<u>Diamond Cut Productions, Inc.</u> **Application Notes**

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The New Way of Transferring Records to Your Hard Drive

Abstract: This paper describes a method for transferring mechanical audio records into the digital domain using a technique that is substantially more accurate than pervious methods. It uses a "flat response" preamplifier and then uses "closed form mathematical algorithms" to reverse the recording EQ curve. We call it "The New Way".

The New Way is simply a whole new approach and new way of thinking regarding the way you play and record your treasured records. This is rather lengthy, but if you bear with us, we think you'll be glad you did.

For those of you who might be put off by the length of this, here's a summary of "The New Way":

1. No longer will we approach Vinyl in the same way we did in 1975
2. We'll use a Flat preamp without a time-constant based RIAA EQ to get our audio into the PC
3. We'll apply the reverse RIAA curve using the Diamond Cut Virtual Phono Preamp and we'll show how to apply a custom EQ curve with will flatten out the signal from your phono cartridge, tone arm and turntable.
4. We'll record at the highest bit width and sample rates available to us using new super high quality audiophile sound cards (24 bits with up to 192 kHz sampling rates).
5. We'll admit that this is no specific reference sound we are trying to emulate. Rather, we'll use DC Six to create just perfect sonic signature matched to our ears, our equipment and our tastes.
6. We'll write this personal sound to the best medium we can – be it either CD or high resolution audio DVD.
7. You will hear audio that is remarkably better!

That's the summary for you speed-readers. Now join us as we explain it all.

The Old Way

First, let's look at how we have traditionally have worked with our mechanical recordings.

Since the late 1800s, music has been distributed to the masses thru the medium of records. If there was a song or artist whose performance you liked, you could buy that performance cheaply and enjoy it at home. In other words, records were meant for playing. Many of us have fond memories of carefully choosing stereo components and setting up our home audio systems in order to get the most enjoyment from these records.

Things changed radically in the 1980s as records were supplanted by CDs. Even though there are still modern recordings being issued on new vinyl, the era of vinyl recordings was over.

But a funny thing happened. People didn't rush out to dump their old vinyl records. In fact, there are many people out there today who still prefer the analog sound of vinyl to the sound of CDs. Even if you like CDs generally, many vinyl recordings never made it to CD or, if they did, suffered from inferior restoration. Let's also not forget that since we already owned the record, we'd actually have to pay twice to buy the CD – an important point for the more thrifty among us!

So vinyl remains a very popular medium 25 years after it "died" and appears to actually be increasing in popularity as many audiophiles continue to search for the best combination of cartridge, stylus, turntable, preamp, etc.

Many of us, however, had a somewhat simpler idea. Why not just play our records into our computer, remove the clicks, pops, surface noise and rumble and make a CD from them? For years, we have thought about this subject just like you did and have offered the best noise reduction and audio enhancement tools on the market – recently culminating in the very successful release of DC Six. In effect, the idea has all along been that we would play our records just like it was 1975, record them into the computer and clean them up. This is a good approach and will yield some good sounding CDs.

But today, we think we can do better because times have changed. Indeed, we are suggesting that we no longer think of our vinyl records the way we did in 1975. We are suggesting that we not simply play them into the computer in exactly the same way we played a popular album such as "Dark Side of the Moon" on our stereos in 1985.

Our present method of restoring our vinyl and ending up with a CD is, in our opinion, more akin to copying a computer disk than it is to lovingly restoring a vintage vinyl recording. Imagine for a second that your vinyl record is really just music data like any other computer data. To transfer to CD, we start with music data on a vinyl disk. We copy it to our hard drive. Then we remove extraneous stuff from the recording and write this data to a CD. Doesn't this sound as much like editing a spreadsheet as it does bringing analog music to life? Certainly, a recording engineer pays MUCH more attention to the overall sound of the music than to the tools that he uses. His digital recording tools and effects units are there simply to allow him to achieve his artistic vision – not to do rote transfer of one medium to another.

