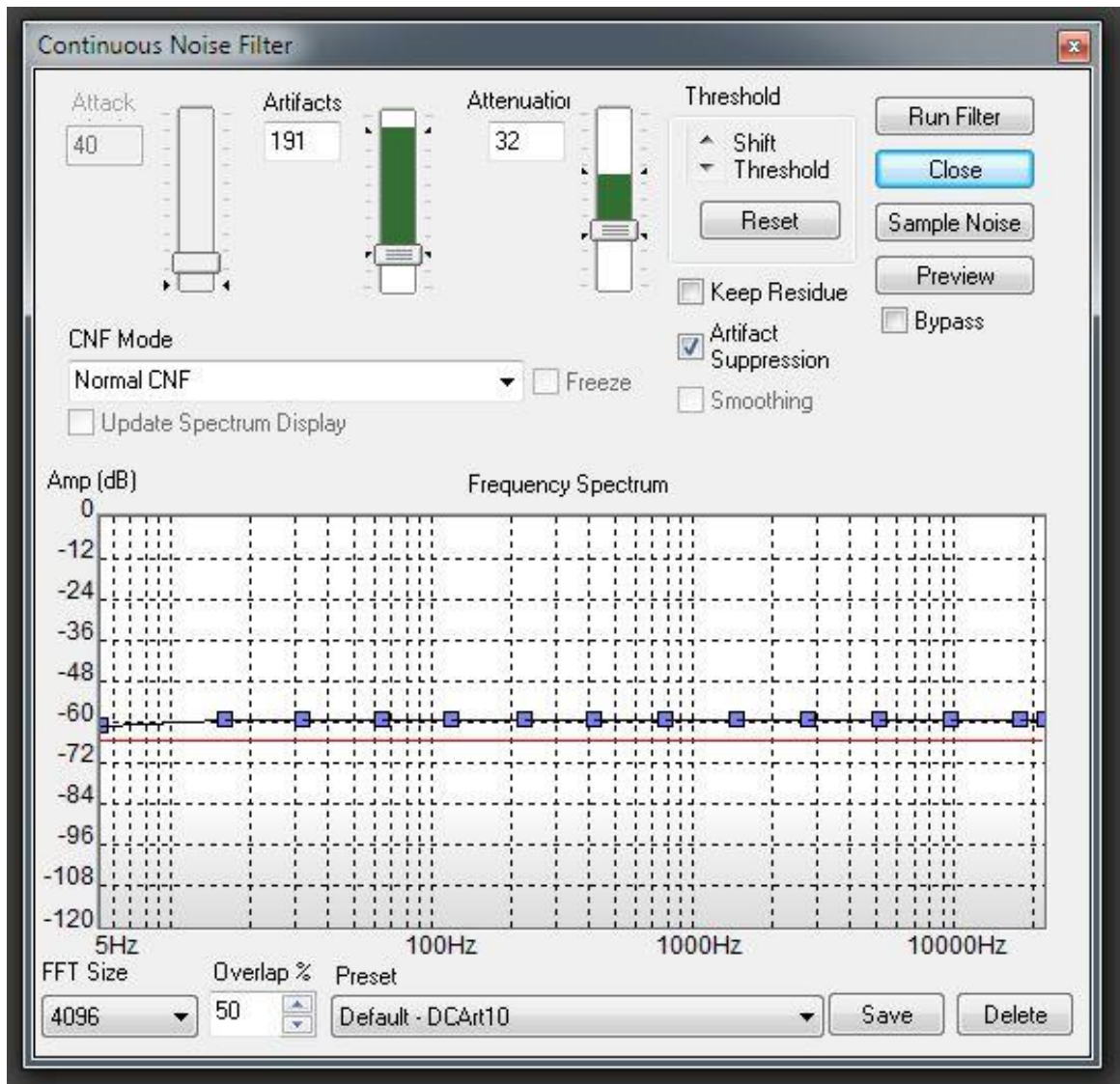


Figure 17-1 Flat CNF

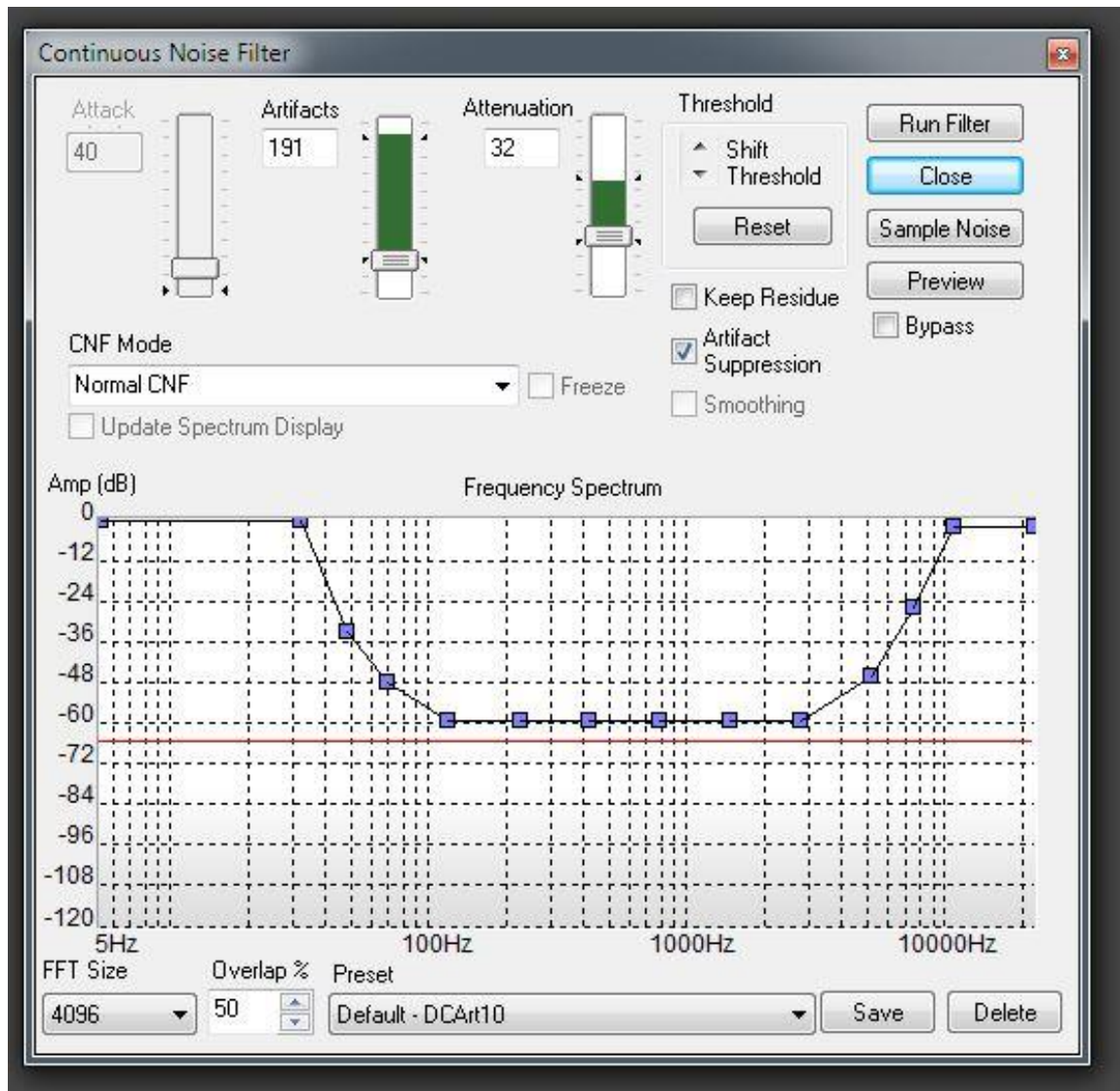


Figure 17-2 Decreased Frequency Range

In figure 17-1, the CNF will pass all frequencies with little amplitude change since the bar for change is low. However, figure 17-2 shows a reduced range of approximately 60-6000 Hz set by raising the level that the signal must exceed, to have no amplitude effect. The other settings in the CNF should be adjusted for best effect. The concept demonstrated by figure 17-2 is that an adjustable range to the music can be performed by using the CNF.

18.0 Fourth Type Enhancement

This method allows for your *creativity* for changes to the original sound. An example for 78 Electric Records, that highlights different effects in the DCart11 software, is in figure 18-1. The same method can work for other types of records. The settings can be modified for the best sound for each song.

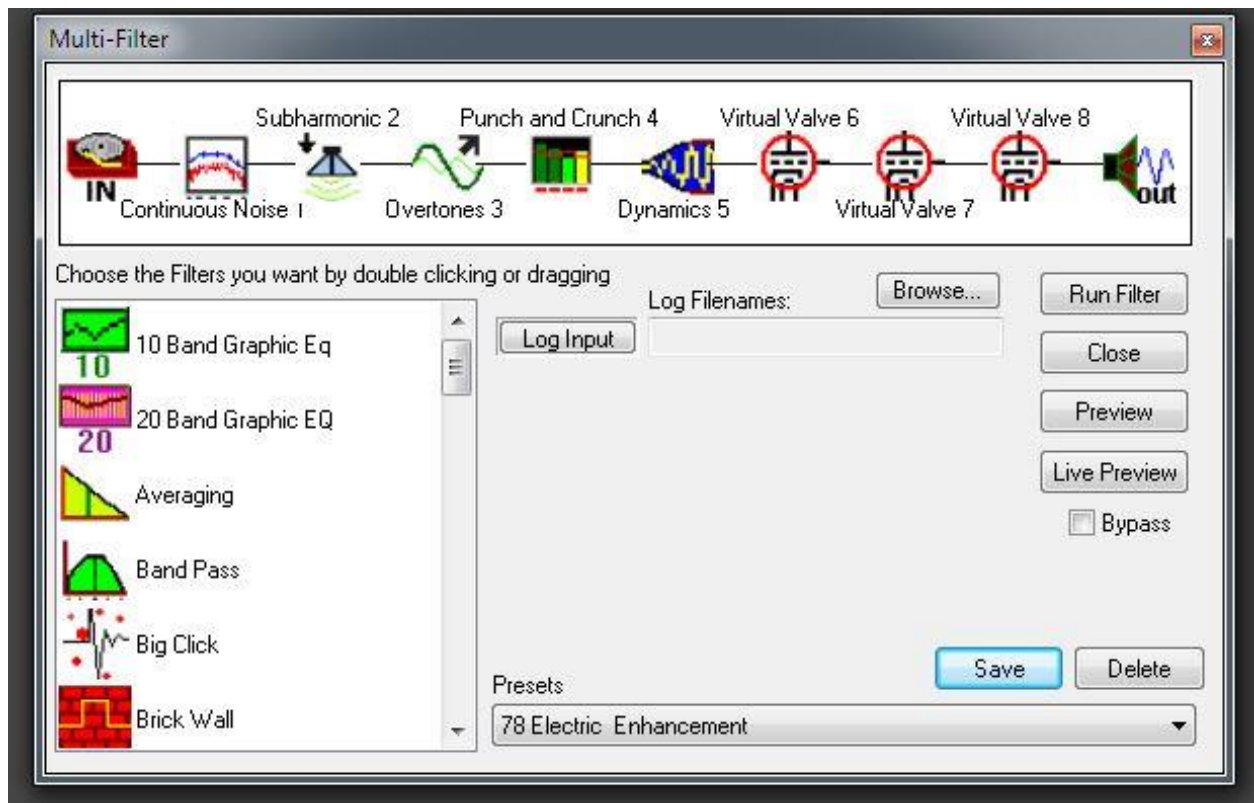


Figure 18-1 78 Electric Enhancement Multifilter

This multifilter is composed of many filters and effects. The order and the resulting processing are important for this filter and by moving the locations within the multifilter the result will be different. The reason that the specific signal processing order is important is due to the non-linear operation of some of these effects and the fact that one stage expects that some specific signal processing has already occurred. You can modify the individual settings later after gaining a good understanding of the operation of each one. The filters and effects are grouped into general sections:

1. Noise Filter. The first filter is a continuous filter that has more aggressive settings than the general one used in Part Two of the 78 Electric Multifilters.
2. The effects labeled two and three provide a method to create new frequencies that are related to those present from the music. Their use can be very powerful in restoring music frequencies that were either not recorded due to early technology limitations or lost due to record wear.
3. Effects labeled four and five are used to expand the amplitude of certain frequencies and then to compress some of the highest levels. While this expand and compress may appear to be *at odds with each other* using them together is the best method to *balance* the overall dynamic range of the music.
4. The last three, labeled six, seven, and eight, can add extra frequencies that are related to the music frequencies present in the music. This relationship to the recorded frequencies is different than the effects in locations two.

Each section of the multifilter, along with the current settings are shown in the following figures 18-2 to 18-9:

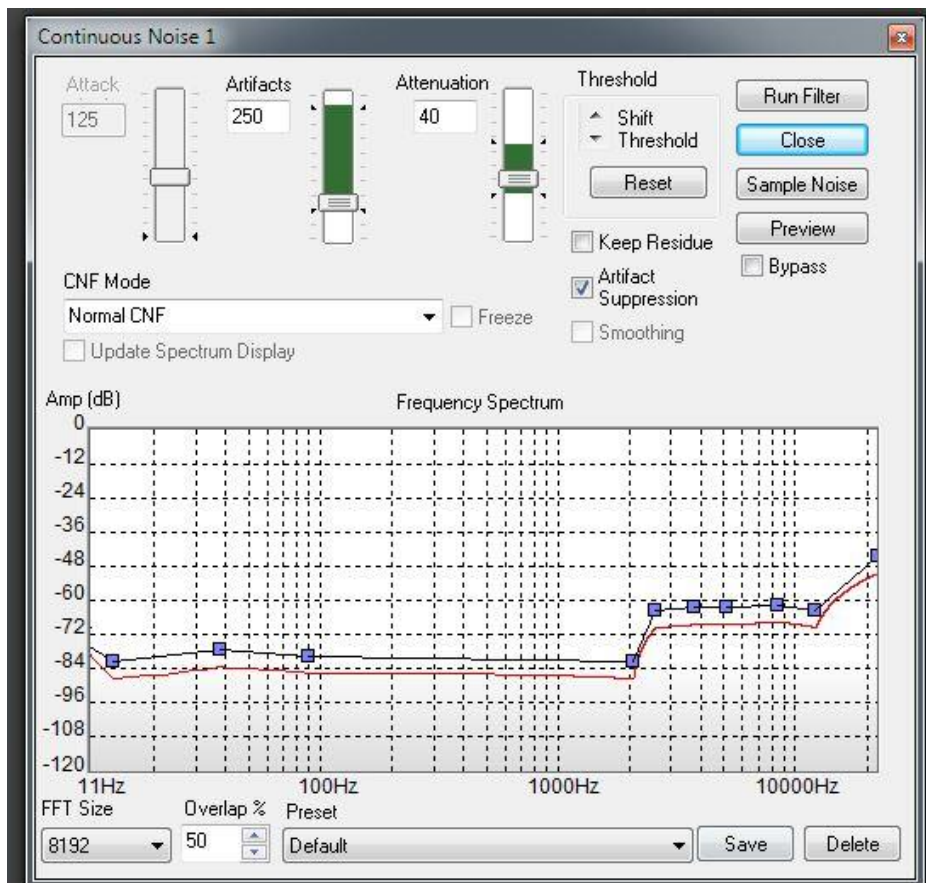


Figure 18-2 CNF (1) 78 Electric Enhancement

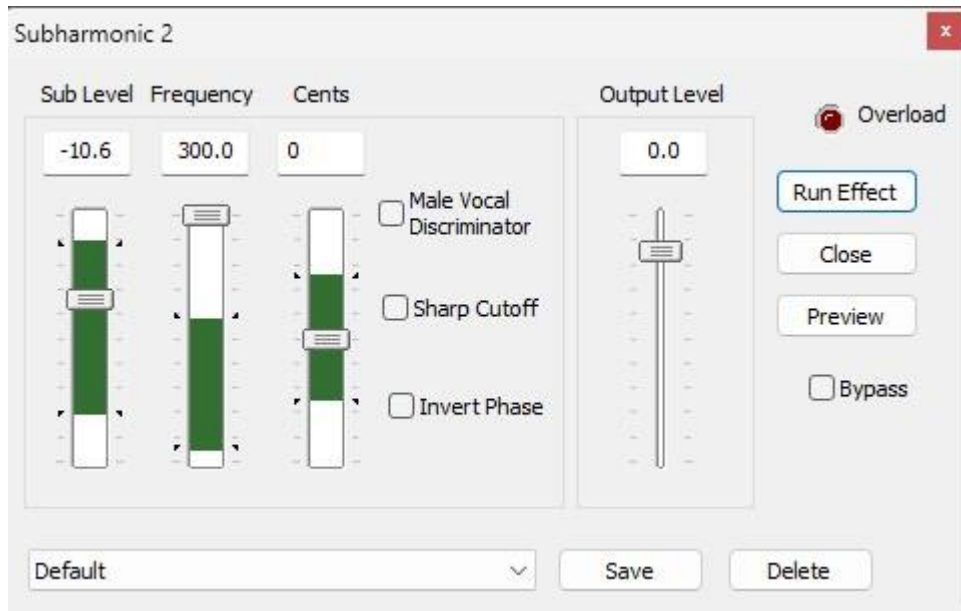


Figure18- 3 Subharmonic Filter (2) 78 Electric Enhancement

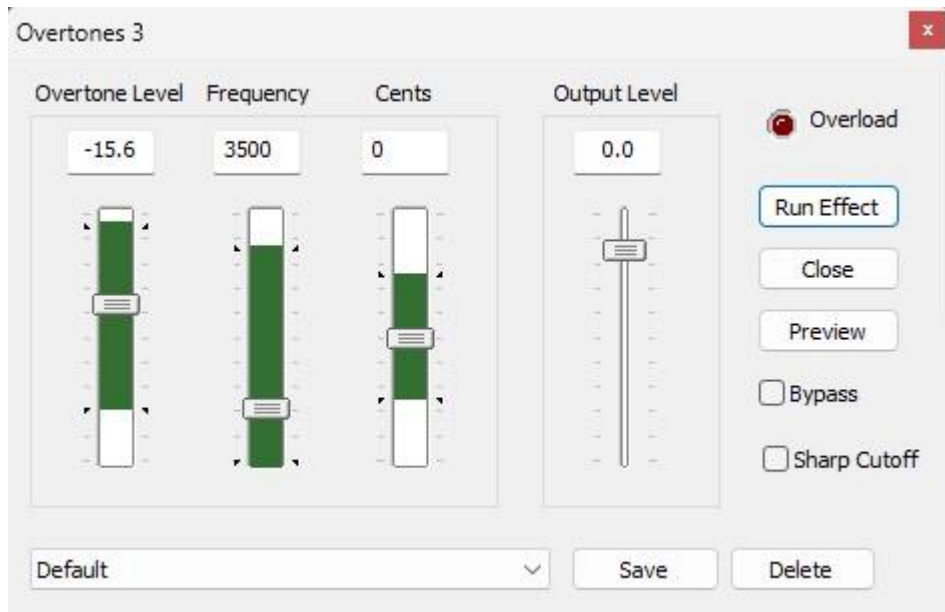


Figure 18-4 Overtones (3) 78 Electric Enhancement

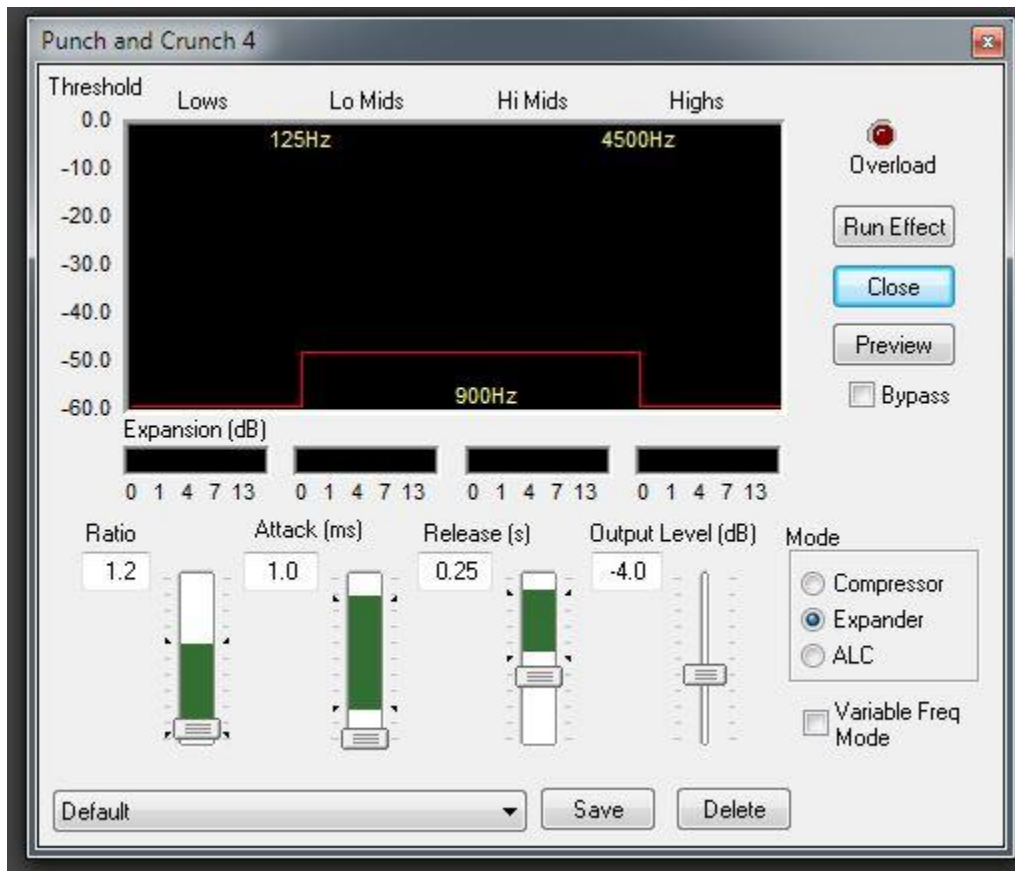


Figure 18-5 Punch & Crunch (4) 78 Electric Enhancement

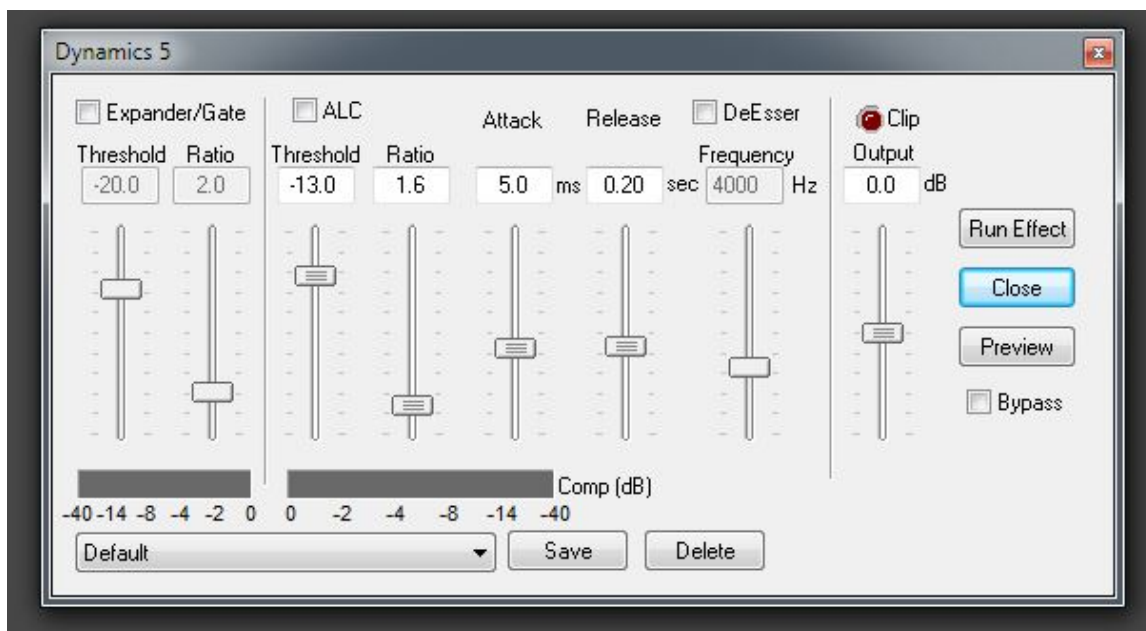


Figure 18-6 Dynamics (5) 78 Electric Enhancement

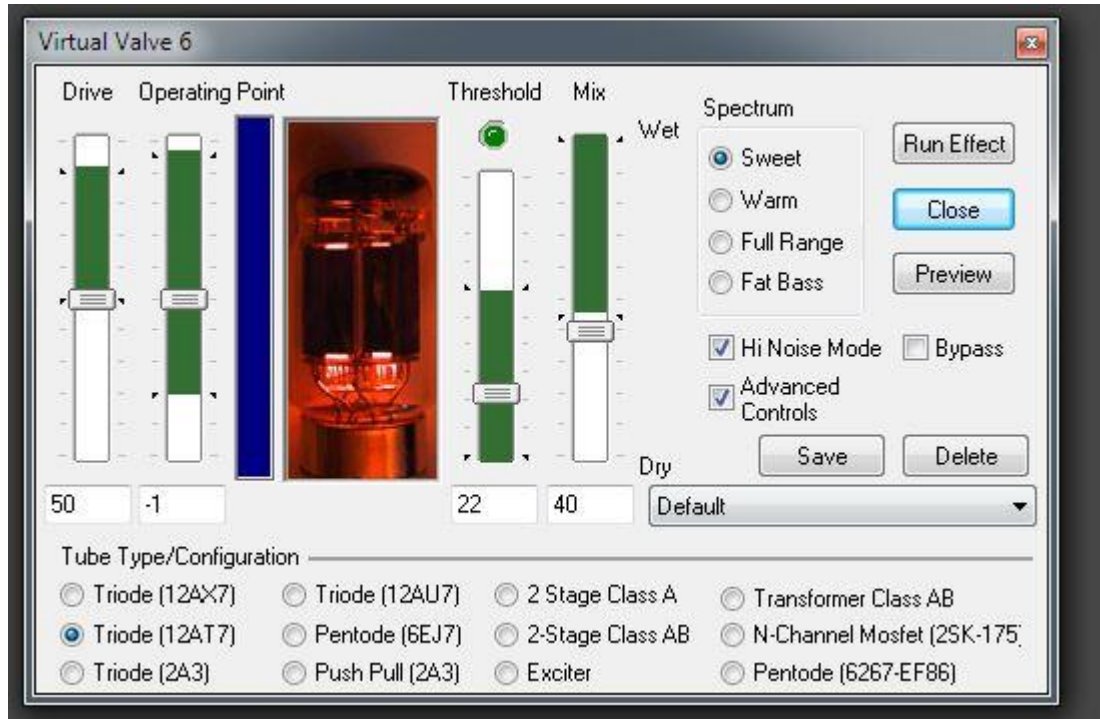


Figure 18-7 Virtual Valve (6) 78 Electric Enhancement

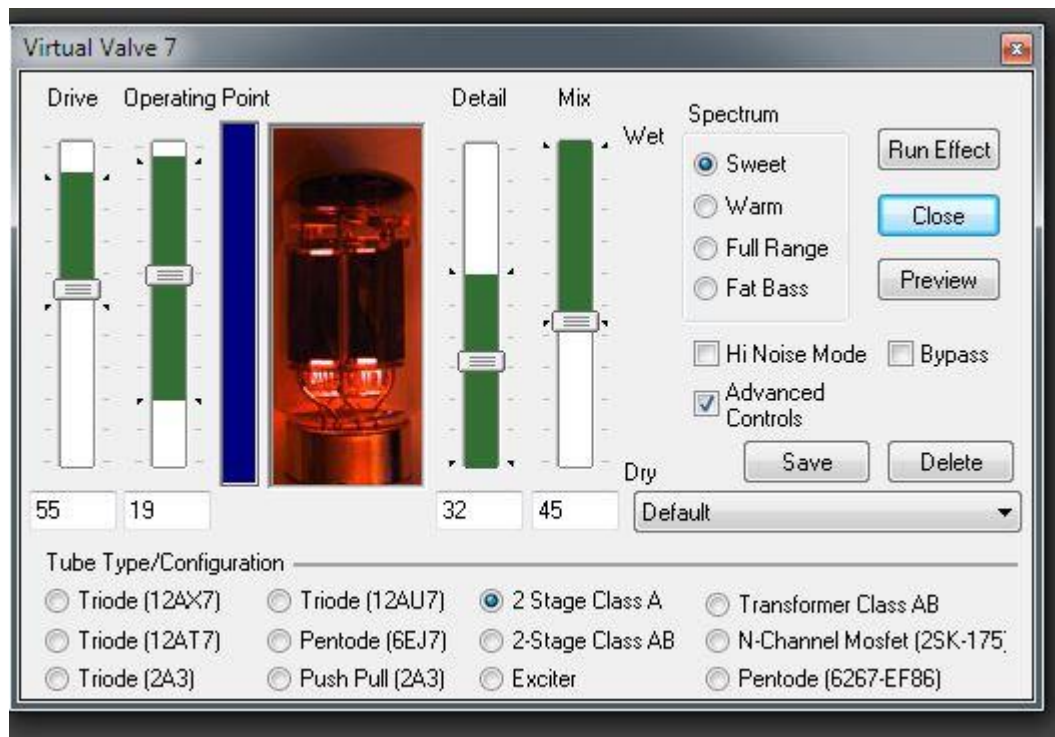


Figure 18- 8 Virtual Valve (7) 78 Electric Enhancement

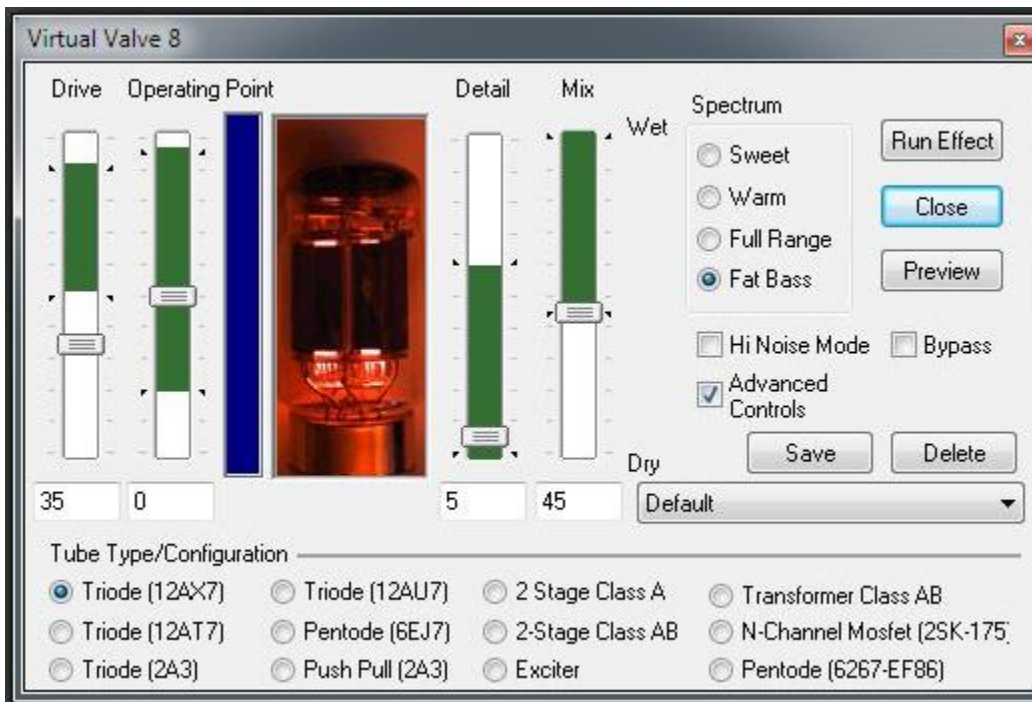


Figure 18-9 Virtual Valve (8) 78 Electric Enhancement

18.1 Operation and Flow of the Enhancement Features

The overall music flow is from left to right within the multfilter. The music enters the CNF where the specific range of frequencies to be enhanced are set. After the music has had as much noise removed, without adversely affecting the quality of the song, the signal enters stages that will add frequencies to the music. The multfilter can be viewed as first removing noise and then adding to the music.