We can hear you saying, "Now wait a minute. My record contains the music in exactly the form that the original artist and recording engineer wanted. The best I can ever do is faithfully record this into my computer and remove the noise. If I do this, I'll end up with something that is true to the artistic vision of the people who made the record."

Sorry. Not true. For the reasons why, let's look at the world of audiophiles.

We have recently been studying audiophile catalogs and web sites. These outlets offer literally hundreds of preamps, cartridges, amplifiers, speakers, and other devices to make your audio sound better. On audiophile newsgroups, you'll find them discussing why a $1500 preamp actually sounds better than a $2000 preamp. There will be arguments as to which cartridge sounds the best – and the same for all the major stereo components. You'll hear discussion that preamp A provides "upper frequencies that are sweet and delicate", "great mid-range punch" and "warmth and liquidity".

This may sound a bit strange, but consider how difficult it is to describe small differences in sound using English words. We can certainly understand the problems our audiophile friends have in describing their audio especially when wine tasters have to resort to something like, "smooth with just a touch of impertinence".

But here's the bottom line – there is no reference standard for what you hear when you simply play a record (or a CD for that matter). Even in the multi-thousand dollar audiophile world, every piece of equipment is different and will produce a different result. This is why audiophiles go to great lengths – and expense – to get a sound that is pleasing to them. For them, the answer is to accumulate a series of components that produces this sound. However, even in the audiophile world, there is no gold standard for what recordings should sound like. Play the same record on 10 different $30,000 stereo systems, and you'll get 10 different sounds. That's just a fact.

So where do these differences come from and why do they exist? There are 4 major answers:

No piece of stereo equipment is "perfect". From turntable to speakers, they will all have their unique sound.

No record is perfect. Even if we assume that the master recording was "perfect" (which we certainly can't), there are differences in the record manufacturing processes, the wear of old records, the noise on them and the time in which they were recorded. Record collectors have for years discussed the merits of various record pressings and record companies often had "high quality" and "regular quality" divisions.

No ears are perfect nor are our tastes the same. Most of us lose some of our high frequency hearing as we age. We may no longer get the full impact of a sizzling cymbal hit even if our stereos are playing it just the way it was intended – which is highly unlikely. Also, all of us have personal preferences as to what sounds good. You can tell when someone likes a lot of bass when they share it with you from their cars when you are 6 lanes away.

No listening environment is perfect. Your listening room with its curtains, carpets and sound reflecting walls will make a difference in what you hear. This is why recording studios are covered in sound absorbing material.

Finally to the point

Ok, we've finally come to the point. It's not 1975 anymore and, for many of us, records are not meant to be played as they used to be. Instead, times have changed and now records are meant to be transferred. And, this process of transferring the music on records to another medium needs to be based on our individual preferences. In effect, we are going to suggest that we start thinking like audiophiles and change our method of restoring our vinyl from one that is akin to copying a computer disk to one that gives us a much better opportunity to create a final result that is just perfect to our ears. After all, this is what audiophiles have been doing for years. It's 2004, and with a new approach, you can now create recordings that are dramatically better. This is The New Way.

So - - - Here's The New Way

What does this mean specifically as regards transferring vinyl to CD or DVD? Here's the overview.

First, we are going to provide a fully flat recording from the record to the computer. We're going to do this with a **phono preamp** that has no built in RIAA equalization; it will have a ruler straight (flat) frequency response.

Next, we are going to use a test record and your Diamond Cut Software to fully "dial in" your system to remove non-linearities in all the components from the cartridge to the sound card. You'll only do this once and we'll then know we are getting the best (and flattest) audio from the record into the computer.

We'll also now suggest we record at high bit widths and sample rates – all the way up to 24 bit 192 khz. Today we have the ability to make audio DVDs at these quality levels, but even standard CDs will greatly benefit from this new approach.

Now we'll provide an almost perfect RIAA EQ to this recorded audio. We'll not use the much more error prone RIAA EQ that is in normal phono preamps, but rather the almost perfect one in DC Six. Then, in another break with the past, we'll use audio enhancement tools in DC Six to adjust the audio so it sounds best on our equipment and to our ears.