Both sections two and three increase the frequency content by adding frequencies to the music that are harmonically related to the music. Help files in DCart11 provide additional information for these effects ,2 & 3. The music frequencies are then selectively increased in section four greater than they are in the original music (punch effect). Section five is then used to smooth out or limit any signal levels that were boosted too much. The last sections help to add in various harmonics related to the song in a manner that has more control over the music than simple bass and treble tone controls would.

19.0 Enhancement Type Five

19.1 Introduction to this Method

This technique was originally described in an IEEE (Institute of Electrical and Electronics Engineers) Spectrum Magazine article (February 2017). This chapter contains an expanded version of the method tailored to the restoration methods and software described in this Handbook along with recent updates.

I have enjoyed listening to early recorded music on Period Phonographs. The sound from these antique phonographs, while enjoyable, always lacked many frequencies due to the technology used at the time. For many years I tried to improve the sound by boosting the low and high frequencies using tone controls and then multi-band equalizers. These static methods failed to improve the music but did increase the record noise. I developed this *new method* specifically for Acoustical Recorded Music (Music recorded without any electrical devices) and the results are a significant improvement in the overall *balance* of this new version of the old music.

This method uses a special type of dynamic frequency modification for use on acoustically recorded records by taking advantage of the *Harmonic Relationship of Music from Instruments*.

The purpose of this modification is to bring the overall music experience back to what was originally heard in the recording studio by placing in the song additional new frequencies, that were *present* in the recording studio, but were not *recorded* in the original records.

In this chapter the phrase *Correction* is used to emphasize that this effort will restore the sound to what was heard in the studio.

Most of these recordings are now more than 100 years old. There are no musicians or production engineers available to guide this correction work on these recordings. The best that can be done, is to listen to the results and then use your musical listening experience to select the modifications to the original acoustic recordings. The result should produce a song that sounds as close as possible to a natural sound from the instruments and singers.

The example shown is from an acoustically recorded Edison Diamond Disc; this method can work for other Acoustical Recorded Songs.

A description of the method is in 19.2 and the implementation is in 19.3.

19.2 Description of Correction Method

The method requires several processing steps. The starting condition for the process is the result from a Level Two operation. The Acoustic Recorded Song, *The Charleston*, played by the Golden Gate Orchestra on an Edison Diamond Disc from April of 1925 will be used as an example.

When acoustic recorded music was first heard by the public, the sound from the musicians had been modified by the recording of the studio master and in turn by the mechanical playback of the record used in the period phonographs. By using the results from the transcription, and then removing the noise, the limitations to the Acoustic Process from the mechanical playback have been removed and placed in the Level Two location. What now remains in the music is the result of the original mechanical recording method.

At this point in the correction, the *Level Two* music contains Three general frequency sections. The first section is the Low frequency area from 0 to about 400 Hz which contains noise, some music, and the *remaining harmonic frequencies from fundamental notes that were NOT recorded*. The second section is a sub-division within this Low frequency area from 150 Hz to about 400 Hz which contains music frequencies, both fundamental and harmonics, that had a reduction in their proper amplitude. The last section from 400 Hz up to 6000 Hz (or even higher) contains a good, recording of the studio music. The development of electrical recording was a huge improvement to the music sound, but the late stage acoustically recorded music did a good job at capturing the frequencies from 400 Hz to a higher value.

The *Key* to this Correction method is to create *new music low frequencies* below 400 Hz and to remove the low frequency noise. Music frequencies from 75 Hz to 400 Hz will now contain *both* new frequencies and originally recorded frequencies with the new frequencies being the missing fundamental notes. The area from 400 Hz and lower will become the **Low** frequency section which is merged with the original music.

The method used to create these new fundamental frequencies uses a software algorithm from DCart11 that takes the harmonics of musical instruments and then produces a new corresponding lower fundamental note (that was missing) from these higher harmonics for each instrument.

19.2.1 *First; Generate A Source of Music*

The acoustically recorded songs contain a significant amount of low frequency noise that would not be heard using a Period Phonograph. The same songs' frequency content, when played back on a Period Phonograph (Edison Sheraton Model S-19), is shown in figure 19-1.

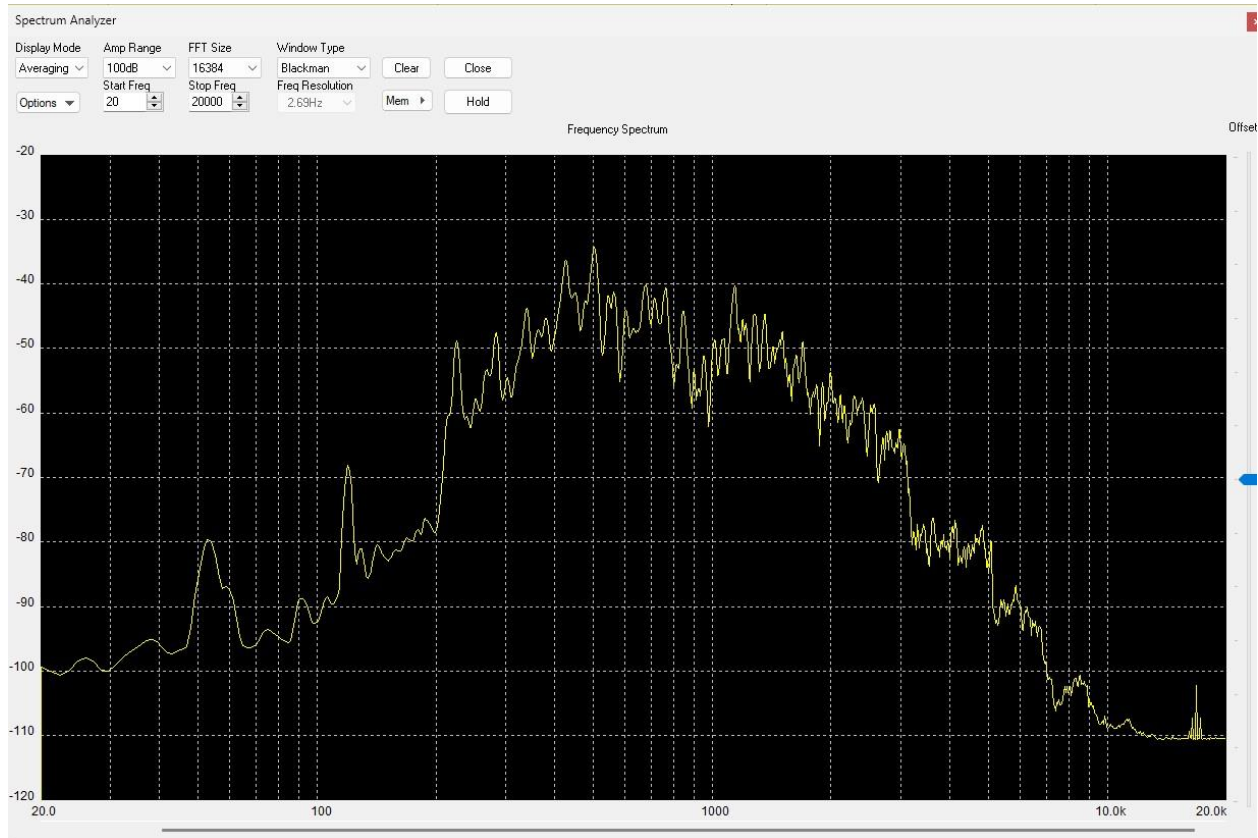


Figure 19-1 Frequency Content of Charleston Song from Period Phonograph

The frequency content of the same song transcribed and restored to a Level Two File is in figure 19-2:

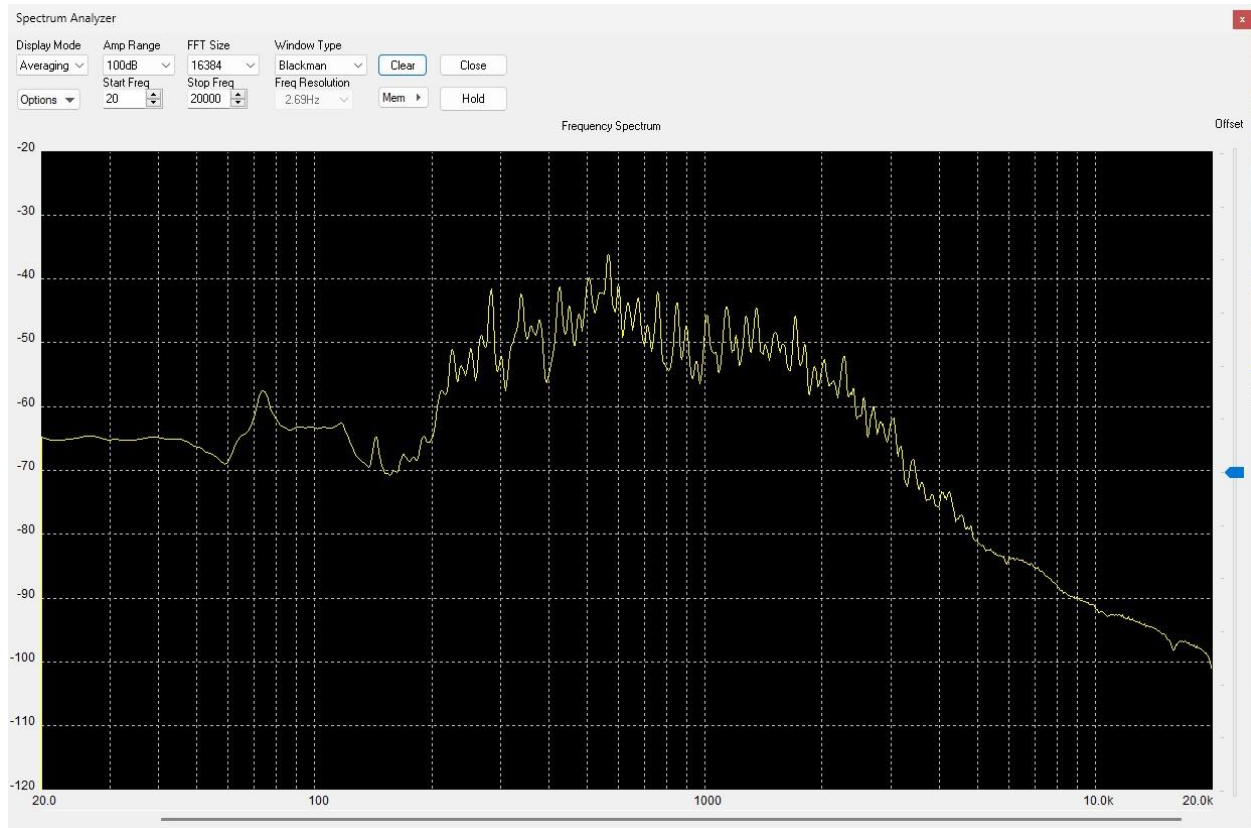


Figure 19-2 Level 2 Frequency Spectrum of Song the Charleston

In this first step, a *New Source File* will be generated which is the result of the Rumble Removal process using the First Type of Enhancement, described in Chapter 15 for the Low Frequency noise. This low frequency noise can also be described as having a *Thump* sound when the record is played on today's hi-fi equipment. This cleaned up file will now become the *Source File for generating new frequencies*. This source file will provide both the portion of the music that needs new low frequencies and the portion of the music that was recorded correctly. The results of the De-Thumper (Low Frequency Noise Removal) Multifilter is in figure 19-3.

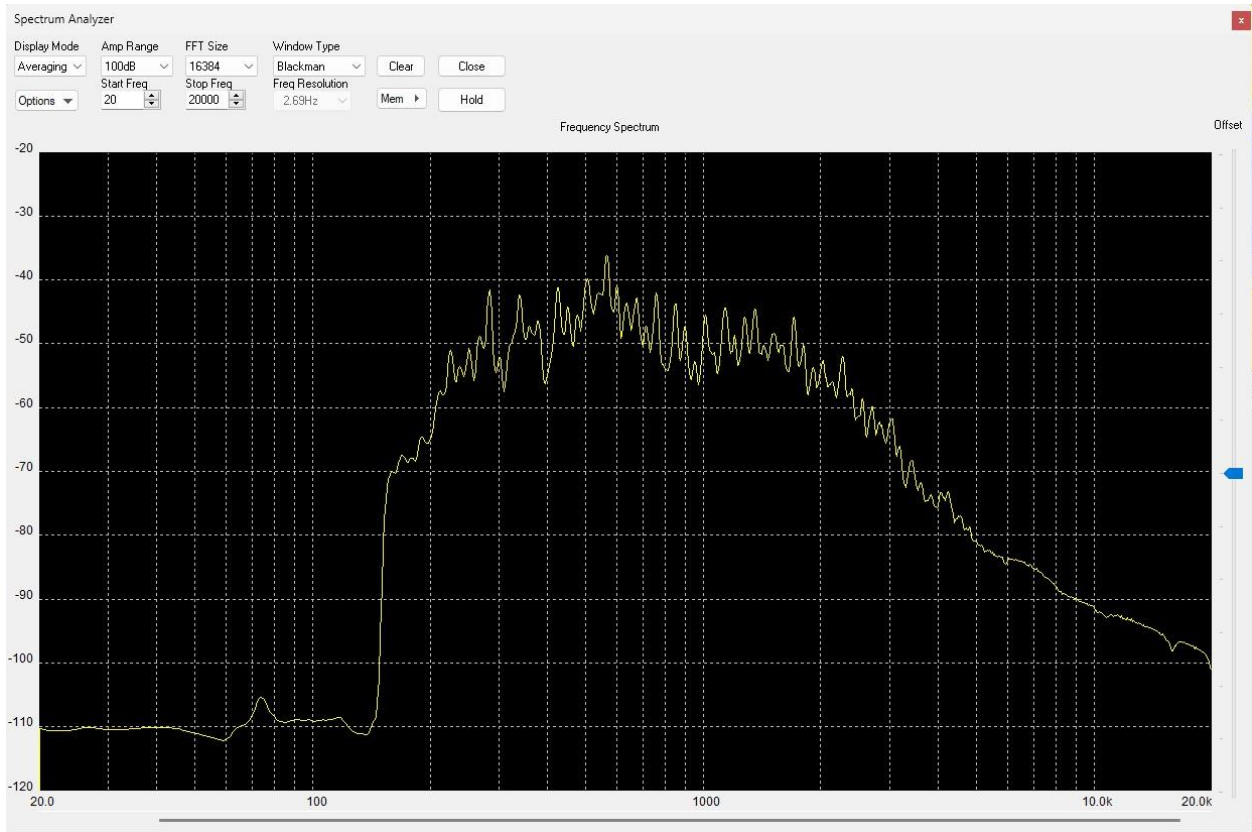


Figure 19-3 Charleston New Source File

19.2.2 *Second; Generate Low Frequencies*

The acoustic recording process had a limited range with the lowest frequencies *recorded*, but *not accurately*, at approximately 150 Hz. Many musical instruments have a fundamental or base note lower than 400 Hz. Music charts and tables will show the various frequency ranges needed for each instrument to be correctly heard. Since Middle C has a fundamental frequency of 261.63 Hz, the fact that Acoustical Recorded songs sound *tinny* is not un-expected. The understanding that instruments have harmonics that are related to the fundamental note of each instrument presents a method for generating the missing fundamental notes via an application of an algorithm in the Diamond Cut Production software program described as a *Subharmonic Synthesizer*.

The operation of this software is that, when a frequency enters the algorithm, a *new frequency and the original will exit*. The *new* generated frequency will be *one-half* of the original value. The software contains adjustable settings that are shown in figure 19-4.

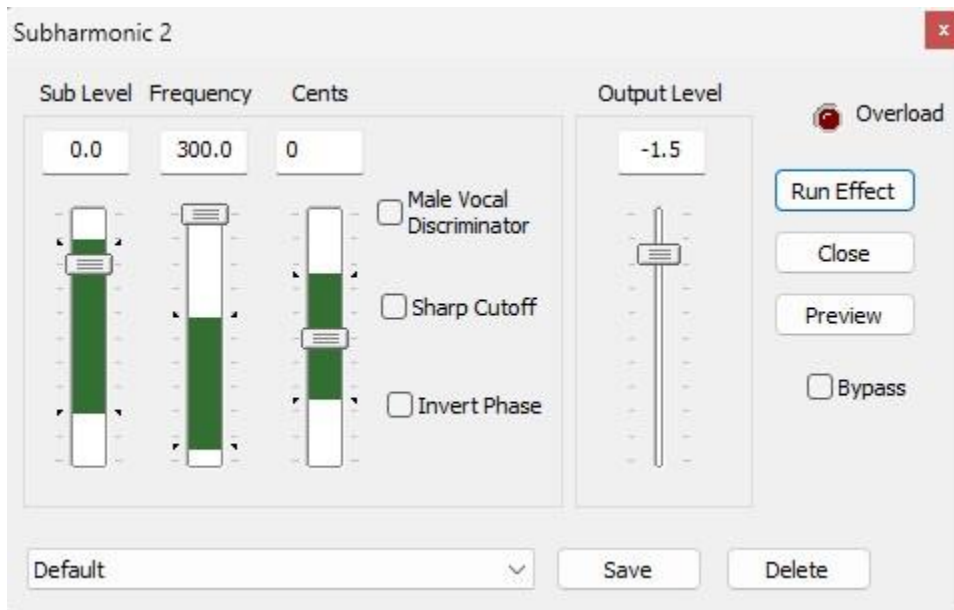


Figure 19-4 Subharmonic Controls

Sub Level controls the amplitude of the new generated frequencies, **Frequency** establishes the general high end of the operation, **Cents** adds fine frequency control, the **Male Vocal Discriminator** helps to make a male singer sound natural, the **Sharp Cutoff** helps to limit the filters bandwidth, **Invert Phase** is not used and the **Output Level** controls the amplitude of the mixed result. To demonstrate this synthesizer, a sine wave of 200 Hz was fed into this synthesizer, using the settings shown and the output frequency spectrum is in figure 19-5:

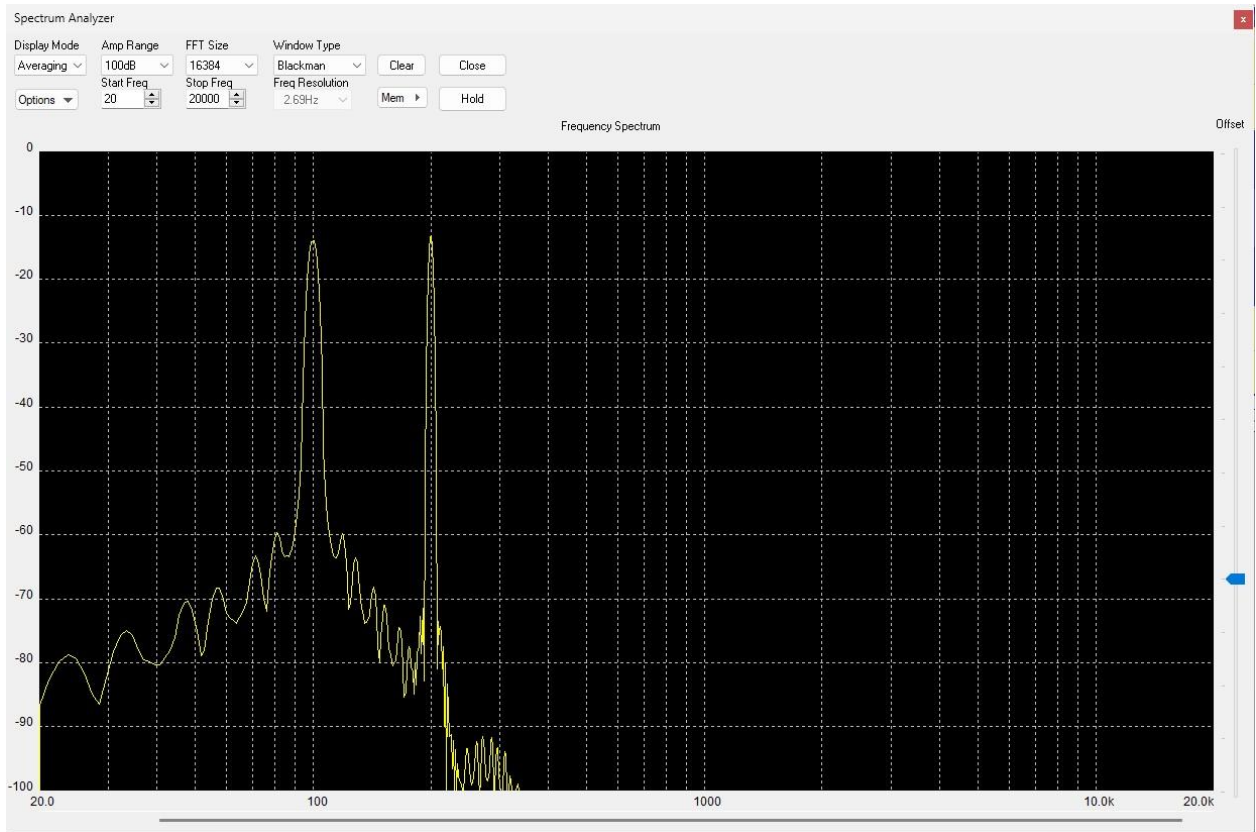


Figure 19-5 A New frequency of 100 Hz along with the original 200 Hz.

The Sub-Harmonic Synthesizer will always produce frequencies that are related to the *input* values in a fixed relationship of one-half (with the cents at 0). If a tuba or bass instrument has a harmonic at 250 Hz then the original fundamental at 125 Hz (which would not be recorded) will now be re-introduced into the music. The potential for a large increase in the quality of the song is possible since you are not simply boosting low notes, rather you are creating the notes *that were originally present and were lost in the recording process but present in the music studio. This is a Dynamic Process in Sharp Contrast to a Static Process that uses fixed Levels of Boost.*

19.2.3 *Third; Merging of Original & New Low Frequency Files*

The source file will enter the subharmonic generator, and the original music and new low frequencies will be merged and leave for more frequency modification.

The songs music does not have a sharply frequency location from *needing*, to *not needing*, new low frequencies. The original acoustic recording process had a frequency range over which the recorded frequencies slowly improved to be fully captured by the master recording. Because of this gradual change-over in the recording performance, the selection of the settings in the sub-harmonic generator can help to create this needed transition frequency range and to make the music sound natural.

19.2.4 Fourth; Cleanup / Final Modification

After the merging of the files further frequency modification is *now* possible since low and high frequencies are present that have low distortion. A slight high-end boost followed by a steep decline will add some brilliance to the music while limiting surface noise. The low end can be boosted because music frequencies are *now present* rather than just noise. The cleanup multifilter (details in the implementation section) is used to perform the final frequency changes shown in figure 19-6:

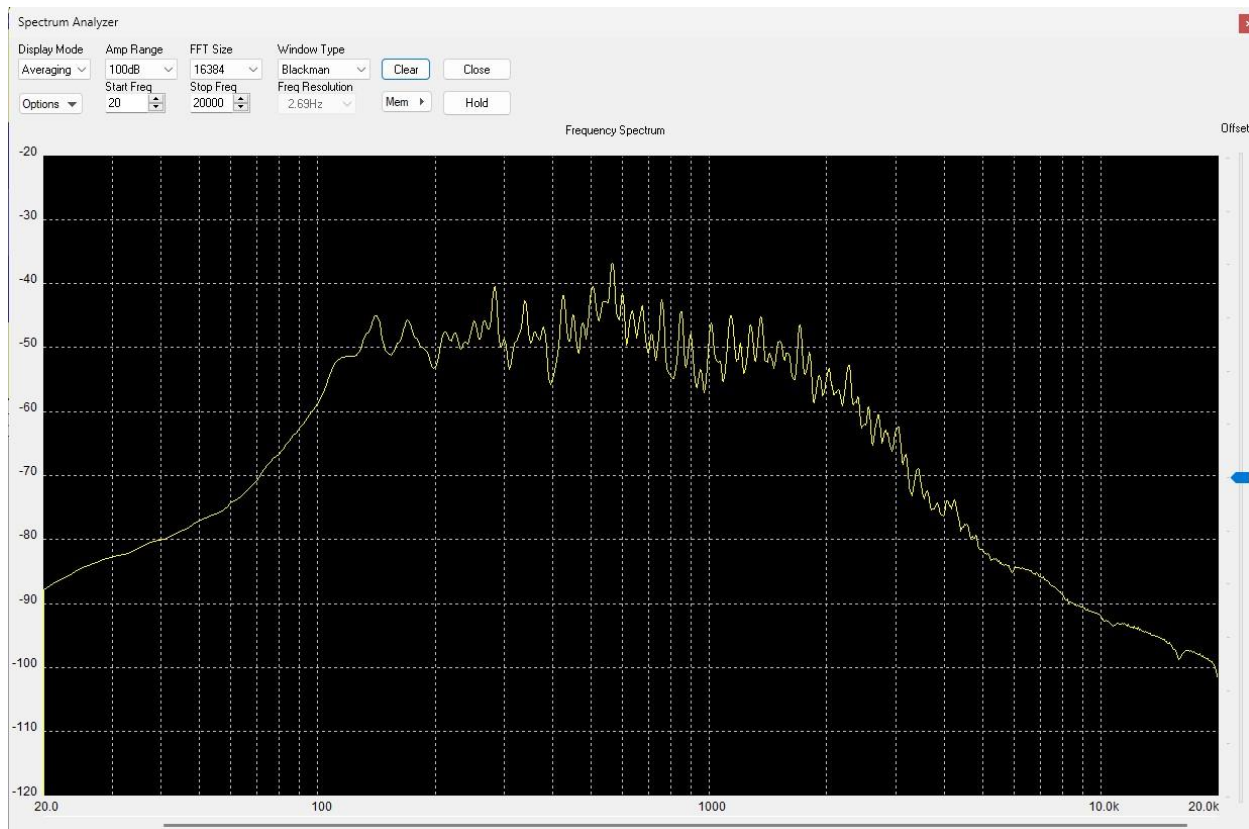


Figure 19-6 Finished Music File.

Figure 19-6 shows the large increase in the music range after using this Acoustical Correction Method.

Figure 19-7 shows an overlay of the finished correction (yellow trace) with the same song as heard on the Period Phonograph (white trace) using the same vertical scales. The finished result greatly expands the overall range of music from the original range heard on the period phonograph.

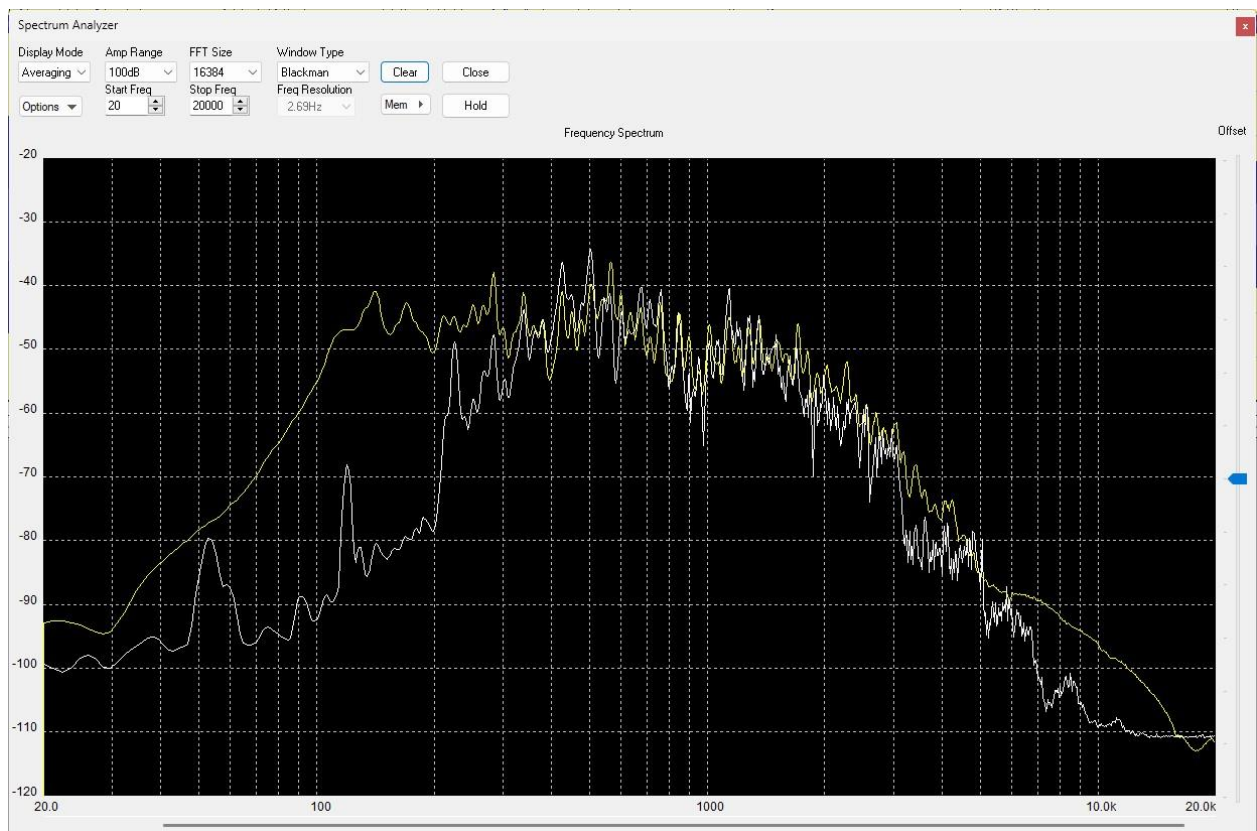


Figure 19-7 Final Correction vs Version Heard on Period Phonograph.

The result should be judged by hearing the two different versions rather than just looking at the spectrum values from figure 19-6. Most people that have heard this *new* version of the Charleston said that the sound has a balanced and full sound to it and enjoy hearing this *new version of an old song*.

The various multifilters used do not have special *intelligence* about what frequencies are created. Users of this technique can experiment and may find that adjustments to the described settings may help to improve the sound for their music.

This technique has also been used on Acoustical Recorded 78 RPM records with similar success. An implementation of the method using Diamond Cut Productions software will now be shown.

19.3 Implementation of the Acoustical Correction Process

The settings shown in the implementation represent *starting values* that have provided good results for many Acoustic Recorded Records. An optimum setting for each song may be needed since songs have unique frequencies and combination of musical notes.

19.3.1 Create A Working Copy

A copy of your Level 2 file will be used as the *working* file to allow a return to the Level 2 work for future improvements if wanted. The extra files that will be created from this working file should be saved with unique names.

Steps in 19.3.2, 19.3.3, and 19.3.4 will be combined into one multifilter. First, create the music with a clean low end.

19.3.2 Create a Clean Low End.

The method used in section 15.0 can produced the needed clean source for generating the new low frequencies.

19.3.3 Create New Low Frequencies

The sub-harmonic synthesizer will generate the new frequencies.

19.3.4 Create New Low Frequencies & Merge with Source

The working file will now be used as the input to an overall Multifilter that removes low end noise, creates new low frequencies, and merges the result into a new file. Details are in figure 19-8, 19-9, and 19-10.

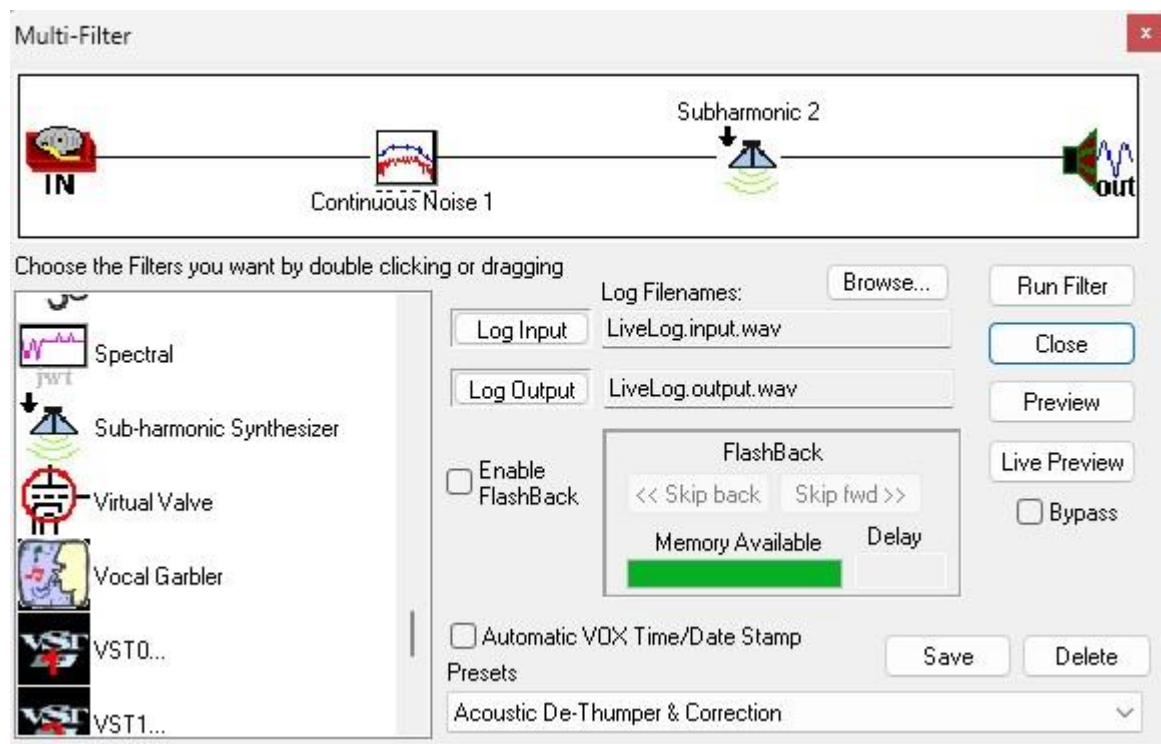


Figure 19-8 Acoustic De-Thumper & Correction

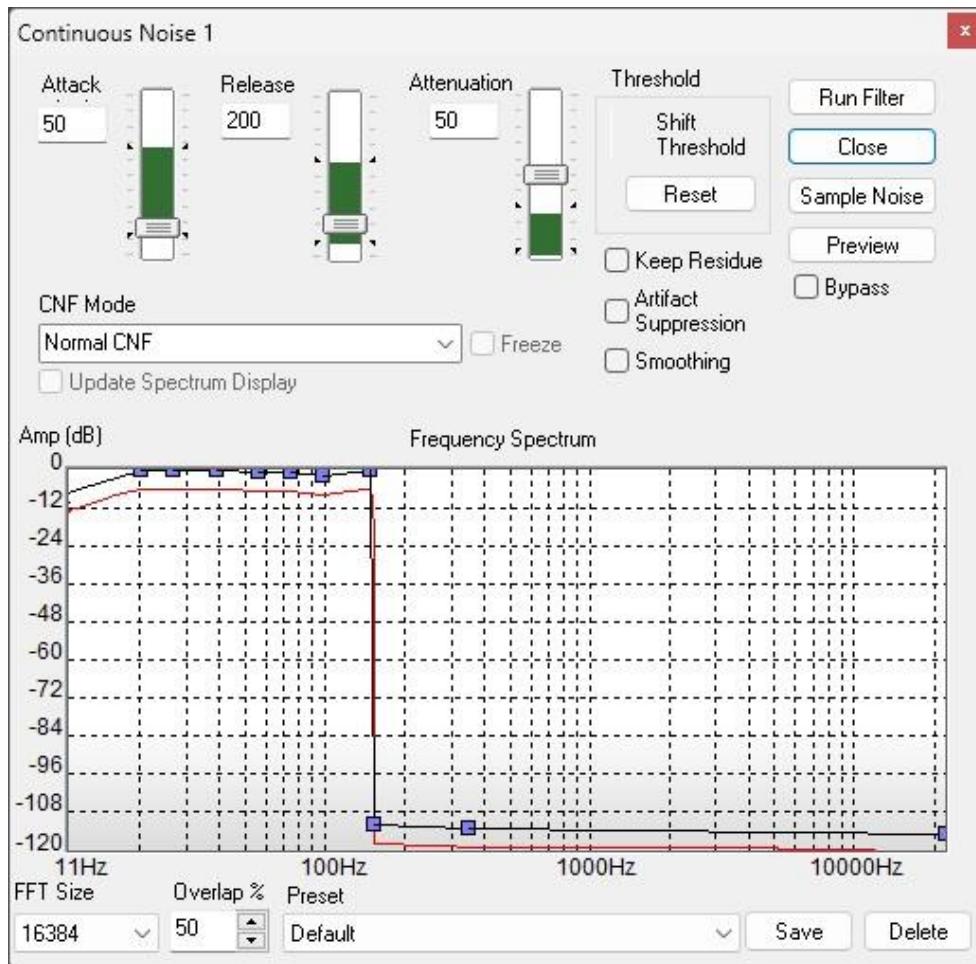


Figure 19-9 CNF (1) Acoustic De-Thumper & Correction

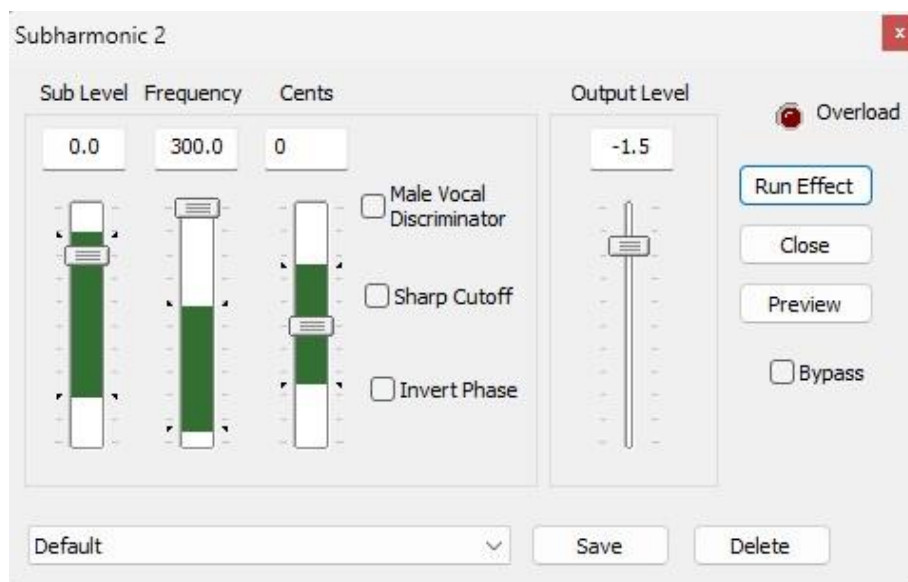


Figure 19-10 Subharmonic (2) Acoustic De-Thumper & Correction

The output of the multifilter will now contain both the original music minus low end noise and new low frequencies. Note that the 300 Hz value in the filter, although shown as an upper limit, is not a sharp cutoff. The subharmonic generator will continue to work above 300 Hz, but with gradually less amplitude on new frequencies if the Sharp Cutoff or Male Vocal Discriminator options are not selected.

19.3.5 Create the Final Cleanup Version

The last operation is to send the merged file through the cleanup multifilter to add some bass and treble and to eliminate any high frequencies that are not music but just surface noise. Refer to the Cleanup Multifilter in figure 19-11:

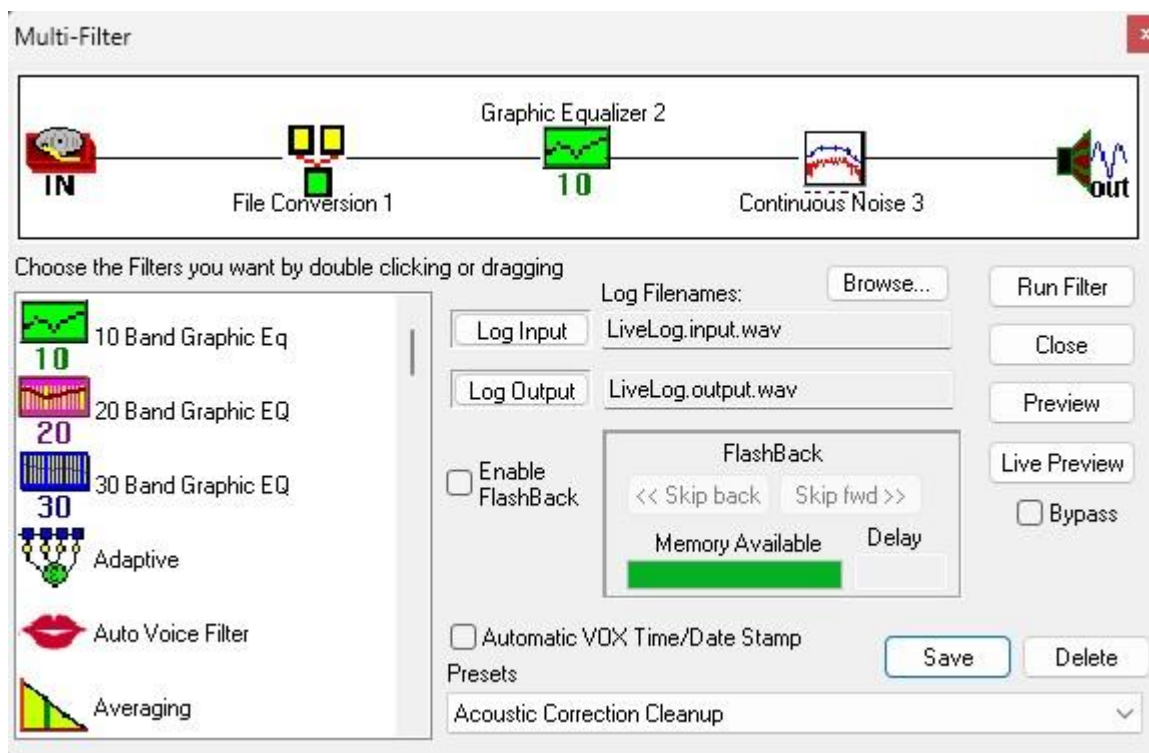


Figure 19-11 Acoustic Cleanup Multifilter

The first part of the multifilter uses the file conversion software to reduce the overall amplitude so that any boosted frequencies will not saturate the signal. Figure 19-12 shows the settings.

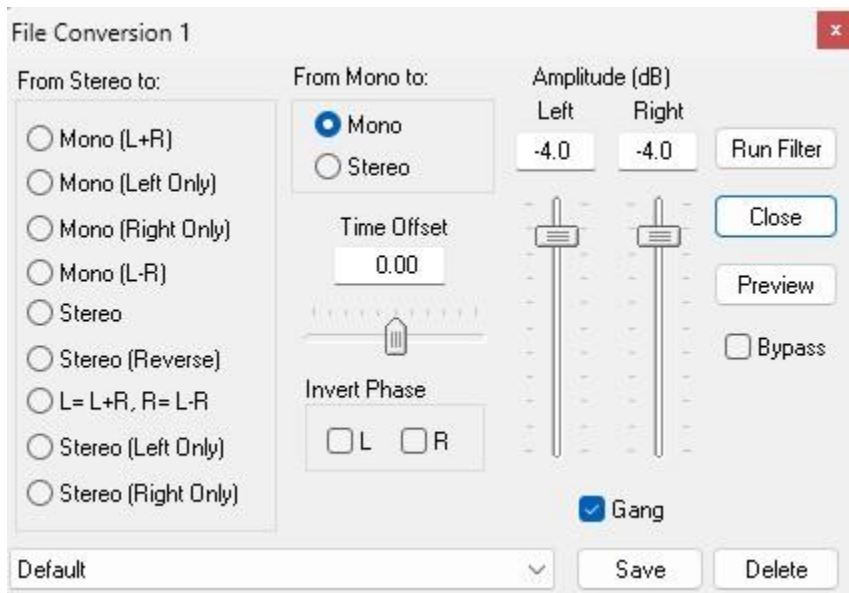


Figure 19-12 Stage 1 gain change in the multfilter

The next section modifies the low and high frequencies by boosting them *now that music frequencies are present*. Refer to figure 19-13 for the details:

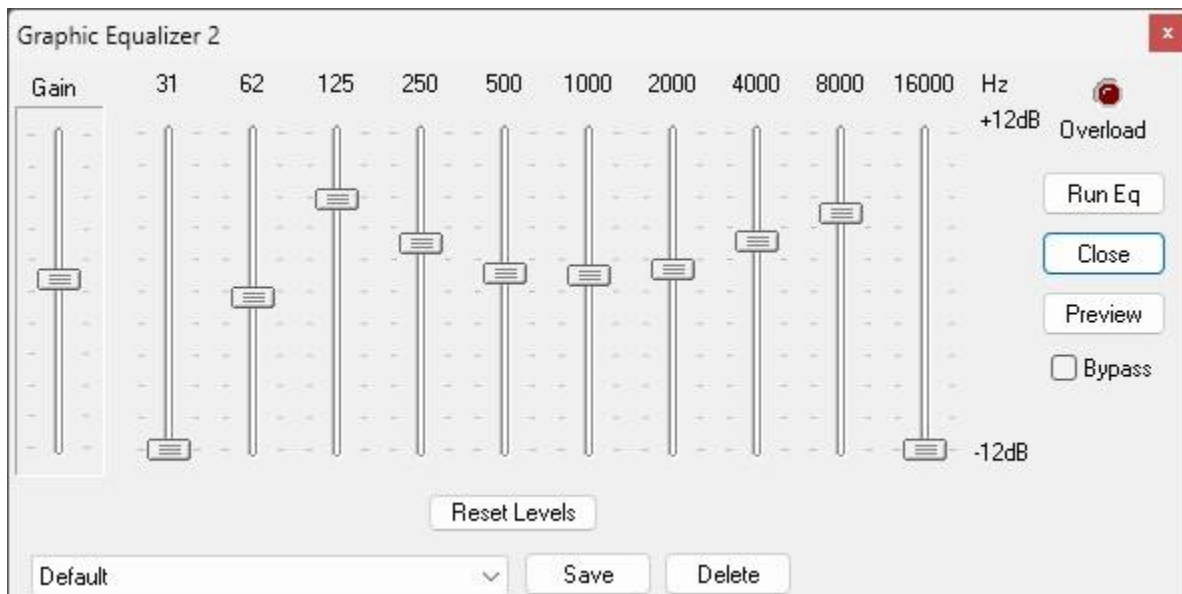


Figure 19-13 Stage 2 frequency changes in the multfilter

The last filter removes any left-over noise that does not contain music. Refer to figure 19-14.

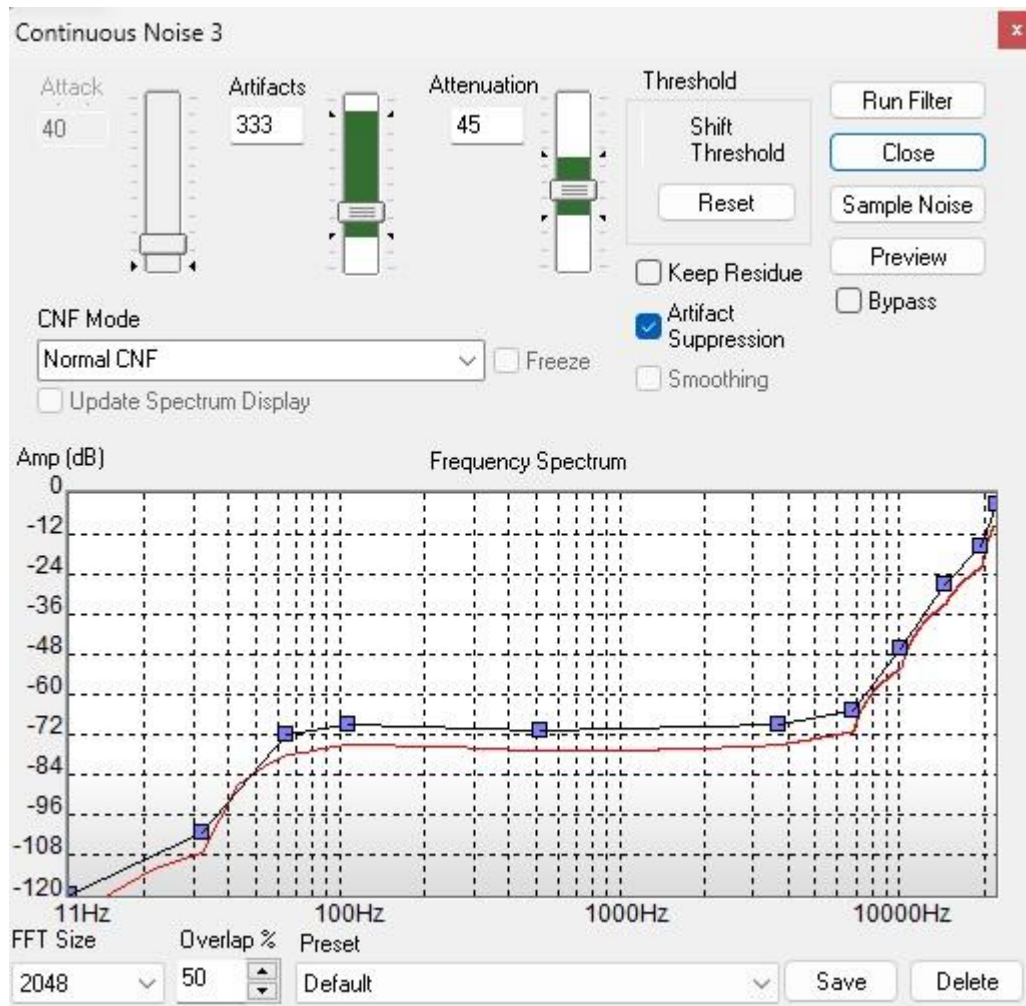


Figure 19-14 Stage 3 CNF filter.

19.6 Conclusion

The purpose of this newly developed method is to restore the music from the record to what was originally heard in the music studio when the recording was made. This is a unique method I have developed and has been long been a pursuit of mine to bring more life and music out of these unique records.

These records preserve an important time in the history of music. This modification to the music should make it sound as if you were in the studio when originally recorded and in turn appeal to a wide audience of music lovers.

Reference Section

The reference section contains additional audio information for your restoration work. The main sections of the handbook contain information that would be used often and the reference section contains additional information that provides technical insight into audio engineering.

As you venture into this wonderful hobby of music restoration you have many technical sources available to add to their knowledge. This section is a beginning step into this learning journey.

20.0 *History of Recorded Sound*

The development of recorded sound has evolved from using cylinder shaped records to the later vinyl long playing records, containing multiple mixes of instruments and singers to create a two-channel stereo record. This chapter contains a brief look at this fascinating subject.

20.1 *Beginning of Recorded Sound*

The first records made for sound recording used mechanical methods for both recording and the playback of sound. The original phonograph patented by Thomas Edison in 1878 used a cylinder made of soft tinfoil to hold the music vibrations. As the music was being recorded the sound waves moved a recording needle that indented the soft metal in response to the acoustic air vibrations. When the recording system was used in reverse the music could be heard from the indentions in the metal moving the same needle. These first machines were powered by turning a handle that rotated the record for recording and playback.

Edison specified several different motions in the recording needle to hold the sound. One motion was vertical, and the other was horizontal with the vertical motion known as *hill and dale* and the horizontal as *lateral cut*.

Edison did not continue the development of this early phonograph any further at this time, as his full-time attention was directed to the development of the electric lighting system. Other inventors picked up the idea of recording sound from Edison's early tin-foil phonograph and they started down two separate development paths; namely the cylinder record and the flat record during the 1880's. One of the early pioneers in the vertical sound method came from a laboratory set up by the new Bell Telephone Company in Washington DC. This lab took the tin-foil design and made many improvements to the early Edison work with the goal of working with Edison in a future with a joint business venture.

Another early pioneer was Emile Berliner who developed a flat rubber type material for the record that in turn used a horizontal motion for holding the sound. Berliner also developed a production method for making his records from a master recording which was a catalyst for developing a new Music Business for records due to this new ability to make a master recording and then duplicate the recording in the form of flat records.

After the electric lighting industry had been created by Edison, he returned to the development of the phonograph after learning about the work done by the Bell Telephone

Lab in Washington DC. As was his style Edison rejected working with the Telephone Lab and proceeded to develop a new and improved phonograph using a Cylinder Record made of soap and wax. The National Phonograph Company was established by Edison to sell his improved phonograph along with various franchises to sell the new records and machines around the world.

The National Phonograph Company initially marketed the technology as a business machine for speech dictation. Indeed, the early salesforce for the Washington District of Columbia area was very successful at selling these dictation machines to the Federal Government. This *Columbia* sales franchise (named because of the District of Columbia) stayed as an independent company during later efforts by Edison to regain sole control over his phonograph business in the late 1880's and would become the independent Columbia Record Company.

Some of the early salesmen for these cylinder records found that the public enjoyed hearing songs on the records more than using them for office dictation. This new use for the cylinder records along with Berliners' method for making multiple record copies was the needed spark to the creation of a brand-new business. As the Music Industry was beginning, the various company lawyers and new patents caused numerous court battles. A winner in one of the court battels took the name the *Victor Phonograph Company* to celebrate their success as a victor, in court. The winner had based his new company on the Berliner flat record patents, his own patents on a tonearm, spring motor, and his manufacturing methods for making copies of the master recording. The speed of these records was set to 78 RPM as this was a compromise by Berliner for music time versus fidelity of the music. A rejected marketing picture submitted to the Edison Company of a dog listening to a phonograph playing the dogs owner's voice was later used by the Victor Company as a Trademark for their phonograph machine. The artist of the picture painted over the original Edison Phonograph with a Victor Phonograph and then submitted the picture to the Victor Company.

The cylinder records continued to use a vertical motion for the sound waves while the flat records used a horizontal motion to capture the sound.

These early manufactures created the record business and this new music industry. The early pioneers in the recording industry would have to wait for the invention of the vacuum tube, microphone, and other devices for electronic recording and playback. Therefore, the major developments for music technology were by mechanical means for many years. The general term for this period of using mechanical methods for recording and playback of music records is referred to as the Acoustical Recording Era, since the power for the actual generation of the record groove was provided solely by the acoustic energy from the vocalists and their music instruments.

20.2 Western Electric Laboratory Introduces Electric Recording

During the 1920's, new electrical inventions were used by a team of engineers, led by Maxfield and Harrison from the Western Electric Laboratory, to develop an electric recording system for records. The team took the existing mechanical process used for recording of records and added practical upgrades using new electronic devices. This development was finished in 1925 and the Columbia and Victor Companies were the first to use this new method to record music. The other record companies adopted similar electrical methods after 1925. In addition to the electrical recording methods, Western Electric developed several important acoustic principles for the mechanical playback of music. While the recording of music was based on electrical devices in the new Western Electric process the playback was still a mechanical, although improved, process. The widespread use of electricity in homes had not yet arrived along with reasonable priced electrical playback equipment. The introduction of electrical playback equipment for consumers would occur later in the end of the 1920's and into the 1930's.

The first Victor Electrical Recorded Records were designed to be played back using the new mechanical playback phonograph called the Orthopedic type.

20.3 History of the Edison Record Company

Early in the production of the Cylinder Records the Edison Company designed a product that could be played back with little wear on the records. The type of stylus material used by Edison's phonographs to play back the music was much harder than the records so that the needle maintained its desired shape as the records were played. The first material used for a Stylus was glass which was followed by sapphire and then diamond. When the flat records were later developed, Edison used a diamond stylus and hence the name Edison Diamond Disc Records. A diamond stylus was also used for the 4-minute blue Amberola cylinders. The records that Edison made started off as a soft wax and soap type of material and then evolved to more durable materials. Edison was one of first companies to develop hard plastic materials for the record surface and these new plastic materials were used in the new Diamond Disc and 4-Minute Blue Cylinder Records. This new Diamond Disc Record was a flat record. Edison later used the Diamond Disc Record as a master recording of a song to be duplicated onto the 4-minute cylinder records.

Edison's business model was to sell a durable record with a precision, quality phonograph. Indeed the 4-minute cylinder records played today on a restored Edison player

sound very good and are often over 100 years old! It seems that the Edison record company felt that the customers would buy more records to increase the number of songs they had rather than replace the records as that wore out.

The Edison Company selected the songs to be recorded in a unique manner. For the Edison Company the main reason to issue a song was determined by how well the recording *captured* the singers and instruments music; not necessarily if the music was *what* would sell or was popular with the people at the time. For Edison it was important that the music sounded as close to the original recording as possible. Since the recording methods were Acoustical, for many years the Edison songs did not include instruments that recorded poorly. The result was that a record may have sounded good, but the sales could be poor if the public did not enjoy the song.

Thomas Edison himself would often decide if a record would be produced after the initial studio recording using his own set of standards up to the early 1920's. While his judgement may have been accurate about how well the recording was made using the acoustical mechanical means, his sense on what the public wanted was not well developed. Except for some very late records in the 1920's, Edison used a vertical motion to record the music sound. Edison's original record patent described both a vertical and horizontal motion that could be used for the sound recording; however, he believed that a vertical motion produced a better recording. The Edison records were designed to be played back at 80 RPM for the flat Diamond Disc records in contrast to the 78 RPM speed used by the other companies. The Diamond Disc records were a quarter inch thick and weighed one pound in sharp contrast to the thinner and lighter 78 RPM records.