Lastly, we'll write this high-resolution audio to CD or DVD-A. We'll end up with something that is much better than we have in the past.

Why a Flat Preamp?

First a question - Do you want to hear the audio EXACTLY as it appears on the record or do you want to hear EXACTLY the audio the original mastering engineer created?

As many of you may know, what is on the record is way yonder different from what the original mastering engineer heard when he mixed the master two track audio tape. In fact, if you were to hear the audio exactly as it is recorded on the record, you may even be shocked. We suspect that most of you have never ever heard the audio exactly as it is on an LP (or 45 rpm or 78 rpm record for that matter).

Let's take an example. Suppose you are the head engineer on Pink Floyd's Dark Side of the Moon, Alan Parsons. You are given a tape with multiple audio tracks that contain all the musicians' performances. You ever so carefully mix the 16 individual tracks of audio together. You make sure the volume of each musician is perfect. You use your years of experience and a wondrous array of technical tools to create just the perfect sound. After hundreds of hours of painstaking work, the band signs off on the master tape and you listen one more time to the final result. "Perfect", you think.

You audience thinks so too. Your album remains in the top 200 selling albums in America for 724 weeks.

It's a classic album that has just an amazing sound.

But that sound is not on your LP. It's nothing like what the original mix-down master tape sounded like. Instead, what is on the record has almost no bass. It's has very shrill high frequencies. It sounds like a poor AM radio on steroids. And we don't have to pick on Pink Floyd, every album in your collection sounds this way.

"Now wait just a doggone minute", I can hear you say. "My albums don't sound like that at all. With my Mitsuwama vintage tube preamp, and my Hectordyne 9000 gravitationally coupled amplifier, I get great sound. These are the finest stereo components ever made. They were hand made by Elves in a clean room without any air in it. They are absolutely transparent and are almost perfect in reproducing the sound on the album".

If you screamed this at your computer screen, please take a break now to get your blood pressure back to safe levels.

Feeling better?

Actually you are half right. You probably do have good stereo equipment, but the sound on your LP has almost no bass and is very shrill and bright in its high frequencies. Bank on it.

This is simply the way records are made. They are made this way on purpose. The bass frequencies are reduced by a whopping 20 db and the high frequencies are increased dramatically BEFORE the master tape is written to a record. The result is just what we have been saying - very little bass and shrill high frequencies. That's what is actually on your records.

Ok, you are starting to believe us. But why in the world would the record companies not record the exact audio from the master tapes right on the records? There are two reasons - one has to do with recording physics and the other has to do with our old friend, noise.

First, let's start with the physics. The amount of bass on a record is a function of the width of the groove. The more low frequencies and the louder the bass, the wider will be the groove. This is the way records work. So, if we have a recording with lots of loud bass, we'll make very wide grooves. The wider the grooves, the more space that is required between them and therefore the fewer that we can put on a record - and the less time we can record on an album side. Heck, a rap record might only last for a minute or two on a record because the grooves would be so wide. This may be a good thing, but it's beside the point. So the record companies reduce the bass simply so that the record grooves are not unnaturally wide. The frequency below which this occurs is referred to the turnover frequency. This is the frequency at which the cutting head reverts from a constant velocity mode of operation to a constant amplitude mode. For the RIAA EQ curve, this value is 500 Hz.

That covers the low amount of bass on records but how about the excess of high frequencies? To understand this, let's cover how a master record is actually made.

Basically, a record mastering machine consists of a cutting head which is similar to your stereo cartridge. The stylus in this cutting head actually has a heat element in it to help make the grooves on the record. This cutting element is connected to an amplifier which drives the cutting head in time with the music. Thus, your music is translated into a series of spiral grooves on the disk.