An interesting piece of history is that Edison was very hard of hearing for most of his adult life and would listen at times to the record music by biting the side of the record player and having the sound travel to his inner ear through his jaw. The inventor of the Phonograph was hard of hearing, yet had developed an industry selling music to be heard.

Another unique feature of the Edison records that helped to reduce groove wear was that the arm that carried the reproducer was driven across the record by a separate leadscrew assembly. In this manner the groove walls of the record did not provide any force to move the arm and therefore the groove maintained its shape with repeated record playing.

The Edison records were not compatible with the other record producers which forced the consumers to purchase a specific phonograph to play back the Edison records. The result was that the sales of the Edison Diamond Disc records were not as good as the other record companies and after a peak in 1923, they continued to decline each year.

As the 1920s' were coming to an end the Edison Company introduced a new record that used a lateral method of recording and were thinner than the Diamond Disc records.

These records were designed to compete with the traditional records made by the other companies and used electrical recording. However, these records did not sell well at all.

The inventor of the Phonograph, Thomas Edison, stopped the manufacture of Records weeks before the Stock Market Crash in October of 1929.

20.4 Victor Company and Others Business Models

Most of the companies that were competitors of Edison used a different business model for their products. Their records were flat and made of a hard material that had as its primary component ground-up rock held in place using a binder of shellac. The needles that were used in the reproducer were made from soft steel and had an initial sharp point that was *not* designed to fit the record groove. The reason for the sharp point was that the needle or stylus, being softer than the record material, would quickly wear down to the correct shape of the *specific* groove for that record. This initial wearing in time would occur within a couple of revolutions of the record. Since the needle was now shaped to the specific groove for that record the rest of the song on the record would sound good. As you would expect you should replace the needle after each new record that was played. Many steel needles were sold.

The fact that the records were hard enough to wear down steel needles meant that metal particles are often present in the record grooves from the initial wear in period. In addition, while the record was made of a hard material, groove wear would still occur with each playback as it shaped the steel stylus.

The motion that drove the tone arm containing the reproducer was driven by a force provided by the groove walls of the record. The Victor Company and other record companies derived the needed force to move the tonearm directly from the same grooves that contained the recorded sound. While the tonearm design was light-weight, this method caused additional wear to the groove walls as the records were played.

The business model for the Victor company and others was to provide a low-cost phonograph to customers by using the rotating record to provide mechanical energy to operate the phonograph. By having the records wear out and be low cost, customers would buy more copies of their favorite records to replace them.

The Victor Company was very aware of the public's music tastes and produced many records that followed the current trends in songs. The records were marketed and sold to satisfy the people's desires in contrast to Edison's approach to produce a recording that reproduced the instruments as well as possible. Therefore the 78 RPM records often had the first jazz music and would introduce many future singers and players in the field of recorded

music. The Edison records, in contrast, did not have many of the newer types of music and as many new singing artists until the early 1920's, when Edison's direct involvement with song selection ended.

In the late 1920's the owner of the Victor Record Company sold his company to the RCA Radio Company. The new RCA Victor Company would continue for many years to produce music long after Edison's Company had stopped making records.

The early Columbia Phonograph company, started by selling cylinders for office dictation, also continued for many years to produce records.

20.5 New Field of Audio Engineering Created

The pioneering work by the Western Electric Laboratory in developing the science of electric recording started a new engineering effort into audio recording and playback. This Era of Audio Engineering after 1925, benefited from new low-cost electrical devices that were rapidly developed for the Radio, Phonograph, and Motion Picture businesses. Reading the early audio technical journals reveals the work that was started towards Stereo Recording, Vinyl Low Noise Records, FM radio, and True High-Fidelity performance. Much of the audio engineering work would be delayed till after World War Two ended. When the War was over the new electrical technology developed during the War, coupled with the high demand for consumer goods, led to an explosion in the quest for High Fidelity. For more information the technical Audio magazines, and the numerous College Textbooks from the 1940's and 50's should be consulted. Many well-known audio companies were started to satisfy the demand for high fidelity audio such as Heathkit, Scott, Marantz, Fisher, McIntosh, and many more.

The new Tape-Recording technology developed during and after World War Two was now used to produce the Master Recordings in the studio; along with continuing refinement of phonograph records led to the Stereo Long Playing Records we have today. The next step to the Digital Storage of the music was not so much as an improvement in the Recording of the Music as it was an improvement in the manufacture and distribution of music.

20.6 Can Edison Records be played on Victor Phonographs?

Both the Edison and other Music companies felt that their records used the correct technology for recording sound. However, their records were not compatible with each other's record players. Unfortunately, an Edison Diamond Disc record will be seriously damaged when a steel needle is used to play the record as would happen when using the Victor phonographs to play an Edison Flat record.

The used record stores will often have Edison Diamond Disc records which appear to be O.K. but when played back have a scratchy sound for the first moments of the song. These few seconds was the amount of time for a person to realize that a steel needle was the wrong stylus to be used for playback!

With the current low mass turntable arms and diamond stylus used in today's turntables the Edison Flat records are not damaged when now played back, but they will sound rather thin and low in volume before the correct vertical stylus motion is selected from the initial recording.

21.0 Recording Details of Music

21.1 Physics of Sound

Some principles about acoustic waves will be used in this section to understand the electrical recording process. Many texts and resources are available, if needed, to understand these concepts to a deeper level. The first step is the General Acoustic Wave Equation that describes the motion of the displacement particles from a vibrating source of sound.

The equation for the displacement or amplitude of the particles in an acoustic wave at any instant and point in space for a wave traveling in the positive x direction is given by the equation:

$$\xi = \left(\frac{Vm}{\omega}\right) \sin \omega \left(t - \frac{x}{c}\right) \quad eq (21 - 1)$$

Where ξ = the displacement of a particle from its normal position

Vm = the maximum particle velocity (a constant value)

$\omega = 2\pi f$, with f being the vibration frequency

c = velocity of sound

t = time

x = distance from source

The sine function is for a pure tone.

It is assumed that the maximum particle velocity will be constant and independent of frequency. The velocity of sound in air is about 1,130 ft. per sec. for room temperature and at sea level. If the equation is used to describe a particle at the source itself, then x/c is equal to zero.

The relationship between frequency and displacement of such a vibrating source, for the same applied force and same power, is an inverse one (Vm/ω). What this means is that as the frequency of a vibrating source goes up the amount of displacement required to generate the same amount of sound power decreases proportionately and vice versa the lower the frequency the greater the displacement for the same power. This inverse relationship is *central* to understanding the need for amplitude modification to the recorded frequencies for records. The velocity term will now be used to describe the change in the displacement waveform versus time. This velocity term does not refer to the speed that the sound travels through the air (c).

21.2 Constant-Velocity and Constant-amplitude

The velocity of a sine wave has a maximum value as the waveform is crossing the zero axes. The average velocity is defined as the displacement per time over a cycle of the waveform. For a constant sound power of force, the maximum displacement will decrease for an increase in frequency as an inverse relationship and the velocity of the waveform will remain constant for the frequency change. The phrase *constant-velocity* for constant signal power is shown using figure 21-1:



Figure 21-1

The first waveform shows a 50 Hz sinewave with a displacement of P-P (Peak to Peak) 2.0 inch; next are two waveforms at 100 Hz with displacements of P-P 2.0 and P-P 1.0 inch.

As the frequency of the 50 Hz sinewave is increased; we want to keep the power of the waveform constant. There are two options that can show a change in frequency as either the

P-P 2.0, 100 Hz waveform or P-P 1.0, 100 Hz waveform. The P-P 2.0, 100 Hz waveform keeps the amplitude displacement the same but now the velocity of the signal, as it crosses the zero axis, is twice as steep (slope is 2X), while the P-P 1.0, 100 Hz waveform has the amplitude reduced by a factor of two and the velocity as it crosses the zero axis is the same as the original 50 Hz waveform. The velocity must stay the same for constant power therefore, the change from the P-P 2.0, 50 Hz waveform to a P-P 1.0, 100 Hz waveform demonstrates how the amplitude must change for a constant-velocity factor with the same signal power.

These terms can be applied to the tip of the recording or reproducing stylus as it traces the record groove. In *constant-amplitude* recording, the stylus tip, at a constant signal power, moves a fixed distance each side of its center or rest position for any frequency. The amplitude of the swing of the stylus tip is constant for all the frequencies in this region for the same amount of sound intensity. For example, if the sound intensity drops, then the amplitude of the swing of the stylus will drop. However, if the sound intensity stays the same as the frequency changes, then the amplitude swing value of the stylus will stay the same.

To demonstrate *constant-amplitude* refer to figure 21-1 and start with the signal at 100 Hz and amplitude P-P 2.0. If the swing of the stylus tip stays the same as we decrease the frequency to 50 Hz and amplitude 2.0 P-P, then that is an example of *constant-amplitude*. What is important to understand is that the *power* of the signal at 50 Hz is now less than the power at 100 Hz to keep the swing or displacement the same. This is because the velocity of the 50 Hz waveform is now half of the 100 Hz waveform and the velocity is related to the power. To implement *constant-amplitude* recording will require a modification to the signal as it is recorded that will be dependent on its frequency.

For *constant-velocity* recording, the maximum velocity of the stylus tip at a given signal level remains constant for any frequency as was shown, for constant power. The use of *constant-velocity* will not require a modification to the signal as it is recorded dependent on its frequency.

Constant-Velocity recording occurs for most of the frequency range of Acoustic Records.

21.3 Constant-Acceleration (*Pre-Emphasis*)

Constant-acceleration is another term that helps to describe a type of modification to the frequency response of the music, useful for surface noise reduction. Today this term is called *Pre-Emphasis* instead of constant-acceleration. Refer to figure 21-1 and examine the difference between the 50 Hz P-P 2.0 waveform and the 100 Hz P-P 2.0 waveform. The

velocity is changing, and the energy is increasing from the 50 Hz to the 100 Hz waveform. If we cause this to happen above a certain frequency by increasing the energy in the signal at an increasing constant rate, then since the velocity is also steadily increasing, we can call the result *Constant-Acceleration*.

The reason that this method can help the over-all recording of music is related to the surface noise on the record. The surface noise on the record is mostly random noise with uniform noise frequency. As music frequencies increase, the amplitude of these frequencies drops to a point where the random noise and the music are similar in amplitude. Now the music frequencies are buried in the surface noise. By steadily increasing the amplitude of the recorded frequencies, above a certain value during the production of the master recording the music rises above the surface noise. When the record is played back this increase in amplitude will be reduced to the correct level, but during this reduction the surface noise will *also* be reduced by a corresponding amount. The net result is that the range of the upper frequencies that can be heard from the music has been extended upward by reducing the effect of the surface noise that was limiting the hearing of these music sounds.

The implementation of this method specifies the time constant of the circuits rather than a specific frequency for this change in music power to occur. However, the reverse specification can be either in time constant values or in frequency. The term roll-off refers to a frequency where the correction curve starts.

This *Constant-Acceleration* concept to extend the upper range of recorded music was discussed in the original work by the Western Electric Labs in 1925, but was not used on a regular basis until many years later at which time the term Pre-Emphasis replaced the original term. The term Pre-Emphasis was used for records since prior to the wide use of this concept for records the FM radio transmitters and receivers had developed a similar method by boosting the high frequencies during transmission followed by reduction at the receiver. This Radio concept was referred to as Pre-Emphasis. Hence the record companies used the same term as it was familiar with the public.

21.4 Early Recording of the Sound Grooves

The early sound recording engineers did not have many tools or reference information available to them as they created the first commercial records. Many trial and error experiments occurred using mechanical equipment for recording and playback.

All Acoustic Recording used a Horn to capture the music sound waves which would then move the recording groove cutting device. The horns mechanical dimensions were

selected to match, as well as possible, the acoustic impedance between the recording studio and the cutting device.

The Acoustic Era recording engineers provided damping in the recording mechanics to produce a constant-velocity recording for the music power from about 400 Hz to 3000 Hz. Below 400 Hz the motion was constant amplitude. The early records using the acoustic process had a turnover around 400 Hz due to the technology, not the desire. This constant-velocity recording occurred for both the Vertical and Lateral motion for the record grooves.

21.5 Playback of Acoustic Era Records

The magnetic cartridge that is used to convert the groove motions to sound produces a signal due to the rate of change in magnetic flux inside the cartridge. These signal levels are directly proportional to the velocity of the stylus (flux changes) as it follows the groove motion. Thus, the magnetic cartridge is a *constant-velocity* device. If the input to the magnetic cartridge has a constant amplitude as the frequency is increased, then the output amplitude versus frequency would rise at a rate of 6 dB per octave (doubling of frequency). However, the recorded music sound on a record, using constant-velocity recording, has a decreasing amplitude with frequency and the use of a magnetic cartridge with a constant-velocity recording will provide a resulting flat response over frequency. This is not true for the crystal type of pickups that were occasionally used as they are sensitive to the displacement not the rate of change of displacement.

21.6 Amplitude Modification with Frequency for Electric Recording

Electric recording with microphones and amplifiers allows new low and high music frequencies to be recorded with normal amplitudes for the music. The previous section provides the background that there will be problems in *directly recording music* into the record grooves without some type of amplitude modifications to these *new* low frequencies.

For constant acoustic power the amplitude or the swing of the groove cutter increases as the frequency is decreased and a point is reached, as the music frequencies decrease, where there is a chance that one groove would cut into the adjacent. The space between the grooves cannot be changed without decreasing the playing time of the record and since the record companies had an existing product being sold, acoustic recorded records, with a certain playing time something had to be done to make the new electric process work with the existing record groove spacing. The previous acoustic recording method did not record low

frequencies very well so that the increasing amplitude motion with lower frequencies had not been a problem with the cutter touching the adjacent groove during the Acoustical Recording Era.

The solution that was developed by the Western Electric Laboratory, Maxfield and Harrison Team, was to modify the bass frequency content of the recording signal going to the cutting device. Progressive bass attenuation was introduced below a value called the turnover frequency, resulting in the amplitude of the cutter's swing for all signals below this frequency remaining the same for the *same power* of the incoming signal. This relationship is called *constant-amplitude*. In practice this technique is implemented with modifications made to the equipment's frequency response using a 6dB per octave attenuation rate (or 20 dB per decade) starting at the turnover frequency. As the frequencies decrease from the turnover value the amplitude of the signal to the cutter is reduced by the same amount that the amplitude would have been rising for constant input power. By using this modification to the lower frequencies, the same record music time was maintained for the new records as for the previously manufactured records using the previous acoustic recording.

For frequencies above the turnover value the grooves are cut with a *constant-velocity* characteristic which means that no intentional changes are made to the music since this is the natural or normal way that the cutter works with sound waves. The other method *constant-acceleration*, while discussed in the original research paper, was not implemented at this time (1925). You can divide the frequency range of both the recording and playback process into sections that have either *constant-amplitude* or *constant-velocity* characteristics.

When the RIAA curve became standard for records the frequency range of both the recording and playback used three distinct areas. The lowest range was for *constant-amplitude*, the middle for *constant-velocity* and the upper for *Pre-Emphasis* or *constant-acceleration*.

21.7 Comments on Filter Response

When filters are described, a typical description will state the amount that the filter changes per frequency interval (slope). For the typical electrical filters used in audio work a value of 6 dB per octave (factor of 2) or 20 dB per decade (factor of 10) is often stated. These filters include a second value to state where the change in amplitude starts or where the slope changes from 0 to ± 6 dB per octave. The value where the slope changes will be given as a frequency but this change in slope does not happen *exactly* at this value. In an actual filter the slope of the filter's amplitude versus frequency change starts before the quoted frequency

value then changes ± 3 dB at the quoted frequency; and continues to change slope until it equals the 6 dB value. The change in amplitude is gradual. In the graphs that show the filter's response, you can see this gradual change occurring. The frequency of the filter is quoted at the 3 dB change point for the common 6dB slope filters.

21.8 *Early Electric Recording & Playback Conditions*

The work by Western Electric had several goals for accurate and faithful performance across the range of audio hearing. However, as early pioneering work had to deal with the conditions present in the 1920's, the actual performance fell short of their goals. Since the technology prior to their work was a total mechanical system driven by sound power their first product was much better than anything before. The fact that a sensitive microphone was used to record the sound meant that the musicians could perform in a studio under the normal conditions used by musicians for the location of instruments, rather than in an artificial setup for the acoustical recording in which performers shouted directly into the recording horn and pianos were placed up on boxes to balance the recording sound. This new technology allowed drums and bass instruments that had previously been difficult to record to now be used in song arrangements. These new electric recorded songs caused a major revolution in the music world.

At the time of the development of this new Western Electric Recording System the average record buyer did not have access to consistent, low cost, electrical power in their home. In addition, any electrical equipment that could be used for playback of the new records had to be invented and produced. While the record manufactures could afford to spend money for the latest technology for recording the music, the consumers of the records could not. Therefore, the Western Electric company introduced an improved *mechanical* playback of the records using no electronics. This new mechanical system reproduced a larger range of the recorded frequencies now present on the new electric records. They improved the recording of the records but did not change the playback from a mechanical system for the early electric recorded records. When you find records from this early electric era, the record label will often have the words electric recording written with lightning bolts or other visual symbols of this new technology. The very early Victor records will often have the words Orthophonic Recording and VE on the label which means that you would use a new mechanical phonograph to play them back.

The initial recording system developed by Western Electric had a response from 50-5000 Hz with a natural turnover at 200 Hz due to the mechanical performance of the cutting

system for the master. The change from *constant-velocity* to *constant amplitude* around 200 Hz was due to the best that the technology could do at that time (1925). The potential modification to the high frequencies for surface noise reduction was not implemented since the upper frequency limit possible was still below record surface noise. The first electrical recordings could not take advantage of all the design concepts due to the limited manufacturing technology available at the time.

The mechanical devices that cut the grooves in the master record were carefully designed such that the resonances in their structure did not occur within the audio frequencies by having the Q or Quality Factor of the mechanical cutter set to a value around 0.1, which is an under-damped system. What this does is to make the velocity of the cutter (the sharp edge that makes the grooves) the same or constant for the same applied power over the widest audio frequency range possible. You do not want any mechanical resonances to amplify or change the sound of the music. The original work by the Western Electric Team described how they were able to achieve this result using a mechanical damping *rubber line* and other methods.

21.9 Later Recordings

The initial work by the Western Electric company changed the record industry in a major way. The continuing rapid development of electrical devices expanded the audio range that was recorded and the record companies were able to introduce new styles of music since the microphone allowed a wide range of instruments to be used in songs. For singers the microphone allowed a crooning style that became very popular. However, a significant opportunity was lost at this early electric recording moment to develop a set of standard settings for the turnover value and potential pre-emphasis values across the record industry. Due to the human nature of the recording engineers, each record company felt that *their settings* for the recording of the music for turnover values and use of pre-emphasis were the correct values to use. The specific values were at times viewed as trade secrets and not widely described.

The recording engineers at the record companies used unique settings to make the record sound good for the customers by understanding that few phonographs at this time (20's & 30's) were high performing audio devices. Some records had extra high frequency and extra low frequency boost added so that the music would sound good on a typical customer's phonograph *when played back*. The same record that would have sounded good on a 1938

phonograph may not sound correct on a current high performance sound system that has a flat response from 20 to 20,000 Hz.

Later when the new 45 RPM record and LP record were developed, the various record companies realized that a standard playback EQ (Equalization) response was needed; which established a uniform standard (RIAA) for the turnover and pre-emphasis values. During the time 78 RPM records were made no industry wide set of recording standards existed. There even were differences between the European and American manufactures.

21.10 *Edison Electrical Recording*

Even though the Edison Company had pioneered the recording of sound they did not adapt to the new electrical methods until after the initial work by Western Electric. The various engineers within the Edison Company tried to bring in this new technology but the founder of the industry, Thomas Edison, did not want to change the business model from continuous improvement using the Acoustic Recording Method. The records made in 20's by the Edison company are very good recordings, although limited by the Acoustic Recording process in the types of instruments that were used. By 1927 the Edison Company developed their own methods for electrical recording with equipment from the General Electric Company (GE was established to manufacture equipment from Edison Patents). Much of the specific information about Edison's Electric Recording work was not published and detailed information is not available. What is known is that the Diamond Disc Records had a much fuller sound around mid-1927.

Research by the Diamond Cut Productions Company found that a turnover value of 500 Hz had been incorporated into the Edison Phonograph and Radio that was made for a brief time in the late 1920's for playback of the new Needle Type Edison lateral records. At this point the Edison Company was attempting to refresh their product lines to the current state of the art, but they were too late and in October of 1929 (just before the stock market crash) Thomas Alva Edison, the inventor of the Phonograph, shut down the manufacture of all records.

To playback records correctly, knowledge of the recorded *electrical properties* will be needed so that the modifications to the recorded sound can be removed. If the exact opposite to the modifications made during the recording of the master record are used, then the original desired sound will be heard.

22.0 Selection of Cartridge, Stylus Size, & Tracking Force for Transcriptions

22.1 Introduction

The selection of the cartridge, the stylus, and the tracking force for transcribing the sound from the record groove and then converting it to an electrical signal is *critical* to the success of music restoration for several reasons:

1. The transcription process cannot damage the sound grooves since the record is a piece of history.
2. The process must extract the maximum signal and the minimum noise from the record groove. Noise is defined as both surface noise and possible distortion to the original sound from the *shape of the stylus*.
3. The conversion of sound to mechanical motion used different methods by the music companies which in turn requires unique approaches for transcription.

22.2 Cartridge Selection Acoustic and Electric Recording

The Acoustic recording process for lateral and vertical groove motion was a *constant-velocity* type over most of the recorded music frequencies. The low frequency end of the music had reduced energy due to the use of the original mechanical recording equipment which provided a natural *turnover* to *constant amplitude* in the general range of 400 Hz.

The Electric recording method used a combination of constant-velocity and constant-amplitude with the turnover frequency a fixed value versus the acoustic technology.

For constant-velocity recording, the use of a magnetic cartridge will provide the best method for playback of the groove for either vertical or lateral motion. The magnetic cartridge has a constant-velocity response and when used with the same recording method yields the correct *flat frequency response*. Acoustic recording and Electrical recording can use the same magnetic cartridge for playback from the record.

Cartridges have been made from crystal and ceramics, however, the output from these materials depend on the displacement of the material, rather than the rate of change of magnetic flux. If these cartridges are used, additional EQ circuits will be needed.

There are many magnetic cartridge manufacturers available, and the choice of cartridge should also provide a means for the user to change the stylus.

22.3 *Effect of the Stylus Dimensions for Transcriptions*

How does the shape and size of the stylus affect the playback of the recording? One approach for understanding is to consider that the recorded sound wave in the record groove has a finite *wavelength*, defined as the distance from the beginning of the sound wave to the end of a cycle. While this definition is strictly correct for sine waves not sound waves; it can be used to estimate the relationship between the stylus dimensions and the maximum recorded frequency in the following manner.

As the radius of the record changes, the velocity of the stylus tip following the sound groove will also change for a constant turntable speed. For the worst-case situation, consider the velocity at the inner radius of the record and the highest frequency recorded. If the length of the sound wave in the record groove is the same or close to the size of the stylus in contact with the groove, then severe distortion will occur since the shape of the wavelength cannot be accurately traced by the stylus. Although this concept is a starting point, an approach that measures the actual distortion is needed since the sound is degraded before it is a complete loss.

The general term *Tracing Distortion*, is used by audio engineers to describe the music distortion caused by the stylus as it follows, or traces the sound groove is. This playback problem is fundamentally caused by cutting the original music groove with a chisel shape (wide for the groove width and then narrow on the edges) when a spherical shape is used to playback or to trace the music. This general tracing problem was studied in a landmark technical paper in the Journal of the Acoustical Society of America written by J. A. Pierce and F.V. Hunt in the late 1930's. In this technical paper the exact motion of the stylus was calculated as it followed the groove so that a clear understanding of the distortion caused by the stylus shape and recording process could be mathematically described. After this article was written, several researchers used the results to understand the limits for vertical and horizontal recording. The next section analyzes the tracing distortion calculations by Frayne and Wolfe that built on the original work by Pierce and Hunt.

22.3.1 *Frayne and Wolf Analysis*

This information is based on the book "Sound Recording" by Frayne and Wolfe contained in section 13-7 in their book. This reference should be consulted for greater details, if wanted, to learn more about tracing error. Figure 22-1 shows a spherical stylus as it travels

along a hill and dale record groove from their book. Figure 22-1 can also be used to describe lateral motion, if the figure is turned 90 degrees.

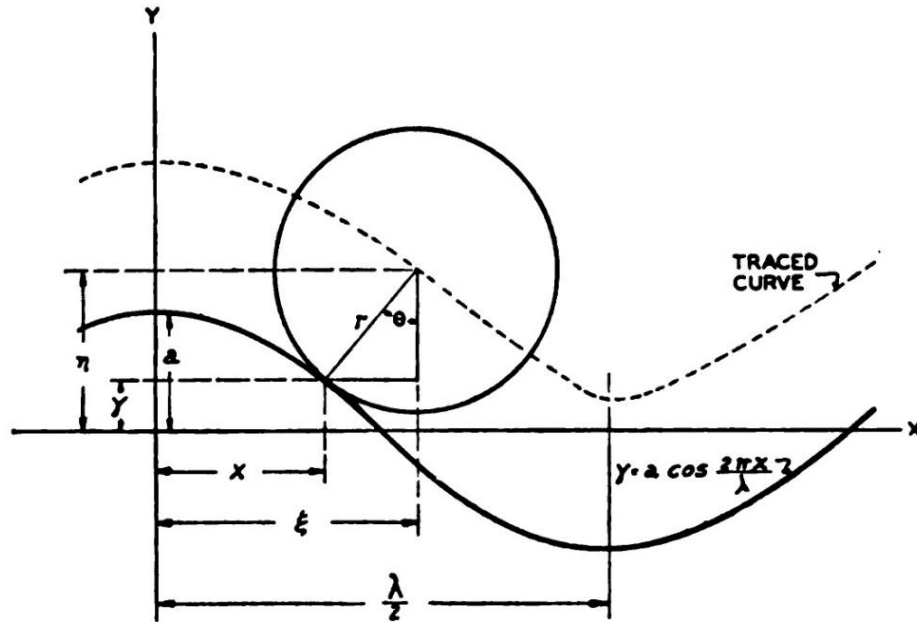


FIG. 13-14. Recorded curve and curve traced in reproduction. (After Pierce and Hunt.)

Figure 22-1 Stylus as it Travels the Sound Groove

The dashed curve shows the shape seen by the stylus and sent to the cartridge and the solid curve shows the original sound curve cut into the material to form a groove. Notice that the original cosine wave shape traced is distorted because the contact location changes from the bottom of the stylus ($x = 0$) to a side (x) and then back to the bottom. The traced curve has been distorted from the original and this general term is called *tracing distortion or error*. The reference book continues to define the various variables and develops a method to describe the traced curve as a series of complex equations. The authors of the book cite the original work on this subject developed by Pierce and Hunt in the 1930's. For the figure shown in 22-1, the variables as defined in the book, section 13-7, are:

$r = \text{radius of stylus}$

$\lambda = \text{wavelength of recorded sound}$

$\theta = \text{angle of contact with reference to perpendicular of stylus}$

η and ξ are the stylus coordinates

x and y are the recorded cosine coordinates

$a = \text{amplitude of cosine and } k = \text{a constant}$

The recorded sound curve is: $y = a \cos \frac{2\pi x}{\lambda} = a \cos kx$ eq (22 - 1)

$$\xi = x + r \sin \theta \text{ eq (22 - 2)}$$

$$\text{and } \eta = y + r \cos \theta = a \cos kx + r \cos \theta \text{ eq (22 - 3)}$$

$$\text{Where } -\tan \theta = -ka \sin kx \text{ eq (22 - 4)}$$

$$\xi = x + \frac{kar \sin kx}{\sqrt{1 + k^2 a^2 \sin^2 kx}} \text{ eq (22 - 5)}$$

$$\eta = a \cos kx + \frac{r}{\sqrt{1 + k^2 a^2 \sin^2 kx}} \text{ eq (22 - 6)}$$

These last two equations describe the motion of the stylus as it traces the groove, hence the potential tracing error. More mathematical development occurs in the book in solving these equations using a Fourier Series Expansion and then developing results for the harmonic distortion in the sound caused by the playback mechanisms. An important result in the referenced book states: "It is evident that the shape of the poind (the traced curve) and hence its percentage harmonic content does not depend on the *actual* dimensions of the original cosine curve and the radius of the tracing circle, but rather its shape depends on the *relative* values of certain of the dimensions. Thus, the shape of the poind can be entirely specified by giving the values of the ratios of (a/λ) and (r/λ) where a is the amplitude of the wave being traced, λ is the wavelength on the record, and r is the stylus-tip radius."

This result gives us great insight into the correct stylus to use.

The next section in the reference develops a chart that shows total harmonic distortion results for various conditions, namely lateral or vertical modulation, stylus radius, amplitude, frequency, and record speed. All these values can be reduced to a plot of ka vs kr . Where:

$$ka = (2\pi a / \lambda) \text{ and } kr = (2\pi r / \lambda) \text{ eq (22 - 7)}$$

The chart of Total Distortion using these values is shown in figure 22-2.

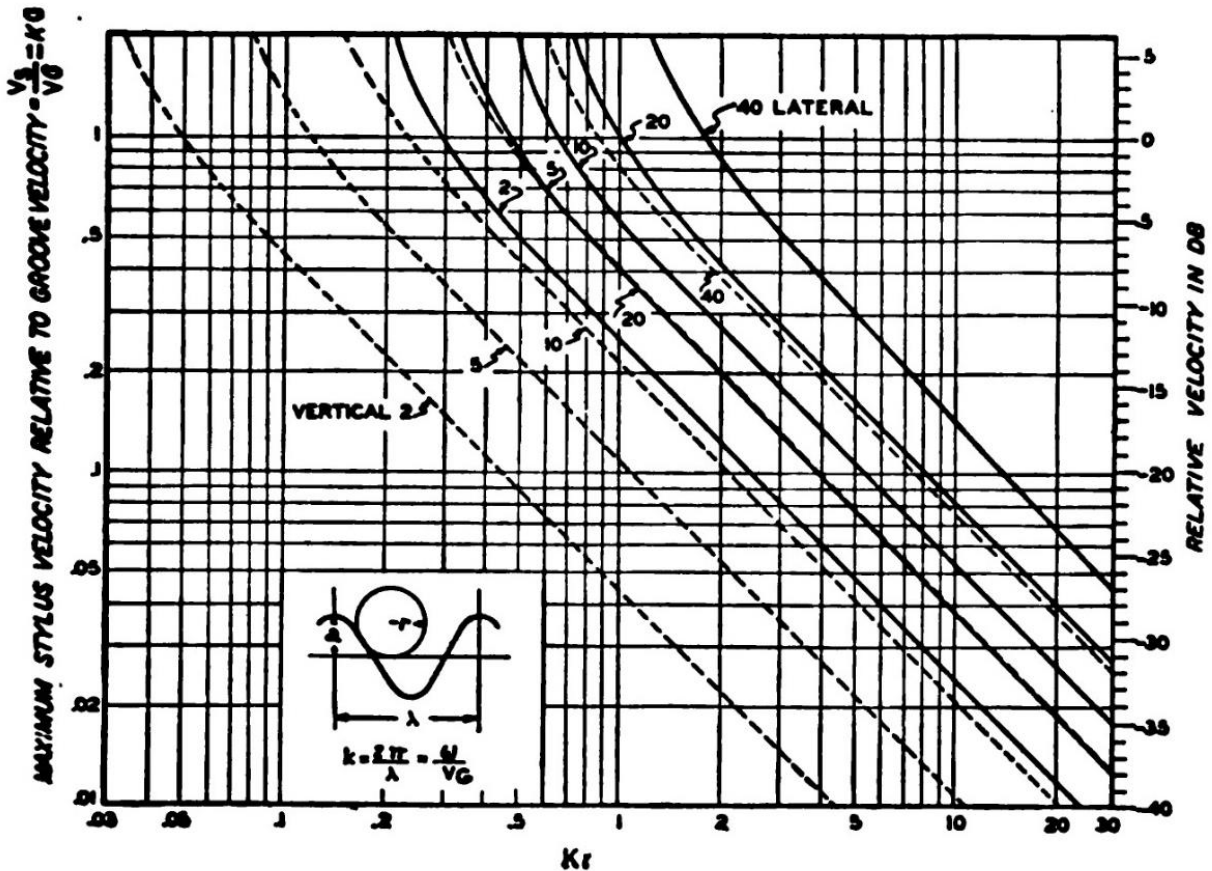


Fig. 13-16. Constant distortion contours for lateral and vertical records. (After Pierce and Hunt.)

Figure 22-2 Constant Distortion Contours

Notice on the chart that there are lines for both Vertical (Dashed) and Horizontal (Solid) motion and that the Vertical values are greater than the Horizontal for similar conditions due to the cancellation of even harmonics for horizontal motion.

The analysis in the book continues with developments of other relationships that can be established with the following conclusions.

The Total Harmonic Distortion Varies:

1. Directly with the recorded amplitude for vertical and as the square of recorded amplitude for lateral records, assuming constant frequency, groove speed, and stylus radius.
2. Directly with frequency for vertical and as the square of frequency for lateral, other qualities remaining constant.
3. Inversely as the groove speed for vertical, inversely as the square of groove speed for lateral, other qualities remaining constant.
4. Directly as the stylus radius for vertical, directly as the square of the stylus radius for lateral, other qualities remaining constant.

These relationships can be written with these equations:

$$\text{Distortion of vertical records} = \text{constant} \times \frac{raf}{vg} \text{ eq (22 - 8)}$$

$$\text{Distortion of lateral records} = \text{constant} \times \frac{r^2 a^2 f^2}{Vg^2} \text{ eq (22 - 9)}$$

Where

$$Vg \text{ (Groove Speed) } \text{inch/sec} = \frac{(2\pi R)x \text{ RPM}}{60} \text{ inch/sec} \text{ eq (22 - 10)}$$

The chapter concludes with a graph, figure 22-3, showing Total Distortion versus Recorded frequency for a spherical stylus with a radius of 2 mils and 2 mils of constant amplitude up to 300 Hz and constant-velocity above 300 Hz. These values represent some lateral record conditions at the time of writing the book.

Notice on the graph the equation that calculates the amplitude change (decreases) as the frequency rises from 300 Hz (turnover value).

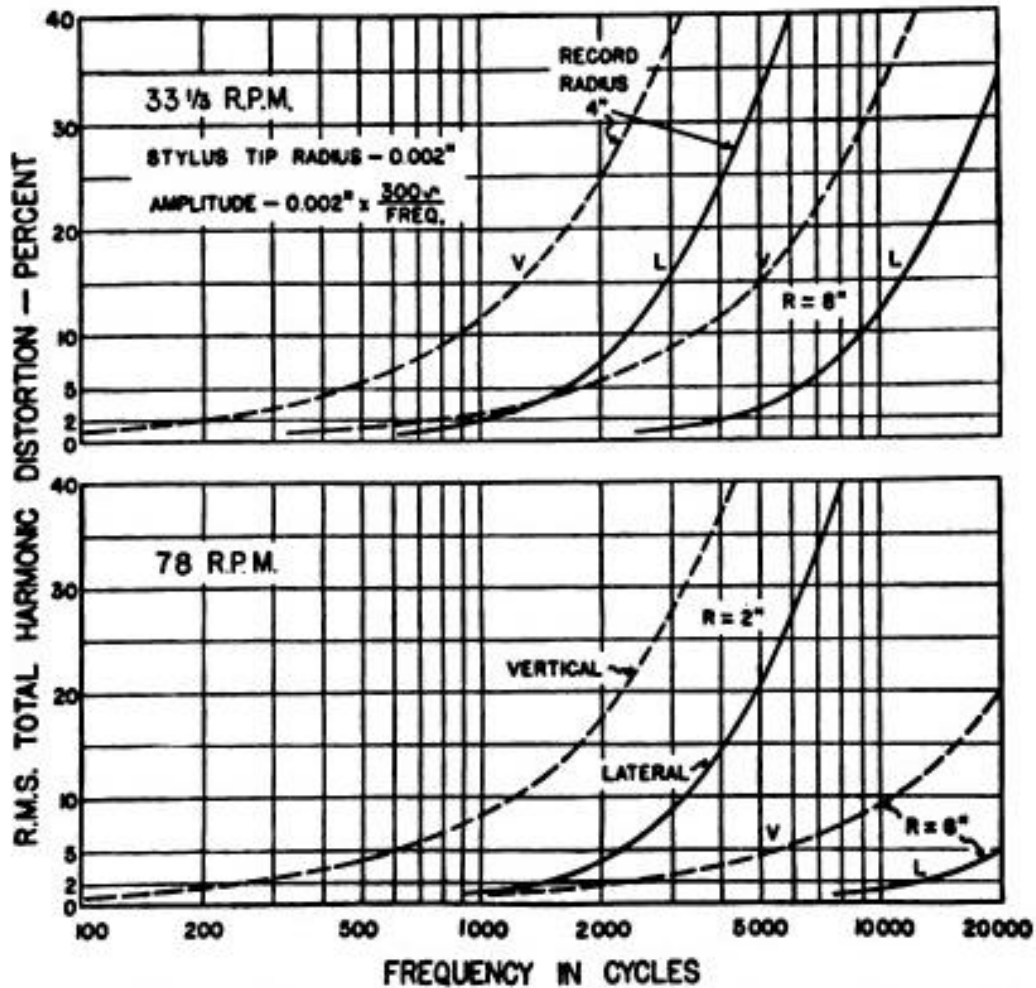


FIG. 13-17. Curves of distortion versus frequency for various recording conditions. (After Pierce and Hunt.)

Figure 22-3 Distortion for Various Record Radius

Using this information, calculations for harmonic distortion for Edison Diamond Disc Records and 78 RPM records can be performed to understand the effect of stylus size on performance. Two cases for vertical diamond disc records will be studied using the original stylus shape by Edison and a current stylus used by DJs for LP records. The DJ stylus was selected from experimental studies that found that a product from the Ortofon company performed very well with the diamond disc records and exceeded the original Edison design for clean sounding music. The lateral records will use a 2.5 mil stylus and a 3.5 mil stylus where the 2.5 mil was originally recommended by cartridge companies for *typical* 78 RPM records and the 3.5 mil value came from experimental studies. First the original diamond disc stylus design.

22.3.2 Edison Diamond Disc Stylus Selection

Diamond Discs records used a hill and dale method to store the sound waves in the record groove. This method had been used by the Edison Company for the cylinder records and continued with some changes for the new flat records. Figures 22-4 and 22-5 are from the Edison National Park and shows the diamond stylus and the stylus in a record groove.

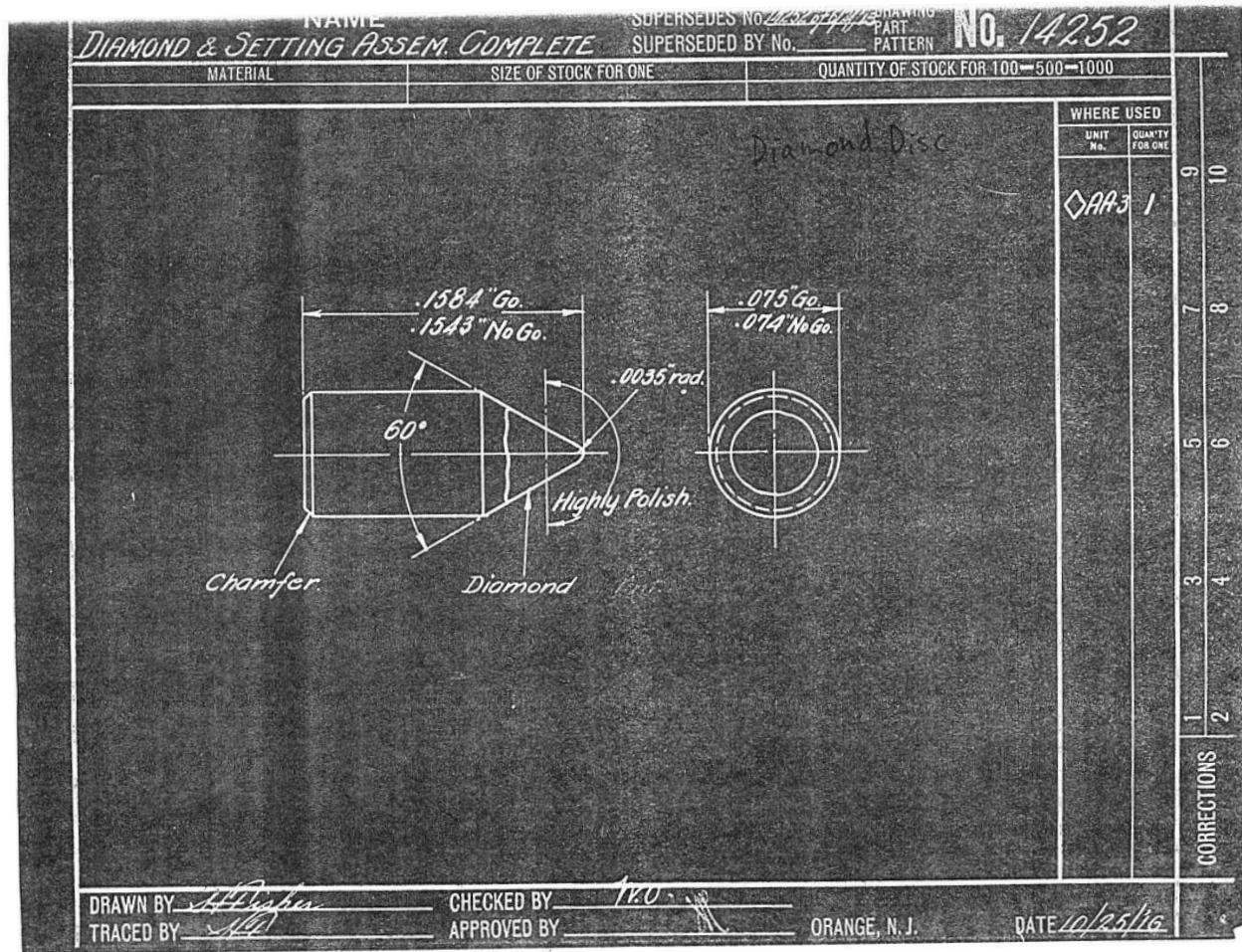


Figure 22-4 Diamond Disc Stylus

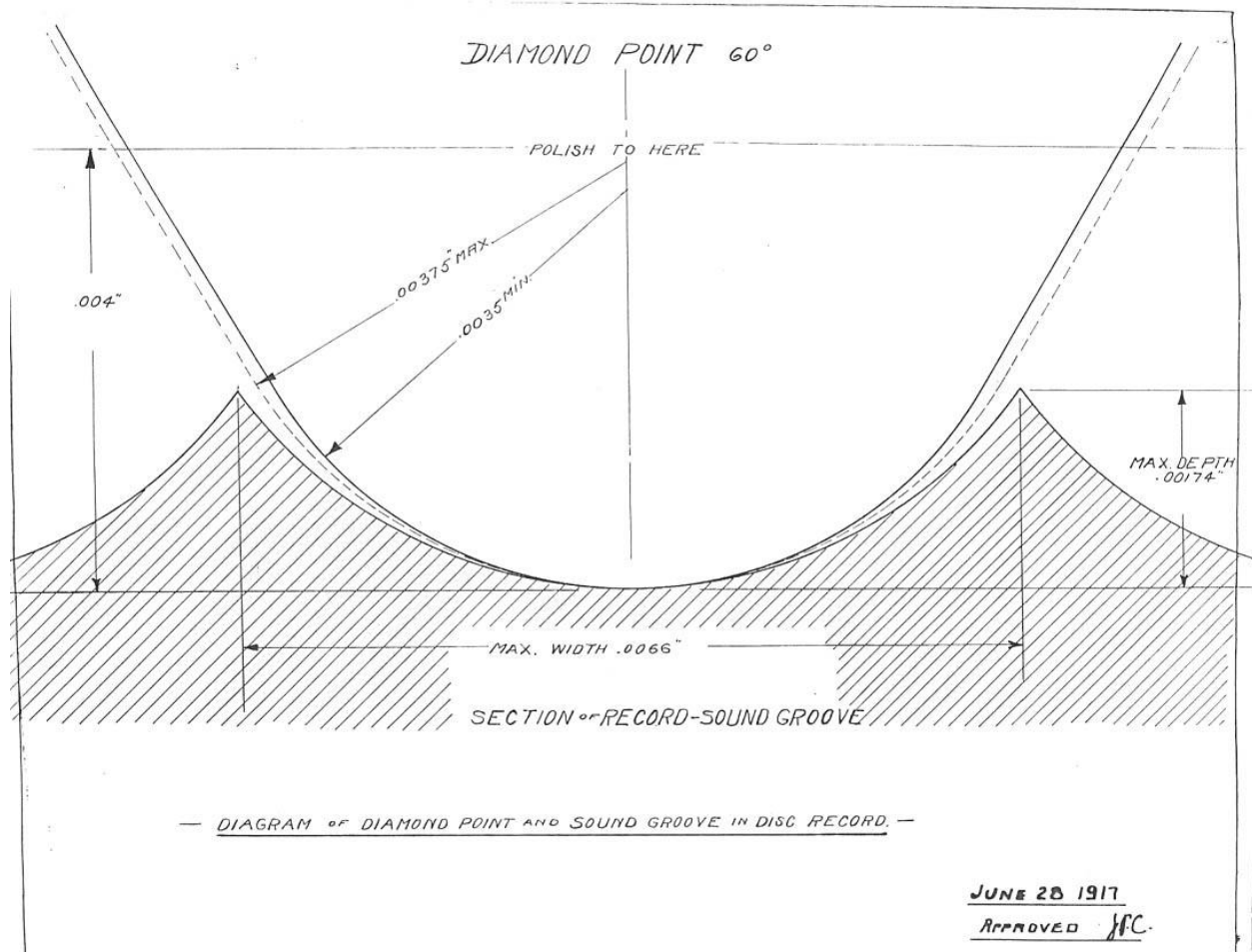


Figure 22-5 Diamond Disc Stylus in Sound Groove

These two drawings provide the needed information to determine the correct stylus size to use for the Diamond Disc Records. Since the modulation of the sound is vertical and the maximum depth is 1.74 mils (thousandth of an inch), the midpoint would be half of this value or 0.87 mil and would correspond to no sound. The peak movement would be ± 0.87 mil which is a very small amount.

The radius of the diamond point as shown fits within the groove and is designed to ride on the *bottom* of the groove. The music information is contained at the bottom of the groove and the width of the groove will vary as the recording cutter travels up and down. The original groove on the record was cut into a soft master recording using a shape described in Edison Patent number 964,221. This patent explains a new method of recording the sound that deviated from the previous cylinder recordings by using a smaller groove size and smaller depth to improve the performance of the cutter so that downward and upward cutting motions

would need similar cutting energy. The previous method introduced some distortion by having more sound energy required to cut down than up.

Figure 22-6 contains drawings from the patent that shows the reduction in the groove size using the new method as shown in patent fig. 4, the diamond disc groove and patent fig. 3, the previous groove on the cylinders. Patent figure 5 is the new method near the end of the record.

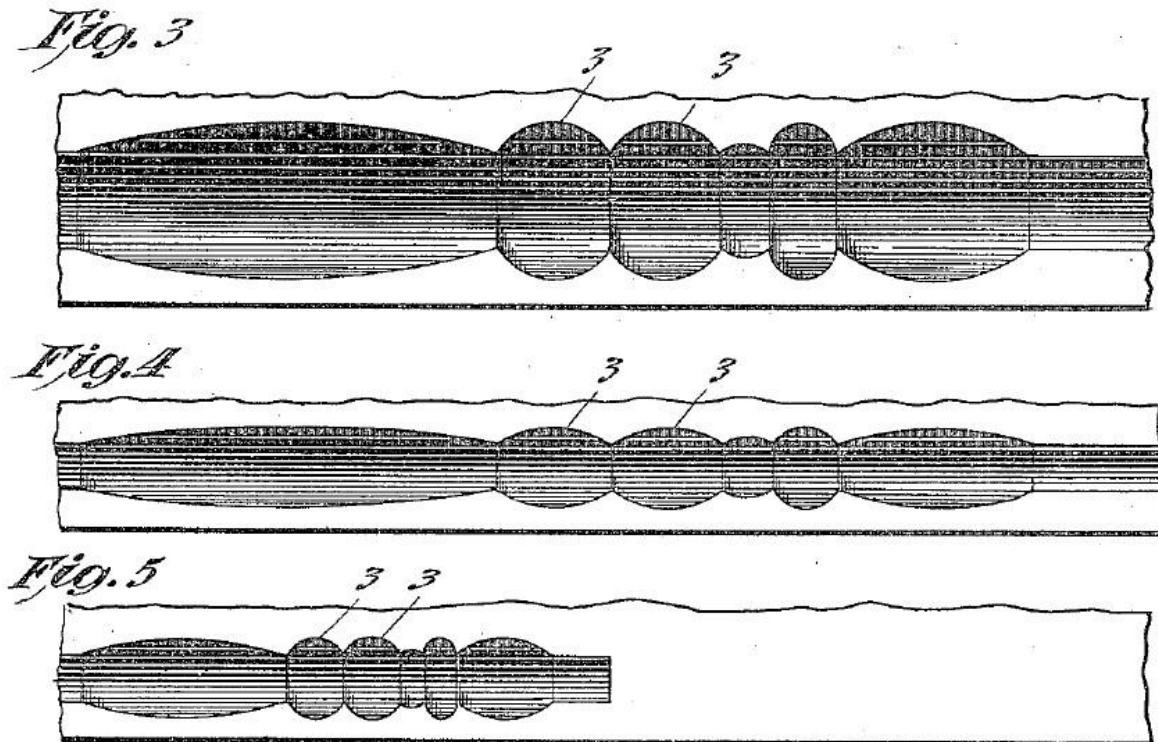


Figure 22-6 Patent Information

The Edison Music Company, as did the other Music companies, did not publish technical information regarding the details of their record production. Changes occurred due to constant improvement so that the drawings and Patent information can only be used as a guide.

Since the Diamond Disc Phonograph used no electrical amplification until much later at the end of the 1920's the motion of the sound grooves had to be amplified via mechanical methods. Within the Edison reproducer was a lever that amplified the groove motion coupled to a heavy floating weight so that when moved by the stylus the motion would in turn move a special diaphragm to produce sound waves. The force applied to the stylus was from a mass of approximately 190 Grams which is equivalent to a Force of 0.42 lb.; and could be more for

different models. This heavy force moved by the sound groove would then produce a significant amount of sound energy for the listener. Because of the large stylus force, Edison used a rather large diamond point in a spherical shape to provide a reasonable lifetime for the records, as the pressure on the record groove is related to the applied force and the size of the contact area. These records had a thick core, about one quarter inch thick, with a hard plastic material on the surface that held the sound groove. When a transcription is made today with electrical amplification the same stylus force and shape originally used does not have to be used. What are the limitations in the reproduction of the music from the shape and size of the stylus?

First, how was the music groove cut in the recording blank? The recording at the music studio was made with a soft material and a cutting tool that was driven by the acoustic energy from the musicians. After the original recording was made, the master stamping molds for the actual records were made from this original recording. When the original recording was made a Patent number 1,024,839 from Edison provides some information about the groove shape. Figure 22-7 shows some drawings from this patent.

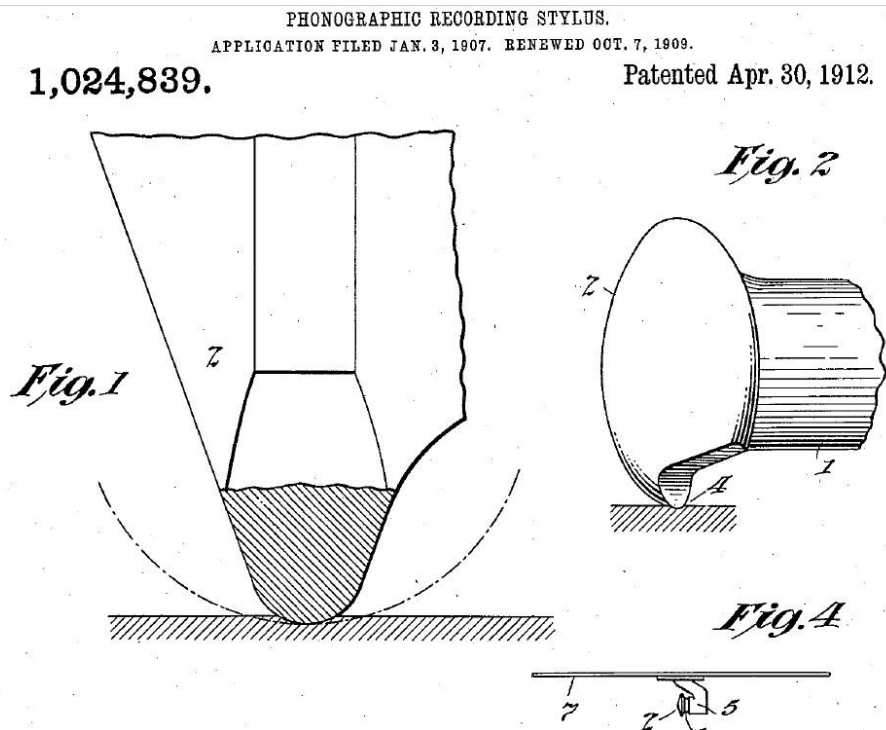


Figure 22-7 Edison Recording Cutter

In the patent description, Edison refers to the shape in the Patent Fig. 2 as that of the head of a pin with a cut made into the surface. From the patent you can see that the groove

shape is like a chisel with the wide portion cutting up and down with the song as the disk rotates. The question is *what is the result of using a certain stylus shape as it rides in this record groove cut with the recording cutter?* The results of the Pierce and Hunt Analysis Method can be used to examine the performance of the original diamond disc stylus and a stylus found from experimental results to provide a *clean sound*, namely the Reeloc DJ 0.7 OM Black from Ortofon. This stylus from Ortofon is a current DJ style that has a 0.7 mil radius on a spherical polished shape.

Worst Case Calculations for Two Different Styli

The *worst-case playback* conditions for the Diamond Disc during a playback are found at the inside record grooves (close to the center) and for high frequencies of short wavelength. For these calculations, two high frequencies of 4 kHz and 8 kHz and two stylus radius sizes will be used; the original Diamond Disc 3.75 mil (Max value) and a 0.7 mil DJ (Disc Jockey) Spherical Shape designed for “Scratching and Playing” LP records.

First Case: Inside Track (2.38 inches) and 8 kHz sound wave.

$$Vg \text{ (Groove Speed) } \textit{inch/sec} = \frac{(2\pi R) \times \textit{RPM}}{60} = 19.94 \textit{ inch/sec} \textit{ eq (22 - 10)}$$

$$a \text{ (peak amplitude) } = 0.87 \textit{ mil} \times \left(\frac{f^0}{f}\right) = 0.04 \textit{ mil} \textit{ eq (22 - 11)}$$

Where $f^0 = 400 \text{ Hz}$ (Where Max Amplitude Occurs) and $f = 8000 \text{ Hz}$ (note that constant-velocity recording was used so that the amplitude will be reduced as the frequency increases per eq (22-11)).

$$\lambda \text{ (Record) } = \frac{Vg}{f(\text{Hz})} = 0.00249 \textit{ inch} \textit{ eq (22 - 12)}$$

$$Ka = \frac{2\pi a}{\lambda} \textit{ and } Kr = \frac{2\pi r}{\lambda} \textit{ eq (22 - 13)}$$

For 3.75 mil (Original Diamond Disc Stylus), $Kr = 9.46$ and $Ka = 0.10$ which gives from figure 22-2 a total distortion value $\approx 50\%$.

For 0.7 mil (DJ LP Stylus), $Kr = 1.77$ and $Ka = 0.10$ which gives from figure 22-2 a total distortion value $\approx 6\%$.

Second Case: Inside Track (2.38 inches) and 4 kHz

Using the same equations as before with value changes $a = 0.00009$ and $\lambda = 0.00498$.

For 3.75 mil, $Kr = 4.73$ and $Ka = 0.11$ which gives from figure 22-2 a total distortion value $\approx 20\%$.

For 0.7 mil, $Kr = 0.88$ and $Ka = 0.11$ which gives from figure 22-2 a total distortion value $\approx 5\%$.

Other calculations can be performed on the outside track and for other stylus values, however, conclusions can be drawn for the best stylus to transcribe Edison Diamond Disc Records. The use of a current DJ style 0.7 mil spherical shape will perform better than the original Diamond Disc stylus and should be used. The 3.5 mil Stylus used by Edison was needed to support the heavy weight to produce a reasonable acoustic output using strictly mechanical means. The fact that a smaller radius for the stylus would provide a better reproduction of the sound was understood by Edison and a patent from 1900 numbered 652,457 describes the elliptical shape of stylus that is used today.

Because the stylus rides on the bottom of the groove for vertical recording, a smooth and polished tip (as shown in the Edison Drawings) is required. The DJ style seems to have a smooth tip even though this stylus is designed for the LP records where the stylus does not ride on the bottom. Figure 22-8 was provided by the Ortofon Company for their Reloop OM Black DJ Style of stylus. The stylus is a spherical shape and will ride with full contact in the sound groove of a diamond disc since this shape is also a spherical shape.

Stylus type - Spherical
 Stylus tip radius - R 18 μm

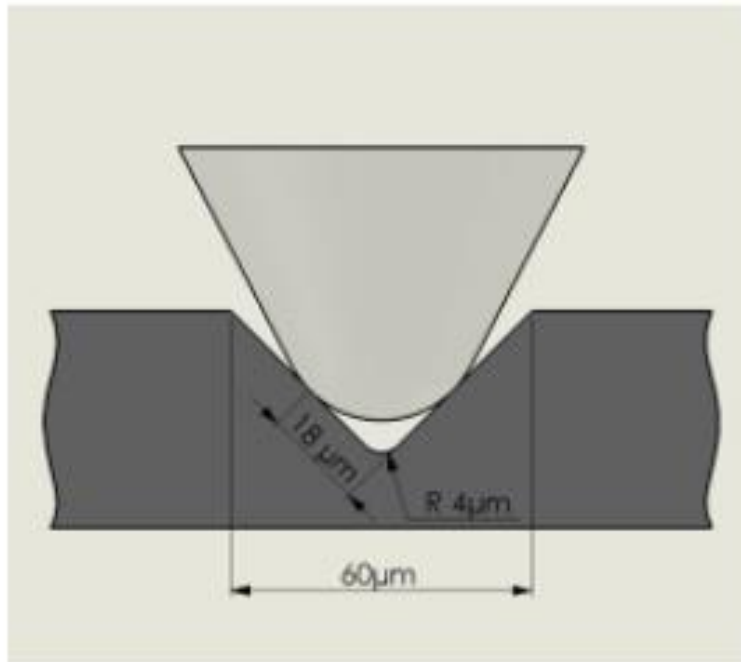


Figure 22-8 Ortofon Stylus Information

18 micrometers correspond to 0.7 mil and the groove shape in the drawing is for lateral modulation (4 micrometers is the shape of the groove bottom) application. Ortofon customer service has stated that the bottom and sides of the stylus are polished.