Since we have an electronic amplifier as an integral part of the record mastering machine, we have broad spectrum noise from the amp which sounds like hiss. This is due to the nature of electronics and can be reduced but not eliminated in the design of the equipment. The record itself has noise as a normal part of the medium due to tiny imperfections in the master record and in the vinyl that you end up owning. There's no way not to have this noise. It'll be there and it'll be audible. So, what should be done?

We can use the fact that the amount of noise generated by the amp and by the record itself is relatively fixed in volume. You'll get some noise but it'll not vary a lot. What if we then simply made the high frequencies in the good audio much louder? This will cause the good audio to be much louder in the high frequencies while the noise added by the mastering equipment and record would be fixed. The result will be a good Signal to Noise Ratio - that is the difference in volume between our good audio and the volume of the noise will be a large number. This process is called Pre-emphasis and is applied during the creation of the lacquer master. The frequency above which this is applied is called the Roll-off frequency. For RIAA this occurs at 2120 Hz.

This is why the high frequencies on records are pumped up a lot. It's simply so our good audio high frequencies are much louder than our noise floor. We'll see how this helps us later.

So now you know why the audio on our records is recorded with very little bass and with lots of high frequencies. The next question has to be, "why do my records sound good when I play them".

The answer is a nice little thing called a Compensation / Equalization Curve.

You see, the problem with wide grooves at low frequencies and with noise in the high frequencies has been known since the beginning of records. And even starting with the electrical recording era, which began in the mid to late 1920s, record companies fixed the problem by reducing the bass. As you might guess, here's the secret - your stereo equipment does exactly the opposite thing when you play a record. It increases the bass and it decreases the highs. Basically, your stereo equipment has a pretty complex tone control inside it to do this job.

The theory is this - if we drop the bass by exactly 20 db when we record, but increase the bass exactly 20 db on playback, we'll get the original sound. The same idea works in the high frequencies; we pump them up when we record and we drop them back to normal when we play back

We are sure you can see that this makes perfect sense. But there is a complication, and it‘s a big one.

To see the problem, let's travel back to the early days of LPs. Don't worry, we won't be there for long. Now let's assume you are going to make a record and you are familiar with the "wide groove" problem at low frequencies. After some experimentation, you decide that dropping the volume of a 30 Hz signal by 20 dB will result in just the right width of the record groove.

That's all well and good, but what about 300 Hz audio? 300 Hz is a higher frequency than 30 Hz, so it won't make as wide a groove on the record to begin with. If we drop it by a whopping 20 dB, we'll end up with relatively narrow grooves. Not a good idea, you think. Again you experiment and find that at 300 Hz you should really only decrease the signal volume by about 2 dB in order to get reasonable groove widths.

Next you turn your attention to the high frequencies. You know that you are going to have noise in your recordings in this frequency range, so you find that at 10,000 Hz you need to increase this frequency by a gigantic 16 db. This makes the good audio at 10 kHz so much louder while, of course, the noise inherent in the recording process is mostly fixed. You realize that on playback, the equipment must now reduce the volume at 10 kHz by 16 db, thus making our good audio sound right while reducing the noise by 16 db. But, like in the low frequencies, you discover that at 2000 Hz the amount of boost while recording, and cut while playing, needs to be only 2 db.

In fact, it's even messier than this. As you continue to experiment, you find that each frequency in the audio spectrum needs to be adjusted individually either up or down in volume in order to achieve reasonably sized groove widths along with low noise. How in the name of Alexander Graham Bell can we do this?

The solution is a compensation curve. This is a playback curve created in DSP software, so the low frequencies are boosted while the highs are attenuated. This curve is exactly the opposite of the one used to encode an LP.

This curve works to create good sounding records, but it's not the only possible one that could be used. In the early days, there were many different curves which meant that records from one company didn't sound exactly as intended when played on another company's equipment. This was especially true with 78 rpm records (and early mono vinyl from the 1950s), which used a variety of compensation curves. This is also why some vintage stereo equipment had switches to choose which curve to use on playback. There's even some very high dollar equipment made today which has this feature. Audiophiles pay this price since they want the best possible reproduction.

When the LP became popular, a standard EQ curve was proposed by the RIAA and was adopted by all manufacturers. This is now known as the RIAA compensation curve and it is implemented in just about every piece of audio equipment which as a jack labeled "phono". If you plug your turntable into a jack labeled "phono", you can be just about 100% certain that there is a circuit inside which attempts to apply an RIAA compensation curve to the audio that comes off your records.

So, that's the history. Time to return to the 21st century. We hope you enjoyed your trip.

What the??

"OK, what's the point?" "I already have a RIAA preamp built into my stereo. I've got a decent turntable and decent stereo system. Seems like I must be hearing what the original recording engineer intended." If Alan Parson dropped by this afternoon, he could listen to your copy of Dark Side of the Moon and he'd hear just what he heard when he made the master tape. Right?

Wrong.

You see, what you actually hear when you play a record depends on a LOT of things. In order to hear EXACTLY what Alan heard on that day he made the mix-down master, you'd have to have the exact same equipment that he did and you'd really have to listen to it at the same spot he did. You see, no piece of equipment is perfect – they all fail at least in some small way to do their job without flaw. This shouldn't be surprising. Your turntable, amp and speakers play a part in presenting the music and these are not perfect devices either. But let's focus on the preamp for a minute.

Yes, you have a preamp even if you don't have a separate box labeled Preamp. If you have an input labeled "phono" (or mag phono), there's a preamp right behind it. If you are playing records, you have one.

Inside your preamp is an electronic circuit that provides the RIAA EQ curve we have been discussing. Ideally, it should provide exactly the curve we see in the illustration above. However, it doesn't. Like all man-made audio equipment, it falls short here or there. If you could plot the actual curve from your own preamp, you'll likely find that in some places yours is above our perfect curve, and in others, it's below it. Ok, that doesn't sound too bad until you realize that even a 3 dB error above the curve will cause the audio to sound louder at these frequencies than it should.

The preamp you use will make a lot of difference in the sound of the resulting audio. Why is this? While there are a lot of things that can make one piece of equipment sound different from another, one of the areas of greatest importance is the RIAA EQ circuit. You see, these circuits are made up of electronic components – typically resistors, capacitors, and op amps, tubes or transistors. These components themselves are less than perfect and each one adds its own imperfections to the resulting audio.

Here is a list of the types of problems that will occur to some greater or lesser degree with any RIAA analog preamp: (note: this list is a bit technical. If it's Greek to you, just skim over it. The point is that analog RIAA circuits do fail to accurately reproduce the original audio by some amount.)

Analog EQ's produce fairly appreciable levels of Harmonic and Intermodulation distortion because of the nonlinearities of the op amps used.

Analog EQ's produce Noise because of the op amps used.
Analog EQ's have sloppy frequency and phase calibration because of resistor and capacitor tolerances (and time drift and temperature impacts on their value).

The left and right channels track poorly because of component variations in terms of resistor and capacitor variance (tolerance).

Aging has a substantial effect on capacitors, so what you hear today will be different than what you hear a year or two from now from an analog system.

Analog circuits pick up some hum because of the physical loop areas which can't be avoided in the circuit layout.

Analog circuits display crosstalk due to stray capacitance which can't be eliminated between the channels.

Analog circuits are somewhat microphonic picking up low levels of room sound or feedback. Try this, - - - turn the gain up on your system and then tap on the EQ. Hmmmm. Like that sound?

Analog components such as resistors, capacitors and transistors (which are the amplification devices found inside the Op Amps) exhibit a temperature dependency referred to as tempco (temperature co-efficient). Therefore, the performance of an analog circuit (gain and break point frequencies) will change as a function of ambient temperature.

Physical capacitors exhibit such anomalies as DA (Dielectric Absorption), ESR (Effective Series Resistance) ESL (Effective Series Inductance), Voltage dependent capacitance or incremental capacitance (dC/dV) (which creates non-linear capacitance vs. signal level), and leakage resistance. All of these component parasitic effects cause them to behave in a less than an ideal manner when used in an analog circuit.