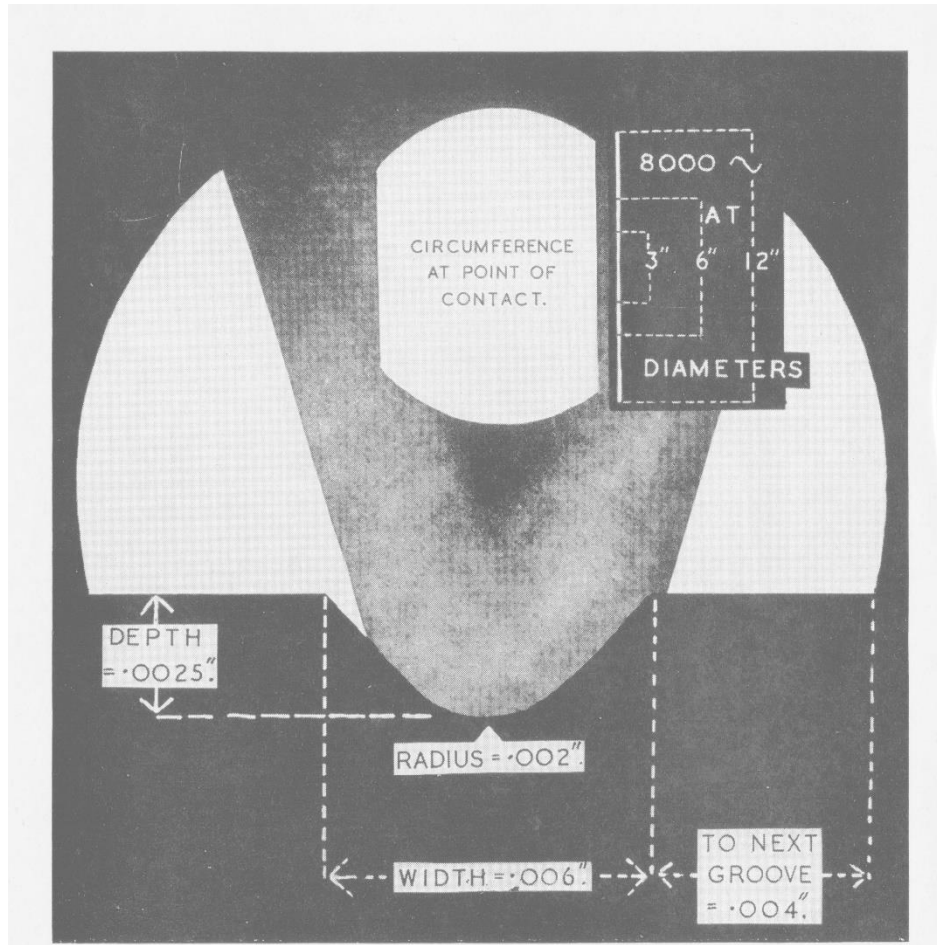
Another very important benefit of using this 0.7 mil tip is that the shape of the noise that remains contains high frequency energy which in turn provides the DCart noise removal algorithms information to remove noise events. When similar records using the larger 3.75 mil stylus are processed with the same noise filters settings, more noise events remain versus using the 0.7 mil stylus.

22.3.3 *Stylus Selection for 78 RPM Records*

Introduction

The original stylus used for playback of 78 RPM records was a soft steel shaped point that was designed to wear *into* the record groove within a couple of record revolutions. Steel

particles and years of dirt will often be found in the 78 RPM record grooves even after a good cleaning. A picture of an actual steel needle worn to fit a 78 RPM record groove can be found in a book by G.A. Briggs titled "Sound Reproduction". This picture is in Figure 22-9.



By C. E. Watts, *Sunbury on Thames.*

FIG. 66.—Diagram showing steel needle worn to fit the groove, with resultant flats.

The flat on one side of the needle is greater than the wavelength of 8,000 cycles even at 8" diameter, and reproduction of this or any higher frequency is therefore impossible under Fig. 66 conditions.

Figure 22-9 Steel Needle in Sound Groove

In the top/right of the picture is a comparison of the size of the worn stylus compared to a wavelength at 8000 Hz, to locations on the record. Clearly the large effective stylus size causes limitations to the music.

Steel needles were later replaced with hard sapphire needles. Many 78 RPM records still contain steel particles and dirt from years ago.

The stylus selection for 78 RPM records requires a shape that does not ride on the bottom and a width that is in contact with the smoothest part of the record groove. The best approach is to have a selection of stylus sizes to experimentally find the best transfer since each record has a unique groove wall location for minimum wear. Two examples will use the original recommended size and a wider size to *ride higher* on a groove wall, for less noise.

Worst Case Calculations for Two Stylus

Experimental results have found that a 3.5 mil spherical shape stylus provided a good transfer for many very worn 78 RPM records. 78 RPM records were designed for a stylus with a radius of about 2.5 mil. The 3.5 mil stylus and the 2.5 mil version will be compared for worst case conditions at the inner radius using 3 kHz and 5 kHz music waves. The equations come from an analysis from section 22.3.1.

First Case: Inside Track (2.0 inches) and 5 kHz sound wave. Using the graph from figure 22-3 and equation 22-9, a 2.0 mil radius gives 20 % total harmonic distortion and a 2.5 mil stylus is $2.5^2/2.0^2 = 1.56$, $1.56 \times 20\%$ or $\approx 31\%$. Using similar calculations, a 3.5 mil stylus gives total harmonic distortion of $3.5^2/2.0^2 = 3.06$, $3.06 \times 20\%$ or $\approx 61\%$.

Second Case: Inside Track (2.0 inches) and 3 kHz sound wave. Using graph from figure 22-3, a 2.0 mil radius gives 8 % total harmonic distortion and a 2.5 mil stylus $\approx 12\%$. For a 3.5 mil stylus and again using figure 22-3 the total harmonic distortion is $\approx 24\%$.

When comparing these numbers with those of the Diamond Disc recall that a larger amplitude of the sine wave has a *squared effect* on the lateral distortion and that the Diamond Discs used a very low amplitude versus the lateral records. The result from using the graph shows that the best stylus to *minimize* the distortion from the recording-playback process is to have a small radius that *does not ride in the bottom of the groove*. With the worn condition of many 78 RPM records, a larger radius helps to reduce the noise from the surface with the tradeoff of more harmonic distortion on the high frequencies.

23.3.4 *Conclusion for Best Sized Stylus for Minimum Distortion*

The mathematical analysis presented in the book “Sound Recording” by Frayne and Wolfe, which built on the original work from Pierce and Hunt, provides useful mathematical relationships between the stylus radius and distortion of the recorded sound groove for both Lateral and Vertical recording. For the Edison Diamond Disc records, calculations were made that demonstrate that a current, DJ style stylus of 0.7 mil radius will provide a low value of distortion for the Diamond Disc Records. This small radius can be used because the stylus rides on the bottom of the groove where the sound information is located. This contrasts with the 78 RPM records that use lateral motion from the groove walls.

The 78 RPM records cannot use a stylus that rides on the bottom of the groove, because the original record groove was made to support a reproducer and shape a soft steel needle using the *groove* walls. The groove walls for many 78 RPM records contain steel particles and wear from supporting the reproducers. Thus, a stylus that rides higher on a new clean part of the groove wall may provide the best result even though high frequency distortion will result.

The original sound wave is placed into the groove using a chisel shape for both vertical and horizontal groove motions and a finite radius to the stylus used to follow or trace the groove motion means that there will always be some distortion to the signal. The lateral motion requires a minimum radius to ride higher than the bottom in contrast to the vertical motion that can have a small radius ride on the bottom of the groove.

Since low mass tonearms and electric amplification are now available, a much smaller stylus radius can be used for the Edison Diamond Disc records than was originally needed. *The result is that for the Edison Diamond Disc Records a lower distortion to the music is possible versus the 78 RPM records. The 78 RPM records will always require a larger stylus radius than the Diamond Disc Records to ride above the groove bottom.* The resulting distortion to the music has been shown to be significant between these two methods.

This section addresses the best shape and size of the stylus for minimum distortion. The next part addresses the amount of force that the stylus exerts on the sound groove due to its shape and tracking force.

22.4 *Selecting the Stylus Tracking Force & Calculating the Groove Pressure*

22.4.1 *Introduction*

A force applied to the stylus to track or response to the groove motions on the record. Force is measured in Newtons using the SI (International System of Units) notation; however, phonograph stylus manufactures use units of Mass, in Grams, for this value. Force comes from a Mass times Acceleration so that Newtons or Pounds would be the correct units, however, manufactures assume that since gravity is somewhat constant, Grams can be used to specify the tracking forces. Today's stylus assembly contains the stylus connected to a mechanical support or suspension that in turn attaches to the cartridge. Since the suspension bends with the applied force, there is a recommended *Tracking Force* that the Stylus manufacture states for use with their product. Typical values today are 2 grams. This is in sharp contrast to Edison diamond disc players that were approximately 190 grams or the 78 RPM players with similar heavy reproducers.

The tracking force *pressure* that bears on the groove depends on the applied force and the area of contact. Both variables combine to produce a value of psi (pounds per square inch). The next section will calculate various psi values for a modern, 2-gram stylus and an Edison Original Diamond Disc stylus used with the Diamond Disc reproducers. This method can be extended to 78 RPM & LP records.

22.4.2 *Circular Contact Area Between Spheres with Applied Force*

When two shapes are pressed together, a resulting contact area occurs over which the applied force produces a resulting psi. The area of the contact contains the total force, so that a very small area can have very large pressures, even with a light force.

With pressure, material will deform and the actual contact area can increase from an initial point of contact that is very small. *Contact Mechanics* is the engineering science that has created equations to describe the interaction between objects and applied forces. The famous German scientist, Henrich Hertz, developed an equation that calculates the resulting contact area when two spheres are in contact. This equation includes variables relating to the material properties, applied force, and the radius of the spheres. This equation can provide a useful comparison between the pressure on the diamond disc groove using the original Edison

reproducer and a current "DJ" style conical stylus with a 0.7 mil radius and 2-gram tracking force.

Hertz Circular Contact Equation

Figure 22.5 Is an Edison Factory Drawing of the Sound Groove for a Diamond Disc record.

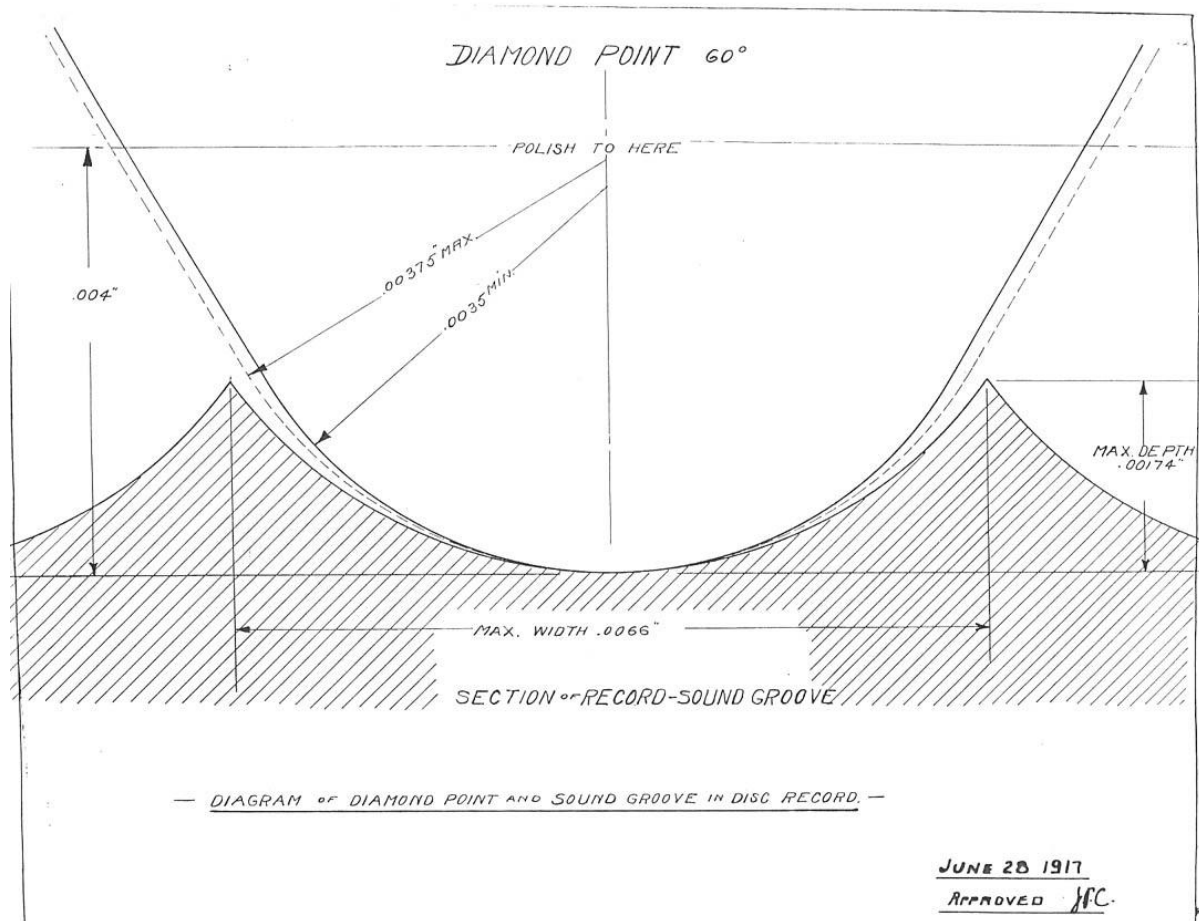


Figure 22.5 Diamond Disc Sound Groove

The equations, developed by Hertz, that calculates the resulting contact area uses a positive number for the radius of a convex shape and a negative number for a concave shape. A positive value for the stylus radius and the recording radius will be used.

Equations

Three equations will be used to calculate the radius of the Contact Circle. The first is the Contact Modulus which includes terms relating to the two sphere materials, 1 and 2.

$$\frac{1}{E_c} = \frac{1 - V_1^2}{E_1} + \frac{1 - V_2^2}{E_2} \quad eq (22 - 14)$$

Where E_c equals the Contact Modulus, E is Modulus of Elasticity for the materials, and V is Poisson's Ratio for the materials.

The second equation calculates the relative Radius with R_1 and R_2 .

$$\frac{1}{R_c} = \frac{1}{R_1} + \frac{1}{R_2} \quad eq (22 - 15)$$

The Radius of the resulting Contact Circle from the applied force is:

$$c = \sqrt[3]{\frac{3FR_c}{4E_c}} \quad eq (22 - 16)$$

Where c is the radius of the resulting circle and F is the applied force.

The Maximum Pressure developed by the applied force and the Contact Circle is equal to:

$$P_{max} = \frac{3F}{2\pi c^2} \quad eq (22 - 17)$$

Examples will calculate the Radius of the Contact Circle and Maximum Pressure for two different stylus radius values.

22.4.3 *Groove Pressure with Diamond Disc Stylus*

Referring to figure 22.5, The stylus riding in the groove will be number 1 sphere. The groove will be number 2 sphere. The radius of the cut sound groove is 5.25 mil (From Edison Patent Information).

Edison Stylus with Diamond Disc

For this calculation, the radius of the stylus is 3.5 mils. The Modulus of Elasticity is 164 million psi for Diamond and 550 thousand psi for Bakelite. The Edison Diamond Disc surface used an Edison product named Condensite, which is very similar to Bakelite. Poisson's Ratio is 0.0691 for Diamond and 0.49 for Bakelite. The applied force, from the Diamond Disc reproducer is 190 Grams or 0.4188 lbs.

The resulting Radius of Contact Circle is 1.05 mil for these conditions and the contact maximum pressure is 181,371 psi.

DJ (0.7 mil) Stylus with 2 Grams Tracking Force

The same values for the materials will be used, with the Stylus radius (1) now 0.7 mil and the applied force 2 grams or 0.00441 lb.

The resulting Radius of Contact Circle is 0.14 mil and the contact maximum pressure is 103,274 psi.

22.4.4 *Discussion of Results*

These values for both cases, are much higher than you would realize and should be understood as a starting point for understanding the mechanics of the stylus and groove interaction. The values for the materials are the best estimate that can be found and improvements are possible to measure the exact values for the Edison Disc material. Engineering studies on contact mechanics refer to how this contact pressure causes micro-movement in the underlying material that is somewhat elastic (obeys Hook's Law) and does not cause permanent damage to the groove.

When the first vinyl records were manufactured, the AES (Audio Engineering Society) published many papers that could *not explain* why the groove were not destroyed with the resulting high contact pressure. Various theories were suggested that the high forces caused elastic motion to the groove that could take up to 30 minutes to *move back to the original location*. This resulted in some audiophiles waiting to replay a record to allow the grooves to return to the correct shape.

The Edison Diamond Disc record players required a heavy tracking force to produce useful sound from an acoustic reproducer, which in turn required a large radius for the stylus. Today, the electric pickup needs a small amount of energy and a light tracking force can be used, *however*, as the calculations show, the light load will still produce a large amount of pressure due the small contact circle.

The current electric cartridge, along with the stylus, states a recommended tracking force for optimum performance. This value should be used, even though the actual pressure on the groove is high (but less than Edison's Record Players).

23.0 Audio Standards

This chapter describes standards that have been developed to measure audio signal levels. These standard measurements provide interoperability between various audio devices when they are connected to make an audio system. When devices are connected to each other the electrical levels need to be compatible and predictable so that when connected they will achieve the desired result. For example, when a CD player is connected to a power audio amplifier and then a set of speakers you expect that all the parts will work together. The volume control on the power amplifier should give a smooth increase from low to high and not start off at a very loud level and increase to a high distortion. This level of expectation requires that the inputs and outputs of the audio equipment use some sort of standard values for audio levels so that different devices work together.

23.1 Logarithms, a Needed Concept

The manner that the human ear perceives changes in pitch and amplitude is not linear but is logarithmic. Logarithms are useful to describe our hearing.

The logarithm of the ratio between two powers, to the base 10, is expressed in bels as:

$$\text{bels} = \log_{10} \frac{P_1}{P_2} \quad \text{eq}(23 - 1)$$

Where the bel is a ratio and by itself has no absolute value without a reference for the 0.0 value. P_1 and P_2 are signal powers.

The bel is too large a value for most audio work so a new term, the decibel (dB), is used which is a tenth of a bel. Therefore:

$$\text{dB} = 10 \cdot \log_{10} \frac{P_1}{P_2} \quad \text{eq}(23 - 2)$$

When the ratio uses voltage or some other value instead of power another change to the formula is needed. For voltage, a squared term is used since power equals voltage squared divided by the real part of the impedance. Therefore, for voltages, the equation becomes (using log properties):

$$dB = 20 \cdot \log_{10} \frac{V_2}{V_1} \quad eq(23 - 3)$$

Where V_1 and V_2 are Volts and V_1 is the reference.

Some examples of the use of dBs in sound (for most people):

1. The minimum change in sound level that can be heard is 1 dB.
2. The threshold of hearing is 0 dB.

23.2 Early Sound Intensity Measurements, the VU Meter

Sound intensity measurements have been needed since the study of acoustics began. An early method to measure sound was a Rayleigh Disk, circa 1882, that would rotate in accordance with the impinging sound wave. The amount of rotation of the disk indicated the strength or intensity of the sound. As audio technology improved during the 1920's to 30's, various companies developed standards for the volume level of sound to allow the audio components to work correctly within a complete system. Initially the work was not unified, and each company and industry used their own methods so that telephone, radio, movies, and record companies all used unique methods for measurement and standard levels.

Late in the 1930's an effort was made by three engineers named H.A. Chinn, D.K. Gannett, and R.M. Morris to develop an industry standard Volume Indicator and Reference Level device for the audio industry. The well-known VU meter and the associated circuitry was the result of their work. This VU meter and the resultant 0 reference reading served to provide a useful device and became widely used within the audio industry. This meter is still used today as either an actual physical meter or as a digital device with a software display. The dynamic ballistics of the meters' mechanical movement was required to meet a specific set of values so that the indicated value would be useful for the complex audio wave shapes to be measured. To understand the reason for the meters' mechanical specification, start with a sine wave at an audio rate of 1 kHz. Refer to the figure 23-1:

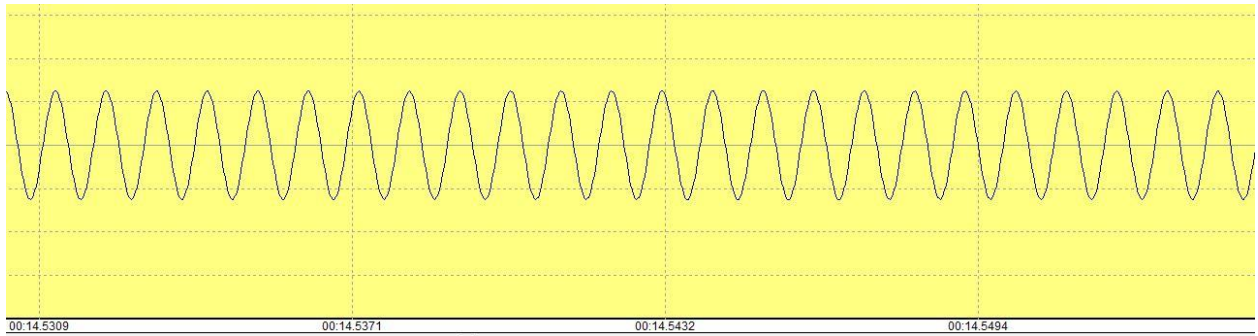


Figure 23-1 1 kHz Sine Wave

For this sine wave, the P-P (Peak-to-Peak) value and the RMS (Root Mean Square) value can be related to each other with a simple formula. The P-P value is the amplitude from the lowest to highest value and the RMS value is the P-P value divided by $2\sqrt{2}$. The power or intensity of this audio wave form is then the RMS value squared divided by the Real part of the Impedance of the load that this wave is applied to.

For music and voice pure sine waves do not occur. An actual waveform from a song is shown in figure 23-2:

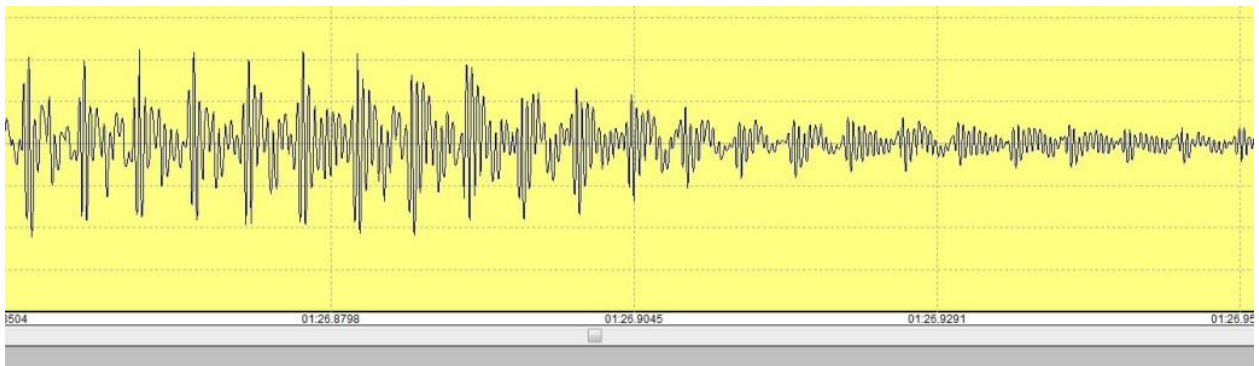


Figure 23-2 Song Wave

To calculate the P-P and RMS values of this song you will need to specify where to perform the measurement. Also, you will need to perform actual integration of the waveform (area under the curve) to measure the RMS value. The term used for sine waves, $\sqrt{2}$, is not valid for *any* shape, just for a sine wave. Since the P-P value is needed to avoid overdrive or saturate of electrical devices and the RMS value is needed to measure the effective power or intensity of the sound wave some type of device is needed that is useful for these complex audio waveforms. The need for a measurement of complex audio waveforms was the reason that the VU meter was designed and had specific mechanical ballistic properties. The response

of the meter's pointer to a complex signal (mechanical ballistic properties) was designed to find a value that was representative of the volume or RMS value of the signal as best that could be for a wide range of different audio waveforms. This meter would then be useful for many types of songs with instruments and vocals. The meter had an initial calibration point where 0 on the scale corresponded to one milliwatt of power with a load of 600 ohms (resistive). The value of 1 milliwatt into a 600-ohm load as a dB reference is known as 0 dBm, where the small m represents the thousandth value. This reference value corresponds to 0.7746 V RMS for a sine wave. Later, the Zero Value was often changed to 4 dBm or 1.228 volts rms.

The VU meter was designed for use with an external attenuator using a calibrated scale in dB. In use, the external attenuator was adjusted so that the music indicated 0 volume units. When the meter's pointer was indicating the Zero value, then the attenuator value was used to give an over-all audio level. The VU meter was used within the audio studio to allow the music producers to set the levels for the different musician's microphones so that the desired balance between the musicians could be made by watching the meter while setting the gains of the audio amplifiers rather than setting a balance by just listening to the music. The VU meter has many other uses in radio broadcasting to provide a useful measurement of the audio sent to the transmitting circuits.

Along with the VU meter additional standards were developed for the new audio equipment available. The DCart11 software provides an audio level meter that is useful but does not simulate an actual VU meter in terms of the meters mechanical movement (ballistics) or 0 dB calibration values.

23.3 Standards for Audio Restoration Work

Audio measurements can use many different units and even have additional definitions for consumer grade and professional grade of audio devices. Another common audio unit is line level for both inputs and outputs. Decibels are common for measurements and they need a reference since the calculation of dB must have some definition for 0.0 dB. Some common units are:

1. 0 dBm is 1 milliwatt of power into 600 ohms of resistance. This equals 0.775 Volts rms.
2. 0 dBV is referenced to 1.0 Volt rms independent of the load resistance. A value of -10dBV would be 0.316 V rms and 0.894 V P-P.

3. 0 dBv or dBu is 0.775 V rms into a High Impedance Load and 2.192 V P-P. The u is to refer to the vu meters' value of 0 for the 0.775 V rms. For Professional Audio use, the 0 dBu value can be greater than the 0.775 V rms reference and instead be 1.228 V rms. 1.228 V rms is 4 dB higher than the 0.775 V rms value.
4. Another term that represents the maximum signal that the converter can accept, without distortion, is known as 0.0 dBFS since the digital output will be all 1's at this point. For example, if the specification said that the peak level: 0 dBFS @ +6dBv that means the maximum output for the A to D converter will occur for an input signal with an RMS value of 2.0.

23.3.1 *Head Room and Alignment Levels*

As shown in figure 23-2, audio waveforms have complex shapes that have a large range from loud to soft passages. The signals complex nature caused the development of VU meters in the past to find a typical or nominal level that represented the most likely power value. Today with digital recording being used, the audio industry uses a phrase called alignment level to represent the 0 value on the VU meter scale or the word *nominal level*. The concept is that the alignment level is set to a value that allows adequate headroom or space from the typical level of the music to the maximum 0.0 dBFS value; in this manner the music will not overdrive or exceed the maximum digital value for the digital word length used. The range for alignment level to maximum is around 20 dB for most audio commercial systems.

For the A to D and D to A converters that are available; most of them just state that they have a line level input and do not tell you what the 0.0 dBFS value is. Thus, the headroom is not specified. Some of the converters used for professional audio use do provide the needed information.

Since the DCart11 software provides meters, we have a way to observe the signals amplitude and can see if it is over-driving or exceeding the 0.0 dBFS value by using the vertical amplitude scale or the DCart11 *version* of a VU meter.

23.4 *Diamond Cut Software Level Indicators, VU, and Vertical Axis*

For the DCart11 software the algorithm receives the output from the converter without any direct information regarding what a maximum value of all 1's equals. When the converter sends a maximum digital value the actual signal level will depend on its design. The DCart11 software assigns a value of 0.0 dB to represent the maximum value from the analog to digital

converter *without a direct linkage to 0.0 dBV or 0.0 dBu*. All that the software knows is that the maximum value will be all 1's and will define 0.0 dB as that value.

For the VU scale that is displayed on the right side of the window the maximum amplitude has a scale value of 0.0 dB at the top of the software meter. The green line is the highest value that has occurred in the song and the white line is a filtered or averaged response. The software meters in the DCart11 software do not correspond to the previously defined ballistic mechanical properties of the original VU meter although there is an adjustable marker reaction time that can be set in the preferences section. The general term VU is displayed by the software but should not be taken to mean that the meter represents the properties from the late 1930's VU meter definitions. The scale can be changed from a display with two scales called linear or log. In both cases the logarithm of the amplitude is shown. The terms linear and log refer to the spacing between the values not the method of calculating the values.

The display of the waveform versus time has an adjustable vertical scale (set in preferences) that can show the output from the converter in digital values or a conversion to dB. For a linear output scale use the counts option.

Figure 23-3 shows a picture while a song is being played with a linear scale using the digital value from the converter. Figure 23-4 shows the same picture with a log scale.

The best method to locate and remove noise from the music (figure 23-3) uses the linear amplitude scale displaying counts.

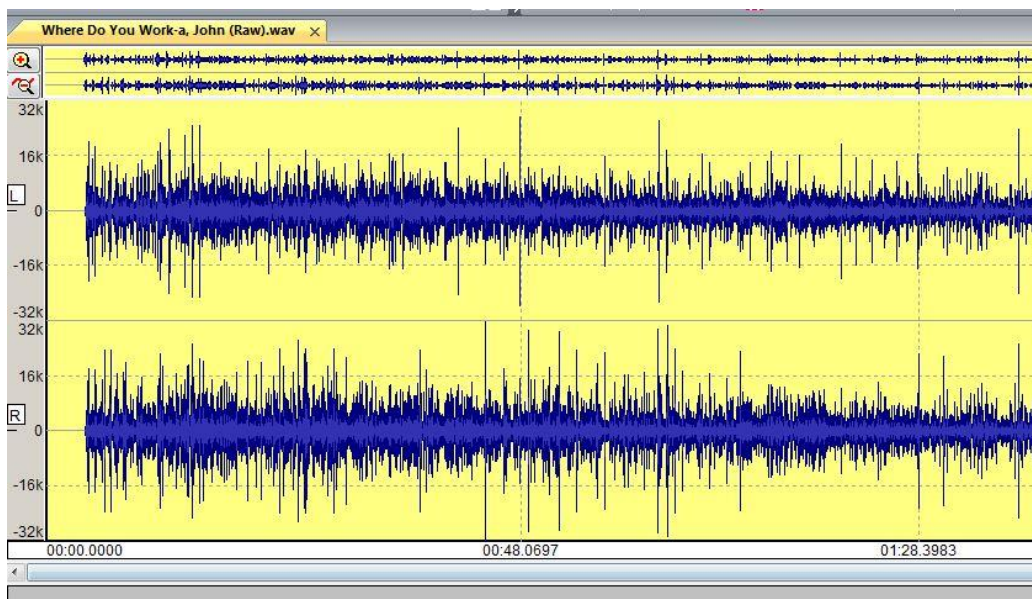


Figure 23-3 Amplitude Scale for Waveform

Figure 23-4 contains the same waveform as in Figure 23-3 with the vertical scale changed to dB.

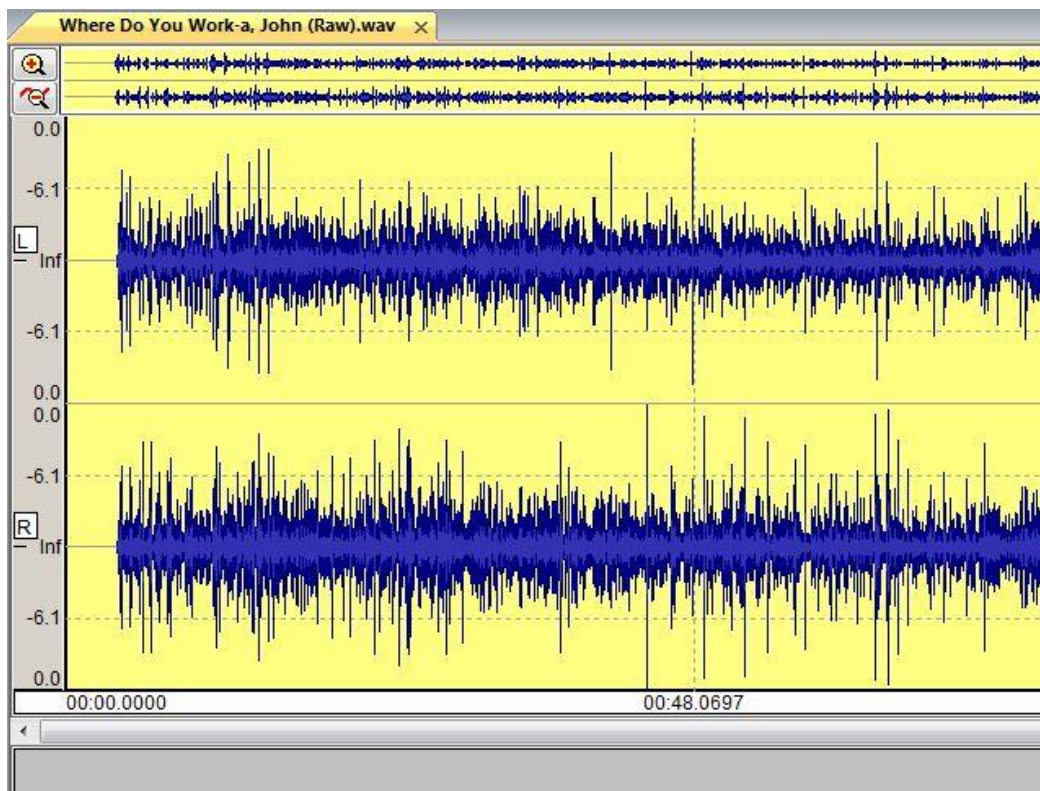


Figure 23-4 dB scale for Waveform

24.0 Correct Recording Level & Number of Bits for Transcription

This chapter will describe the optimum level and number of bits needed to record the music as digital values for your transcription. *Key* concepts will be shown that relate the effective number of bits in the A to D conversion to the selected number of bits.

Calculations using an actual audio system will be done later in this chapter.

24.1 Details of The Analog to Digital Conversion

The signal from the preamplifier, being an analog signal, is a continuous signal. When this analog signal is converted to discrete digital values, that conversion provides a digital value that is close to, but not an exact match, to the actual analog value. How close the digital value is to the analog value depends on how many bits are available to represent the input and how much noise or uncertainty is present in the conversion. Since the digital value is limited to a finite number of bits, the best that can be done in the conversion is to have the conversion accuracy within \pm one LSB or least significant bit. For example, if the input to the A to D ranges between ± 0.5 Volt or 1.0 V full range and there are 8 bits in the word length, then the LSB is:

$$lsb = \frac{\text{full voltage range}}{2^{\text{word length}}} = \frac{1.0}{2^8} = 3.91 \text{ millivolts } eq(24 - 1)$$

This equation shows that the conversion from analog to digital has a finite ability to get an exact value; the conversion will have a potential error of up to ± 3.91 millivolt. While that value is small, it is not zero. As the number of bits is increased the conversion becomes closer and closer to the actual value.

When digital values are used to represent an analog value, a method is needed to indicate that the voltage has a positive or negative value since the analog signals are AC (alternating voltage) with voltage positive and negative. A common method used in digital numbers to indicate polarity is called two's complement format which uses the MSB (most significant bit) to indicate polarity. Using the two's complement format, the largest positive signal is $(2^{b-1} - 1)$ and the largest negative signal is (-2^{b-1}) where b is the number of bits in the digital word. The complement part of the format refers to additional bit changes to the negative word to make math operations less complicated in hardware. There are other methods used to indicate the polarity of the digital word, but they also use a part of the digital

word for this purpose and therefore the number of bits that represent the analog signal is slightly less than the total number you start with.

You may believe that the best approach for transcribing music would be to use the maximum possible number of bits available for the analog to digital conversion process. A practical limit to this number is defined as the effective number of bits (ENOB) in the conversion, which occurs with the presence of electrical noise. The electrical noise, in effect, reduces the number of bits available for the signal by using some bits *just for noise*. Additional bits greater than the ENOB is a waste of system resources and adds no value. A useful concept in Audio Restoration to understand this limitation will be the term *Signal to Noise Ratio*. This term is also used in other fields of engineering. As the equations for the signal to noise ratio are developed, they will show how the noise and the effective number of bits interact with each other.

24.2 *Signal to Noise Ratio (SNR) for Audio*

The Signal is what is desired, and Noise is anything extra that is present when the signal is applied to a device.

The Signal Power to Noise Power, as measured in Decibels, is defined as:

$$SNR_{dB} = 10 \cdot \text{LOG}_{10} \left(\frac{\text{Power Signal}}{\text{Power Noise}} \right) \text{ eq}(24 - 2)$$

Using the Statistical values of the signal, the relationship can also be written as:

$$SNR_{dB} = 10 \cdot \text{LOG}_{10} \left(\frac{\sigma^2 \text{ signal}}{\sigma^2 \text{ noise}} \right) \text{ eq}(24 - 3)$$

Where σ^2 is defined as the Variance of the signal and σ is the RMS or one sigma of the signal. This equation is further modified using logarithmic relationships and with an assumed load of 1.0 ohm:

$$SNR_{dB} = 20 \cdot \text{LOG}_{10} \left(\frac{V1}{V2} \right) \text{ eq}(24 - 4)$$

Where V1 is the voltage of the Signal and V2 is the voltage of the noise in RMS.

24.3 A to D Noise

There is a source of noise within the A to D conversion that occurs from the uncertainty in the LSB value known as *quantization error*. This is not the same as external noise within the system; rather this is noise due solely to the analog data being converted to distinct digital data values. The value of the quantization noise also involves the relationship between the input signals shape and the range of the conversion process. Here the shape refers to how much of the range of the A to D conversion is used for the bulk or majority of the signals values as a comparison to the potential or full range of the converter. For example, if the range of the A to D converter is 2 volts and the signal is only 25 millivolts P-P in value, then much of the converter's range is not used. The opposite of this is the case where the signal uses the total converter's range with all the possible digital values produced. An electrical term to define and measure this is called LF (Load Factor) and is defined as:

$$LF = \frac{\text{rms of the input signal}}{V_p} = \frac{\sigma \text{ signal}}{V_p} \quad \text{eq}(24 - 5)$$

Where V_p is the converter's peak voltage and $\sigma \text{ signal}$ is the one sigma-value of the signal.

After some mathematical work and including the Signal Power to Noise Power (decibels), the relationship showing how quantization (converting an analog signal to a digital value) and signal shape works to define the digital word bits is:

$$SNR = 6.02 \cdot b + 4.77 + 20 \cdot \log_{10}(LF) \quad \text{eq}(24 - 6)$$

Where b is defined as the number of bits in the converter's digital word.

This equation is often seen in a form with the assumption that a sinewave is the sole input to the A to D converter *and* that the peak of this signal is the same value as the peak converter's maximum value. In this case, the equation reduces to:

$$SNR = 6.02 \cdot b + 1.76dB \quad \text{eq}(24 - 7)$$

Equation 24-6 provides a measurement for the analog to digital converter's performance related to the signal to noise ratio. The equation provides important information relating the Signal to Noise to the number of bits used to digitize the analog signal. We want the SNR

value from the converter to be much larger than the audio signal SNR value so that we do not *add* extra noise to the music during the recording process. Some examples of the SNR values using the simplified equation (24-7) are:

1. 8 bits yields 49.92 dB
2. 12 bits yields 74 dB
3. 16 bits yields 98.08 dB
4. 24 bits yields 146.24 dB

These values, are not accurate in a real audio system. In the next section a relationship will be developed that shows how the noise *in the system* further degrades the actual or effective number of bits for the analog to digital process.

24.4 Measuring Signal to Noise Ratio and Effective Number of Bits

When any noise is present in the analog to digital system the ability to use all the digital bits is affected. For example, if the electrical noise is greater than the input analog signal, then the conversion to a unique digital value will not happen as the noise signal will add to or subtract the desired value. Noise that occurs in the recording system comes from many different sources. These sources are often not related to each other or are *uncorrelated* with each other; the typical audio analog signal does not have noise present that will be occurring in exact timing with other noise in the preamplifier or the converter. Before an equation can be developed a modification to the SNR term is needed to reflect how the noise powers and signal powers interact. The concept is to have a term that can be measured in a real system that has a signal present along with noise and distortion. A new term labeled SINAD will be used with stands for signal to noise and distortion ratio.

The equation for this term is:

$$SINAD = 10\text{LOG}\left(\frac{(P_{signal} + P_{noise} + P_{distortion})}{(P_{noise} + P_{distortion})}\right) \quad eq(24 - 8)$$

Where P is defined as the Power of a Variable.

For signals, the RMS and Variance can be used to determine the Power by squaring the result. Therefore:

$$SINAD = 10LOG\left(\frac{(RMS_{signal}+RMS_{noise}+RMS_{distortion})^2}{(RMS_{noise}+RMS_{distortion})}\right) \text{ eq}(24 - 9)$$

A common equation for the ENOB is:

$$b_{eff} = \frac{SINAD - 1.76}{6.02} \text{ eq (24 - 10)}$$

Where b_{eff} is defined as the effective number of digital bits.

This equation is the same as equation (24-7) when solved for digital bits and SINAD substituted for SNR. This equation shows that the effective number of bits (b_{eff}) is equal to a number that is dependent on the SINAD or how much noise and distortion is present and not the actual number of bits in the A to D converter. This equation is a general expression that demonstrates that the presence of noise in a system will have a significant effect on the actual conversion to a digital value.

The audio recording system that is used to transcribe the music uses an A to D converter that will offer a variable number of bits in the digital word to represent the analog value. Equation 24-10 can determine the actual number of bits that are used to convert music on a record to a digital word for an actual audio system.

24.5 System Example of SINAD

The DC Forensics11 software, provides a method to measure the Effective Number of Bits on a recording system, by using the Waveform Statistics Function. This software algorithm is not available on the DCart11 software. The audio system described in Chapter Four will provide actual values of ENOB.

An Edison Diamond Disc number 51408-R, titled “That’s Georgia” will be used for the measurements. This was a record recorded just before the change to electric recording, which represented what was possible for acoustic recording. For the tests, the original Stereo

recording had been converted to a Monaural recording (vertical motion L-R). The recording was from a flat preamplifier with no record noise removed. The monaural recording was used to provide a signal for the Forensics11 Waveform Statistics Function.

The first step is to define *where* the Signal to Noise measurement will be performed, as there are three main locations for the noise measurement.

1. The Total recording system from A to D up to and including the record cartridge.
2. The equipment from step 1 and including the noise from the record groove wall (surface noise).
3. The result from 2 and additional noise from the recorded music.

These locations are shown in figure 24-1.

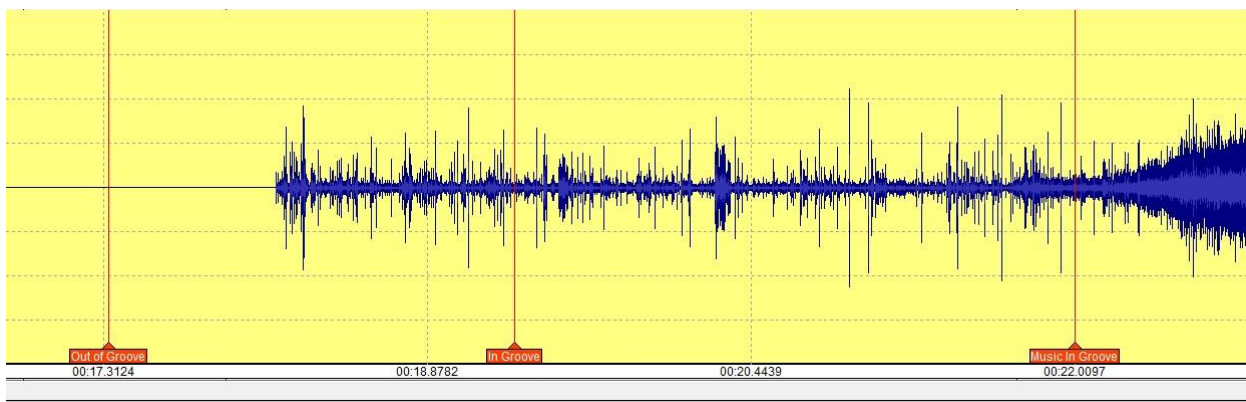


Figure 24-1 Sections of Recorded Music

Prior to the start of the music there is a time where the DCart11 software has started recording and the cartridge stylus has not yet touched the record and started playing music. This location has a marker labeled Out of Groove shown in figure 24-1. Next, the Stylus is in the record groove, but the music has not started. This location is shown in figure 24-1 as In Groove. Later, the music has started, and that area is shown with the marker labeled Music in Groove. The markers are in the general area where the music and quiet section are located; they do not show the actual boundaries of these events.

The beginning section number One, while quiet, does have some noise present. To measure this noise, which is the system noise with no signal present, the section is highlighted and the Forensic Waveform Statistic Function is used. Refer to figure 24-2 for the results.

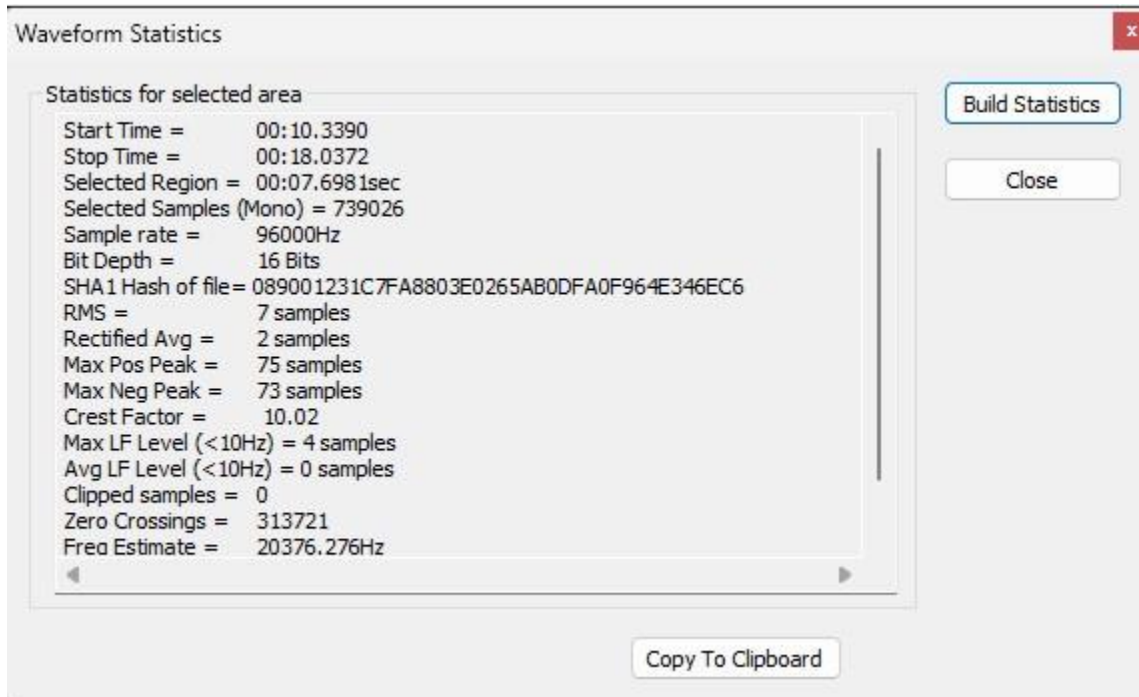


Figure 24-2 Quiet Section (System Noise)

The next section number Two, contains the groove noise before the music starts. Refer to figure 24-2 for the results of the Waveform Statistic.

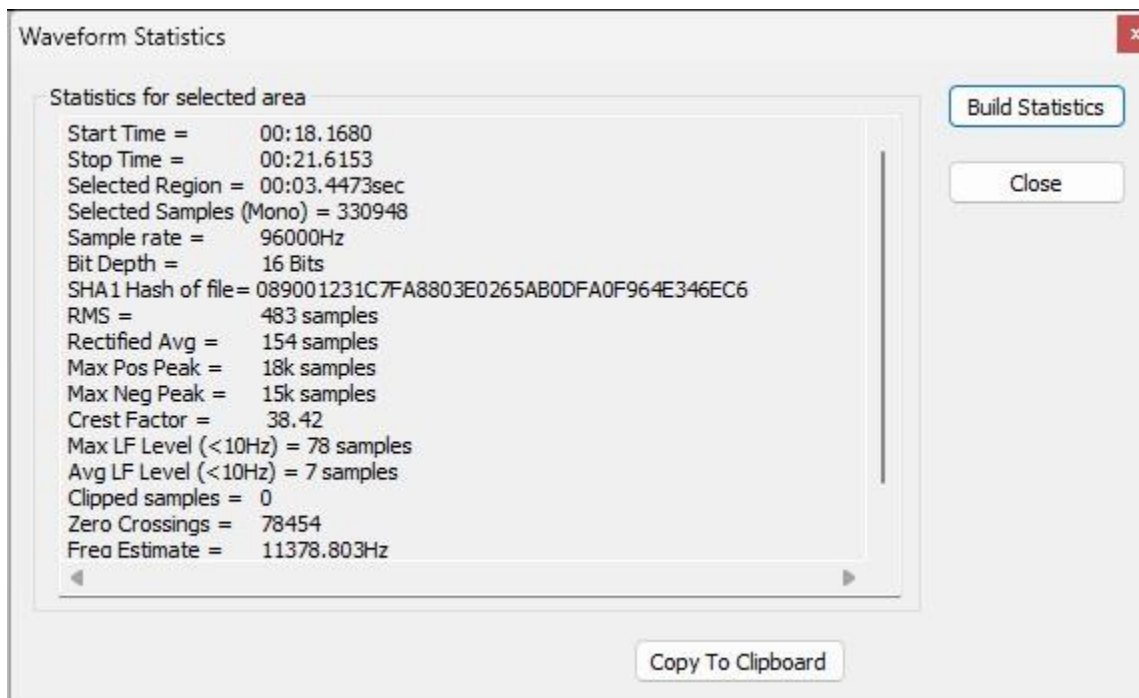


Figure 24-3 Groove Before Music

The last section, number Three, contains the Music. The results of the Waveform Statistics are in figure 24-4.

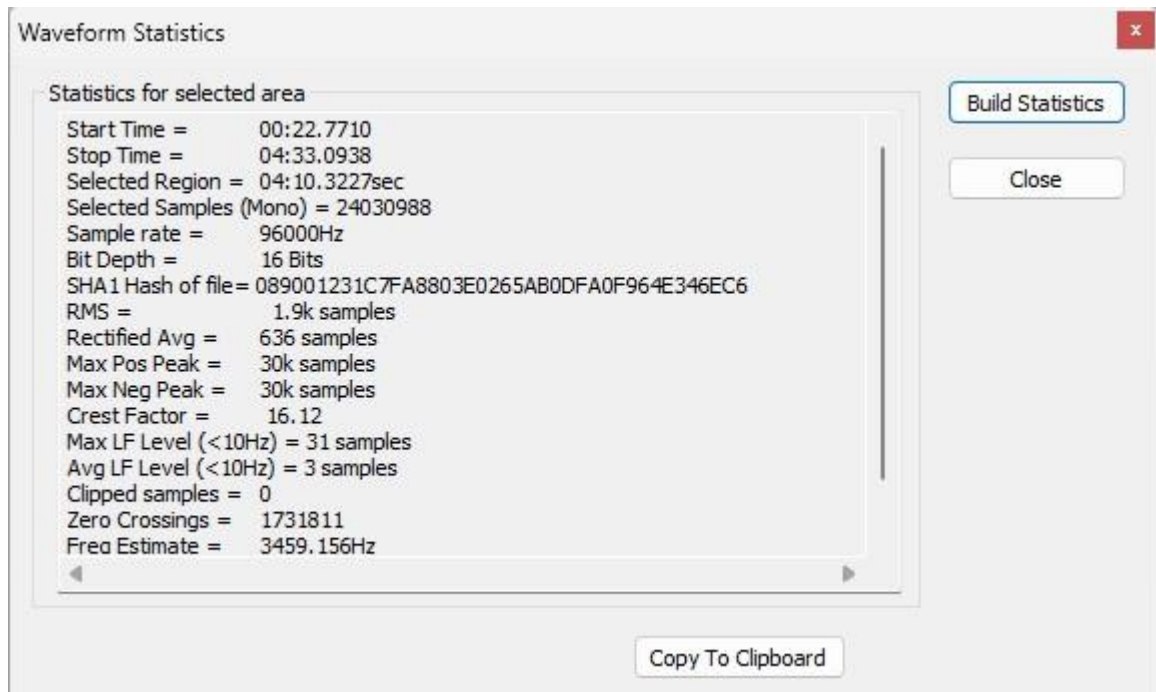


Figure 24-4 Waveform Statistics for Music

Equation 24-9 can be used to calculate the *SINAD* for the Music to the System noise.

1. The RMS of the Music section was 1900 counts.
2. The RMS of the Quiet section was 7 counts.

Using equation (24-9):

$$SINAD = 48.67 \text{ dB}$$

Now, using equation (23-9)

$$b_{eff} = \frac{(48.67 - 1.76)}{6.02} \approx 8 \quad eq(24 - 10)$$

Even though we started with 16 bits in the converter we really have about 8 bits of real digital information representing the music with the lower 8 bits just representing noise. The

8-bit value is an *approximation* but if we started with 16 bits for the converter, we would have more than an adequate range for this music. It is important to note that the calculation for SINAD uses the raw recorded music *before* noise removal and normalization by the DCart11 software. The final SINAD result will be better with noise removed.

The relationship between the Music to Groove SINAD can also be calculated using equation 24-9. The result is $SINAD \approx 12$ dB. After music restoration, this term would increase.

24.6 Conclusion

The calculation of SINAD is a useful concept. If the literature for audio recordings is searched for signal to noise ratio of records; the reported performance can range from as low as 30 dB for early and worn 78 RPM recordings up to as high as 60 dB for special vinyl records used in transmitting music in radio stations. The value of signal to noise for Diamond Disc Records of 48.67 dB, measured in the experiment, is within the general range for audio recordings.

The general subject of audio noise and how it is measured is more complex than the example shown here. There are many additional measurement methods and techniques that have been developed in the audio field that allows a precise measurement of SINAD and the resulting ENOB. For example, the Load Factor (LF) could be measured rather than assumed to be the best that could be used since the simplified equation (24-7) was the basis of the ENOB. An actual measured value for LF would decrease the ENOB value.

For our needs in audio restoration the answer for the number of bits to use is direct and relatively simple; namely the use of *16 bits* for the word length of the digital conversion will provide a potential high value of SNR of 94.56 dB (equation 24-7) and since the typical recording system has noise from the record as the significant term the SINAD is much less than the potential value.

The preamplifier voltage gain used in the transcription, using equations (24-5) and (24-6), demonstrates that the gain should be adjusted so that the Load Factor is as large as possible; which will reduce the SNR number of the A to D process to the minimum possible amount. The Peak Voltage or V_p that the converter uses is a fixed value while the one sigma or RMS value of the music can be varied by using the preamplifier gain control. You can set the gain so that a small amount of head room of about 3.0 dB remains which lets the range of the song extend over a wide range of the A to D process. Thus, the loudest parts of the music should be around 3.0 dB below the maximum value.

25.0 Providing the Needed Preamplifier Gain for Transcription

25.1 Introduction

The analog signal produced by the phonograph cartridge is very low in amplitude when the music is played. The signal needs to be increased as it is fed into the line-input on the converter. The question for restoration work is how much should the cartridge output signal be increased as it connects to the A to D converter.

The amount of amplitude increase needed to the signal from cartridge to A to D Converter can be calculated in the following manner. First the desired result of the digital music file is to have the best or largest Signal to Noise Ratio. The previous chapter covered this in detail, while this chapter will determine the amount of overall gain that is needed for a music signal to produce more than the maximum A to D output.

This example will use the audio system in chapter four and can be adapted for your own audio system. For the amplitude of the signal or S value, the needed gain is to have the ability to always exceed the maximum input of the converter. The 0.0 dB level (maximum) for a typical converter is about 1.0 Volt P-P (Peak to Peak) and would correspond to the maximum digital value. Therefore, the preamplifier's gain should be able to increase the output from the cartridge, to a value exceeding 0.0 dB as shown on the software VU meter. The best practice is to have additional gain control past the 0.0 dB value so that a wide range of record levels can be played.

For a moving magnet Stanton 500.V3 cartridge, the stated output is 4.6 mV (P-P) into a 47 K ohm load while playing a 1 kHz tone. Much of this specification depends on the specific type of recording; however, the 4.6 mV value provides a starting point.

To increase the 4.6 mV output to 1.0 V P-P, a gain of 217.39 is needed. This gain value can also be represented as ≈ 47 dB.

25.2 Preamplifier Gain Values

Most of the commercial preamplifiers do not state how much gain the device produces. Also, most preamplifiers have a fixed. The expected operation by the user is that the gain can be varied by using the software controls on the converter. When a signal is low, and it is amplified any noise that is present is also amplified *by the same amount*. The Load Factor of this conversion process is also low and as demonstrated will increase the system noise. The

best method is to have a large signal swing during the initial transfer, and this will in turn provides the best signal to noise ratio for the restoration work.

Two different methods are used in my Music Studio. Both solutions use a fixed gain flat preamplifier (DCP-47K-F from Diamond Cut Productions). The first solution was to send the preamp output to a Behringer analog mixer (Xenyx 502) with variable gain on each channel and a final overall value. The mixer output then fed into the line-in input channel on the Realtek Chip Set A to D converter. The flat preamplifier from Diamond Cut Productions had been modified to provide a fixed gain of 26.52 dB. This value was chosen so that the output of the flat fixed gain preamplifier would match the expected input level of the Behringer mixer device, thus providing a large range of gain control.

A second solution was to use a variable gain A to D converter from the Focusrite company (Scarlett 2i2 Gen3). The same fixed gain flat preamplifier provided the input to the Focusrite Converter. The Focusrite Converter features adjustable gain on each channel. This variable gain A to D converter, when used with the flat preamplifier, provided an alternative to the Behringer mixer and the Computer Realtek Chip Set.

Both methods provide the needed extra gain, from the flat preamplifier, for the music transcription. The Behringer mixer allows the use of other analog signals besides the turntable, to be converted to digital values.

26.0 *Sample Rate Needed for Restoration*

This section will cover the selection of the sampling rate used for audio restoration for *both* the accurate reconstruction of the signal *and* needed DSP (digital signal processing) in the DCart11 software.

26.1 *History of Sampling Analog Data*

When the question is asked: What is the correct sample rate to accurately convert an analog signal to a digital value? The first thought is that the rate should be at least greater than twice the highest frequency. While this comment is true it needs some additional understanding to make sense when restoring music. Some early history into sampling theory can provide answers.

An early Radio pioneer of was Reginald Fessenden. Fessenden developed several major principles in Radio during the beginning of the 20th Century. He developed the heterodyne frequency principle and has been credited with making the first broadcast of the human voice over radio waves. His first broadcast occurred in the fall of 1906 from a station in Massachusetts. At this time, the phrase Wireless Telephony was used to describe what we now call AM (Amplitude Modulation) Radio. His work was performed without any vacuum tubes or solid-state devices. To generate the high frequency RF energy for transmitting over the air, Fessenden used a rotating spark gap generator to produce a series of impulses to tuned circuits. This RF energy was then modulated to produce AM by using a water-cooled carbon microphone located in the Antenna circuit while he spoke and played music. He wanted to understand the relationship between the speed of the interruptions by the rotating spark gap device to the audio frequency that was being transmitted. At the time of his experiments, he determined that for voice frequencies a minimum frequency of interruption of 10,000 per second was needed for good fidelity, however, because the same interruptions were used to produce the RF energy, he used a higher rate up to 25,000 per second to make an AM RF signal. He stated in an early report that what was really needed was a RF generator that produced a continuous waveform that would be in turn modulated by the audio frequencies. The continuous waveforms were later developed by using rotating high-speed alternators, arc spark gaps and then the vacuum tube. What is interesting is that this early work with the rotating spark gap was in effect an early sampling device and while voice frequencies are at most 3000 Hz, the value found through experiments of 10,000 Hz was not too far from the common 8000 Hz number currently used in telephone circuits. Fessenden presented a paper

on the history of Wireless Telephony to the Smithsonian Institution in 1908 that provided this information, and that paper is a good source of Radio information.