Note: In the above analysis, the words "Discrete Transistors" or "Vacuum Tubes" can be substituted for "Op Amps" which will only make the problems mentioned even worse than the Op Amp case.

We know some of you are now ready to argue that, while your preamp isn't perfect, it still ALMOST dead on. Maybe that is true, but if so, why do reviewers of preamps always stress what it does to the sound? Here are some examples of real reviews:

Very good detail, image focus, very dynamic, very good clarity (improved clarity, attack, focus).
Midrange was the best part which was smooth and warm.
Sweeter and with any violin playing it's simply sonic heaven

These reviewers are right. Preamps do sound different from each other and that difference is, at least in part, caused by various approaches to try and implement an RIAA EQ curve along with the problems we talked about above. Remember, these are comparisons of high end, expensive preamps. Even here, it's obvious that they sound different from each other. And if we assume we want to get the audio the way it was meant to be, they can't all be right.

So what to do? One approach would be run out and buy a bunch of expensive preamps. You can then whittle away some happy hours as you swap them in and out to see which one sounds best. We're not likely to suggest that many folks follow this path.

Another approach is a bit more radical. What if we built a preamp without any RIAA compensation in it at all? We'd not then have any problems with the EQ circuit since it simply wouldn't be there. It would be a "Flat Preamp". How's that for a solution?

Well that might work, but wouldn't we end up hearing what is REALLY on the record? Wouldn't we hear the audio with almost no bass and highs that are way too loud? Wouldn't we have to apply an RIAA EQ somewhere along the line?

The answer is yes, we would, but wouldn't it be great to apply an RIAA EQ curve that was just ever so slightly shy of perfect? You either already have one or can get one for much less than a high end preamp. It's the RIAA EQ choice in the Virtual Phono Preamp (VPP). This RIAA EQ curve is amazingly accurate to the limits of the math we perform. Even the best hardware based RIAA EQ will miss the perfect curve here and there. A great ones would be off by perhaps as little as a few dB while most will be off by as much as 9 dB at places along the curve. There is simply nothing out there that is more accurate than the use of a flat preamplifier along with the Virtual Phono Preamplifier found in Diamond Cut Software.

One More Thing

So, if we apply a precise RIAA EQ curve, we are getting better audio. But that's not all. Your phono cartridge also has its own curve which is, you guessed it, not perfect either. A perfect cartridge would output a fully flat frequency response across all the audio frequencies. You don't have a perfect cartridge, so we're going to suggest you also fix it.

The way we are going to do that is by using a **test record**. We are going to play a pink noise signal into our flat preamp and we are going to record it into the computer. Next, we look at the signal with our spectrum analyzer in your DC software and adjust the signal until it is what it should be. Since we have a known and accurate signal on the record, we can use this to fully flatten out our cartridge, preamp and even sound card input.

When you play the pink noise track you have recorded, you need to create a Multifilter that includes filters – one for RIAA (the VPP or Virtual Phono Preamp) and another to compensate for the problems in your system (the paragraphic EQ).

Now, you just leave the VPP alone and adjust the paragraphic EQ for the best system flatness using your Diamond Cut Spectrum Analyzer. You'll only have to do this once as it won't change unless you change some piece of hardware.

So why is the line sloping downwards and to the right? This is because pink noise should drop by 3dB per octave. Trust us, this is the way it is (constant power spectral desity).

To adjust your own system, use the Spectrum Analyzer set just the way we have it here and duplicate this exact line as closely as possible with the 2nd filter you have in your Multifilter. Note that the Spectrum Analyzer is set to Averaging mode – this means you'll have to be patient after you make an adjustment to let the instrument average to it's real readings. In practice, you make a small adjustment, click on clear in the spectrum analyzer to clear the display and then wait for a minute or so as the line to re-settle.

That's all there is to it.

For information on Flat Preamps, go to our website and look up the CTP-xxxx series.