In 1922, another Radio pioneer name Edwin Armstrong invented a modification of his Regenerative patent that greatly improved the performance of the circuit by interrupting the oscillation in the detector stage at a high audio rate around 20,000 Hz. He called this new idea Super Regeneration. Armstrong had previously invented Super Heterodyne for radios and would go on to invent FM (Frequency Modulation) Radio.

By interrupting the regenerative feedback, the circuit was able to increase the gain of a single tube to over one million. Today this same circuit is used in ultra-high frequency circuits to provide reception with high gain and low cost. At the time that Armstrong invented the circuit he specified that the interrupting frequency had to be about two times the highest audio signal to work correctly. It is not clear from reading the patent for super-regeneration if the mathematical principles for sampling theory were understood by him at the time, but his factor of two was correct.

In 1928, Harry Nyquist published a technical paper, “Certain Topics in Telegraph Theory” that was presented at the Winter Convention of the A.I.E.E. The paper listed several topics centered on the technical aspects of transmitting code signals over wires for the important telegraph industry. This paper describes many principles of digital communication that are used today with wireless transmissions. These important principles would be built upon by many more scientists to provide the needed technology that we use today for our wireless communications. One of the areas that Nyquist discussed in this paper was the relationship between the bandwidth of the electrical circuits that transmit the signal and the frequency content of the pulse that is to be sent. A careful reading of his paper does not reveal all the needed description of a complete sampling system that converts an analog signal to digital and then back again to an exact analog signal. Rather, the work by Nyquist was built upon by Claude Shannon and others to create the Sampling Theorem that we use today for analog to digital and digital to analog conversions. The phrase *Nyquist Sampling Rate* is often used to encompass the work started by Nyquist and later developed by others.

26.2 Sampling Theory

Sampling Theorem details are available in many communication references today if you want to dig deeper into the theory. The core of the idea is that if you can limit the bandwidth of the applied analog data to be less than a certain frequency, f , then the minimum sample rate needed for reconstruction of this band-limited analog signal will be greater than

twice the f value. For example, audio signals that range from 20-20,000 Hz need to be sampled at a minimum rate slightly more than 40,000 samples per second. While this example is correct, there are more practical details needed to make this work.

The sampling theorem requires that the maximum frequency be limited to a value using an electrical filter made of various circuit elements. From the theory of electrical filters, a perfect filter that limits frequencies to an exact upper value cannot be built unless you have infinite time to wait for the results. This means that real filters have rounded edges to the frequency response and do not drop down to a zero value. The result from analog filter design is that you need a guard band or some extra frequency range so that the filters performance is correct for the analog sampling value to work. When the digital to analog conversion occurs at the end of DSP operations a filter is also needed to limit the output frequencies. The filter on the end of the D to A conversion will keep unwanted replicas of the audio frequencies, at multiples of the sample rate, from being produced. The result of the need for practical electrical filters and to be at least higher than twice the highest audio frequency is one of the reasons that CD music uses a 44.1 kHz sampling rate. Within the audio community the subject of what is the needed rate for good music has prompted many pseudo-science discussions and will not be discussed.

The science for sampling theory is solid and the principles developed by Shannon and many others has provided many useful products for us.

26.3 DSP Needs for Sample Rate

DSP filters are implemented using either an FIR (Finite Impulse Response) or IIR (Infinite Impulse Response) topology. Each of these methods has advantages and disadvantages but, in all cases, the digital operations are producing results at discrete values determined by the sampling rate. Due to the properties of these filters, the upper limit to the filters response is the sampling rate divided by two. What this means is that if you have a DSP low pass filter that stops frequencies at 20,000 Hz and you are using periodic sampling of 50,000 Hz then the filters performance for frequencies greater than 25 kHz will not be a *low pass* at all. For DSP filters the response repeats at multiples of the sampling frequency and these extra responses are limited only by the practical frequency response of the electrical components in the hardware circuits. What this means for our restoration work is that if we want to perform any filtering of the frequency content a high sampling frequency will allow the DSP operations to keep working above the filters cutoff frequency and perform as

expected. When a low pass filter is used to limit record noise above a value of 10,000 Hz, we expect that this filter will continue to work to frequencies greater than 10,000 Hz.

A practical sampling frequency for filtering work is to use the common 96 kHz value. This sample rate will keep the filters working to at least 48 kHz. This high sampling rate is greater than the minimum Nyquist sampling rate for the music; but it is a result of using DSP to implement filtering.

The various noise filters in the DCart11 software will also benefit from having many samples to eliminate a noise event and replace it with music. 44.1 thousand samples per second will work quite well but using 96 thousand per second seem to find and fix noise events better than 44.1 thousand per second. The filters have more information to remove the noise event and replace the noise with the correct music.

26.4 Caution with Sample Rate Selection in a Converter

While many converters will offer you the ability to *select* a sampling value of 96 kHz this value will not always be the *actual value* that the device implements. To achieve a real 96 kHz rate, the needed clock signals and circuit frequency response must operate at a high speed. Some vendors have implemented a low-cost solution to high sample rates by sampling at a lower rate and then adding extra digital values using special digital algorithms to yield a *pseudo* 96 kHz rate.

The chapter on Audio Measurements describes an experiment to verify that the converter truly sampled the signal at the advertised rate.

27.0 Audio Measurements

This chapter describes some useful measurements that can be performed using the Diamond Cut Production software and laboratory test equipment. These tests can be used to evaluate the performance of your audio system. The DC Forensics11 software was used as this version has a *Waveform Statistics Function*, which can provide needed information on the digital values from the converter. The Forensics11 software builds on the available functions offered by DCart11.

27.1 Measurement of Actual Converter A to D & D to A Performance

For all tests, the ASIO driver was selected in the Preferences Tab.

27.1.1 Input to Record Level Meter Performance

This experiment measured the relationship between the analog signal applied to the Input and the recording level as shown by the software.

A Focusrite Scarlett model 2i2 (Gen3) converter was used as shown in figure 4-4. This converter has adjustable gain controls for each channel, so that a reference level for 0 dB can be set by the user. The INST and AIR options were not used.

For the test, an input from a Siglent model SDG 1032X Waveform Generator was set to 500 mV RMS and then applied to the left channel; while the gain control for that channel was adjusted for a reading of 0 dB on the DC Forensics11 Record Level, in Paused Mode. A sample rate of 96kHz and 16 bits was selected. The Focusrite product provides a visual indication of the level to the internal A to D converter using circular colors displayed around the gain control, and were recorded during the test. The test consisted of applying a 1 kHz sinewave signal to the input while the generator signal level was varied starting with 500 mv RMS for 0 dB. The results are in Table 27-1.

Input mV RMS	dB Shown on Rec Level	Calculated dB (500mV Ref)	Color
500.0	0.0	0.00	Red
445.6	-1.0	-1.00	Yellow
397.2	-2.0	-2.00	Yellow
354.0	-3.0	-3.00	Yellow
315.5	-4.0	-4.00	Yellow
250.0	-6.0	-6.00	Yellow
199.1	-8.0	-8.00	Green
158.1	-10.0	-10.00	Green
99.8	-14.0	-14.00	Green
50.0	-20.0	-20.00	Green
15.8	-40.0	-30.00	None
5.0	-50.0	-40.00	None
1.6	None	-50.00	None

Table 27-1

A graph showing the performance over the input range of 0 to -40 dB, with the reference for 0 dB as 500 mV RMS, is in figure 27-1. The calculated dB value used 500 mV RMS as the denominator and the test levels from the Siglent as the numerator (multiply Log by 20 for dB). The -50 dB value was not graphed.

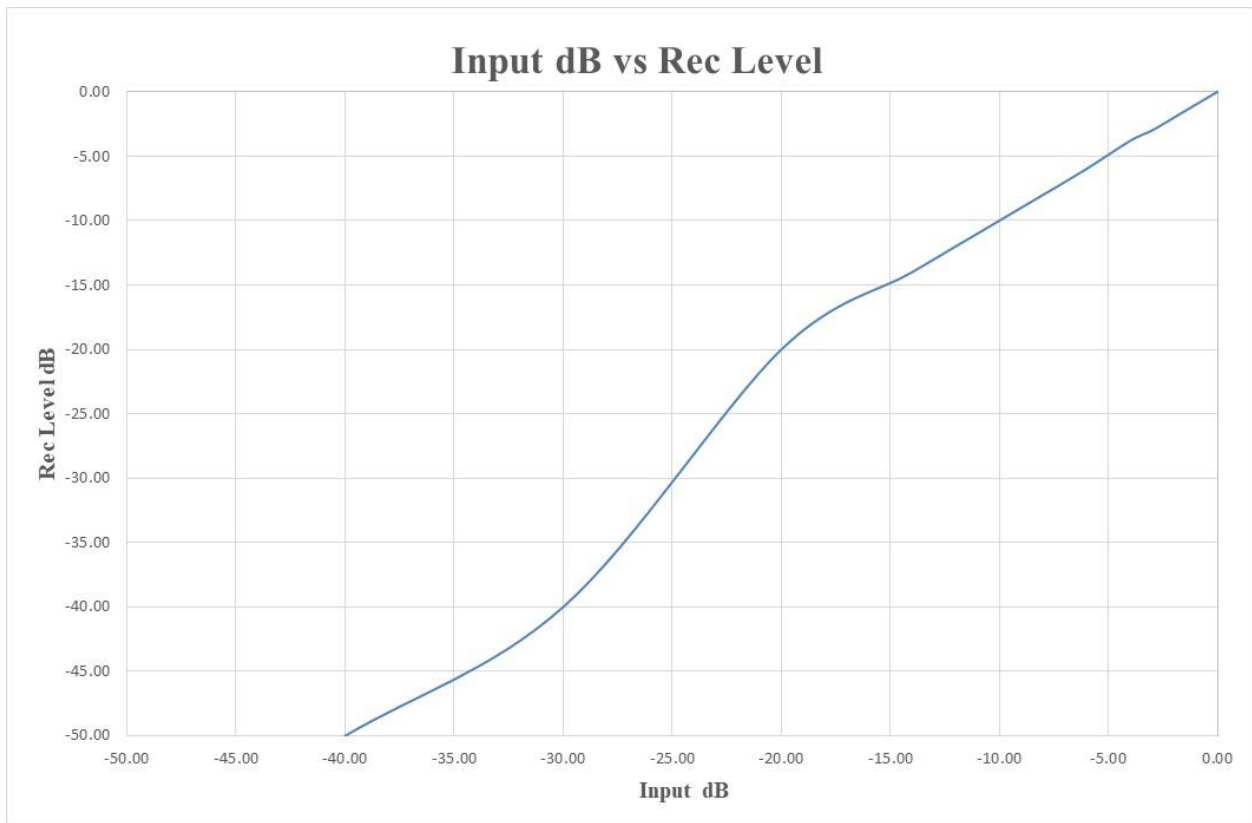


Figure 27-1 Input dB verses Rec Level for Focusrite Converter

The curve is not a straight line, as it should be. The range from 0 to -20 dB is close, but the values below -20 dB are not exact. Also, the low values of the level indicators below -20 dB cannot be read with precision.

The Record Level is useful as a general signal level indicator but cannot be used to accurately measure the A to D performance. The next test measures the performance to a more *detailed level*.

27.1.2 *Input to Digital Count Values*

The input analog signal is converted to digital values by the Focusrite A to D converter. These digital values are in turn used by the software for all file operations. The test in 27.1.1 shows how the digital values are *displayed* as the recording occurs. This test shows the relationship between the input analog signal and the resulting digital count values used by the software.

The test consists of applying a 1 kHz sinewave starting at a 0 dB amplitude, which gives a maximum digital value, for 16 bits, of 32,000 bits Peak to Peak; to a very low value

with a shorted input. A sample rate of 96 kHz was used. The left channel gain control, was adjusted so that a 500 mV RMS applied signal would provide the maximum digital value without clipping the data (The Forensics Waveform Statistics Tool counts bits that are clipped).

The length of the test sample was approximately 10 seconds, and the Waveform Statistics Function was then used to measure the Maximum count value, Minimum count values, and RMS of the waveform over the length of the test sample. This Waveform Statistics Function is available using the DC Forensics11 software. Refer to Table 27-2 for the Test Data.

Input mV RMS	Input Cal. dB	Max/Min Count	RMS Count	dB RMS Count Calculated
500.0	0.00	32k/32k	23k	22627
445.6	-1.00	29k/29k	20k	20166
397.2	-2.00	26k/26k	18k	17975
354.0	-3.00	23k/23k	16k	16020
315.5	-4.00	20k/20k	14k	14278
250.0	-6.02	16k/16k	11k	11314
199.1	-8.00	13k/13k	9.2k	9010
158.1	-10.00	10k/10k	7.3k	7155
99.8	-14.00	6.6k/6.6k	4.6k	4516
50.0	-20.00	3.3k/3.3k	2.3k	2263
15.8	-30.01	1.0k/1.0k	731	715
5.0	-40.00	341/347	230	226
1.6	-49.90	115/122	73	72
0.0	-∞	16/10	5	0

Table 27-2 Analog Input to Digital Count Data

The dB RMS Count Calculated Data, used the relationship that since the input of 500 mV RMS was adjusted 0 dB, this would equal a Peak-to-Peak value of 64 k counts, which in-turn yields a reference value of 22627.42 RMS. This reference value comes from using EQ (27 – 1) for a Peak-to-Peak digital output of 64 k bits, and for a sine wave;

$$RMS = \frac{Peak\ to\ Peak}{2\sqrt{2}} \quad EQ\ (27 - 1)$$

The reference value RMS value for 0 dB was then 22627.42. Next, the calculated dB from the RMS Count Waveform data was:

$$dB = 20 \text{ Log} \left(\frac{\text{RMS Count}}{22627.42} \right) \quad \text{EQ (27 - 2)}$$

These dB results are in the column dB Count Calculated. The Forensics11 Waveform Statistics RMS Count value is calculated by taking the Square Root of the Arithmetic Mean of the squared values of the samples over the length of the waveform. Note that the reported Waveform Statistics value shown is a rounded value. The graph of the Input signal in dB, relative to 500 mV RMS, versus the actual digital count in dB, also relative to 500 mV RMS is in figure 27-2. The data below -50 dB is not shown due to the undefined value of the Log of 0.

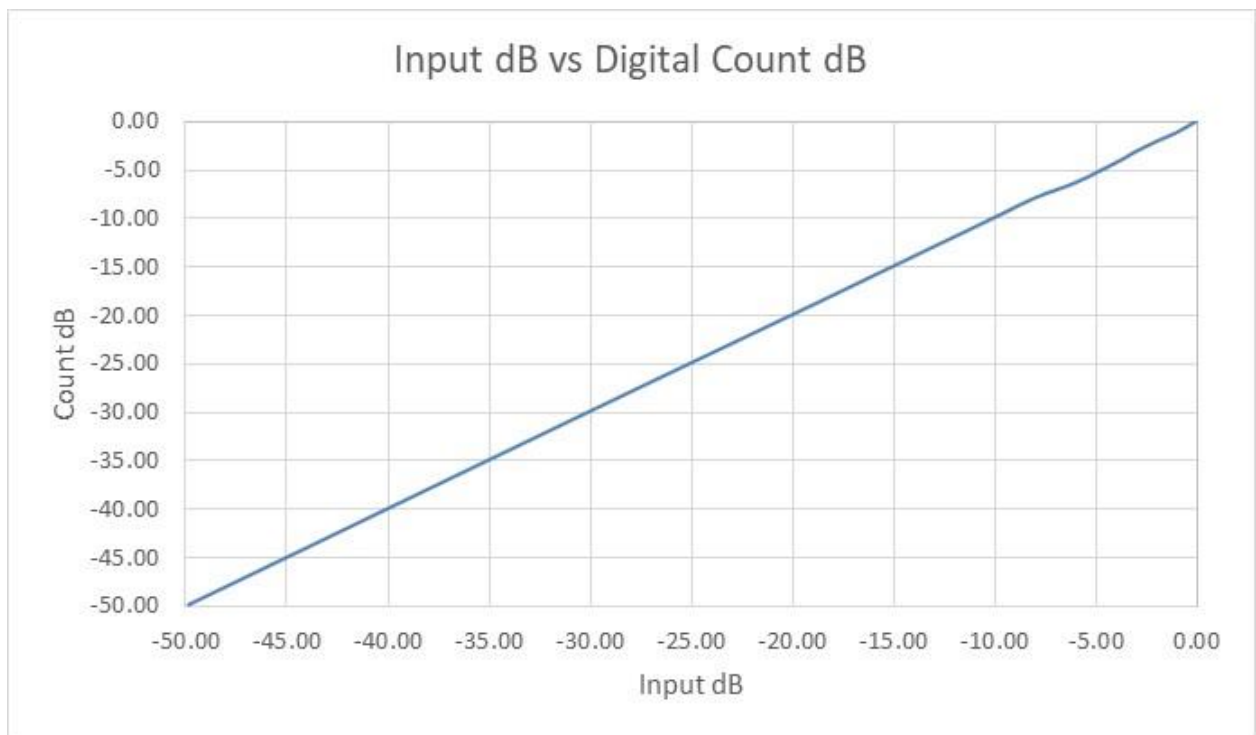


Figure 27-2 Input Analog Signal vs Digital Counts

The graph is a straight line over a very wide range of at least 50 dB. The count RMS value with the input shorted (5) corresponds to a *noise floor* of ≈ -73 dB.

27.1.3 *System Input to Output for Analog Signals*

The same Focusrite converter, as used in the previous tests, was again used for this test. The left channel is used for the input, and the left output channel was connected to an RMS reading volt meter, HP 3400A. The Siglent Waveform Generator is again used for the input signal.

For a 500 mV RMS sinewave (1 kHz), the input gain was adjusted for 0 dB indication on the Rec Level meter and maximum digital count of Peak-to-Peak 64K counts (16 Bits). The Output Gain control on the converter was adjusted for a 500 mV RMS value on the RMS Voltmeter, when the recorded 500 mV RMS waveform was played back. The input level was adjusted from 0 dB down to no input, with the input shorted. Refer to Table 27-3 for the data.

The data from Input to Output, in volts, is plotted in figure 27-3. The same data from Input to Output is plotted in dB in figure 27-4. The minus Infinity input and Output value is not included in the graph.

Input mV RMS	Input dB	Output mV RMS	Output dB
500.0	0.00	500.0	0.0
445.6	-1.00	450.0	-0.92
397.2	-2.00	401.0	-1.92
354.0	-3.00	360.0	-2.85
315.5	-4.00	320.0	-3.88
250.0	-6.02	255.0	-5.85
199.1	-8.00	200.0	-7.96
158.1	-10.00	160.0	-9.90
99.8	-14.00	102.0	-13.81
50.0	-20.00	52.0	-19.66
15.8	-30.01	17.0	-29.37
5.0	-40.00	7.5	-36.48
1.6	-49.90	5.6	-39.02
0.0	$-\infty$	5.6	-39.02

Table 27-3 Input to Output Data

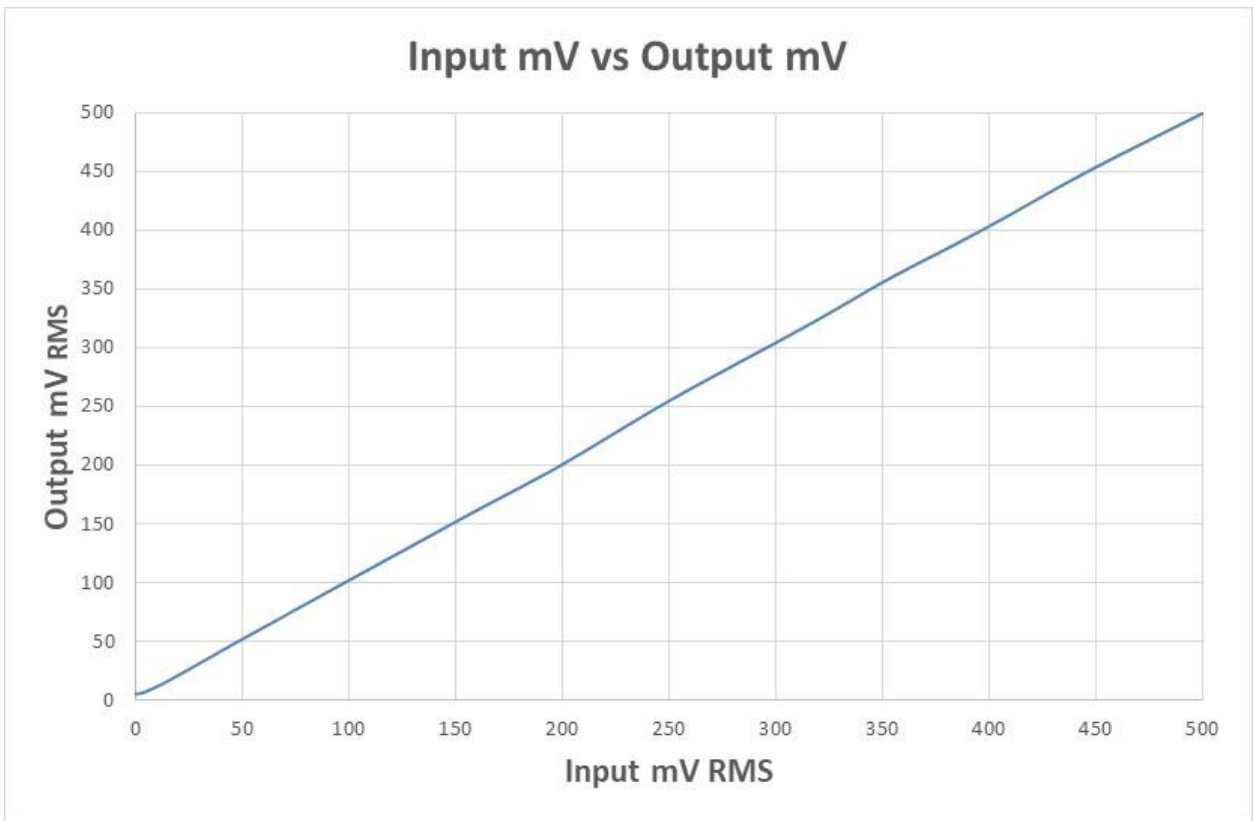


Figure 27-3 Input vs Output for Focusrite Converter and Diamond Cut Software

The relationship appears to be a straight line for a linear range of input values. However, a Logarithmic graph is useful to evaluate audio signal levels. Figure 27-4 graphs the same data using logarithmic results.

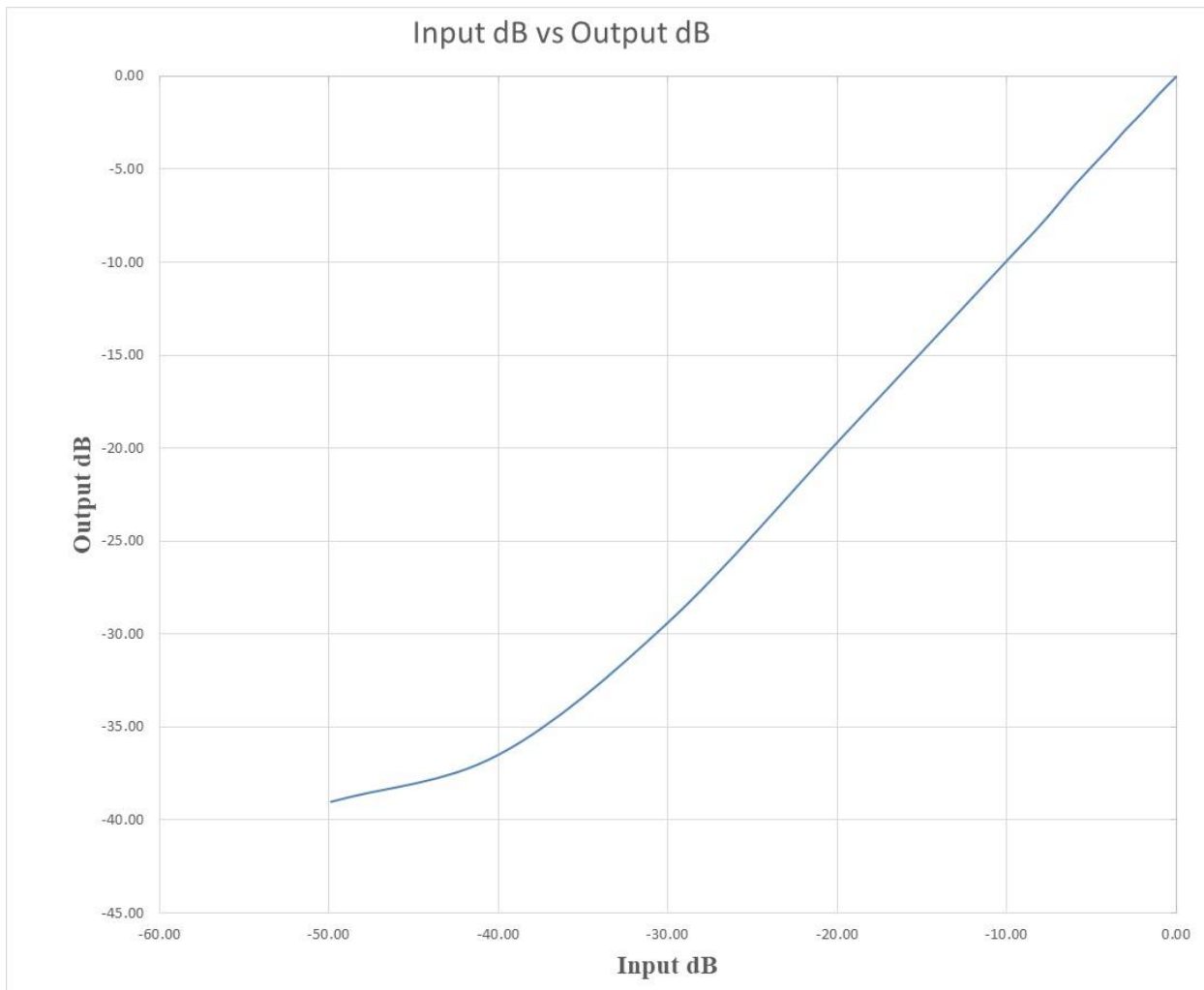


Figure 27-4 Analog Input vs Analog Output

27.1.4 Conclusion

These experiments examined the performance of the Focusrite Converter and the Diamond Cut Software in converting and measuring audio data. Analog audio was converted to a digital value while recording, then stored as a digital waveform within the audio file. Later, the digital values were converted back to an analog value and sent out as an audio signal. Several graphs used a logarithmic scale to reflect the performance as heard by the user, as hearing is logarithmic in nature.

The conversion to a digital value using the recording Level Meter was satisfactory from 0 dB to about -20 dB. Below -20 dB, the performance was poor, however, this is not a problem since these meters are mainly used from -20 dB to 0 dB.

The conversion from the input analog signal to the actual digital count values as used by the Diamond Cut Software, shows satisfactory performance over a range of at least 50 dB for the Focusrite converter and the Diamond Cut Software.

The conversion of the digital values in the Diamond Cut Software to an audio signal via the Focusrite D to A shows a useful range of about 30 dB. Examination of the test data indicates that values below -30dB were corrupted by a noise floor of about 5.6 mV RMS. This noise floor could be internal to the Focusrite converter or due to the measurement equipment. However, a dynamic range of 30 dB for the audio into the speakers should pose no practical problems.

27.2 Verification of Sample Rate in Converters

A test was developed to confirm that the indicated rate on the converter was the actual rate. The Focusrite device previously tested was used and the tested sample rates were 44.1 kHz and 96 kHz.

The test consisted of applying a fixed amplitude sine wave signal of -6 dB to the input while the sample rate selected, and the frequency of the input sine wave was varied. During the test, the dB indicated Rec level meter was recorded while in the Pause State.

The theory was that if the sample rate was correct, the level of the recorded signal should be steady until the frequency became close to or exceeded twice the sample rate, since the input level was held constant for all inputs. The results are shown in Table 27-4 and verify that the *selected* sample rate was the *actual* sample rate.

Input Frequency	Sample Rate 44.1kHz	Sample Rate 96 kHz
20 Hz	-6dB	-6dB
100 Hz	-6dB	-6dB
500 Hz	-6dB	-6dB
1 kHz	-6dB	-6dB
15 kHz	-6dB	-6dB
20 kHz	-6dB	-6dB
22 kHz	-8.5 dB	-6dB
30 kHz	None	-6dB
50 kHz	No Signal	-14 dB

Table 27-4

28.0 *Conclusion and Reader Request*

Writing this book has provided me an opportunity to share with you the joy that I have received from restoring and then hearing *old* music sounding *new* again. My learning process has been a long journey and I wanted to help you along on your journey. I have provided an electronic copy and a paper copy of the Handbook for you to select.

For the paper copy, I have set the font size set to 14 so that you can read the information on the pages while the book is on a table, as you have the computer running the Diamond Cut Productions software. With the larger font size, I have used small margins on the pages to keep the number of pages at a reasonable number. Also, I used a comb binding so that the book could be flat on the table while in use. Please use any *white space* in the Handbook to record your notes.

I would ask that if you find errors in this Handbook to please send that information to me via E-Mail. Also, let me know what you think about this book and if you would like to have more information added to the book in the future.

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Thank you for purchasing this Handbook!

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