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 45rpm or 78 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 mixdown 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 20db 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 RIAA, this value is 500hz.

That covers the low amount of bass on records, 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 hearable. What to do?

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 2120hz.

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 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 20db when we record, but increase the bass exactly 20db 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.

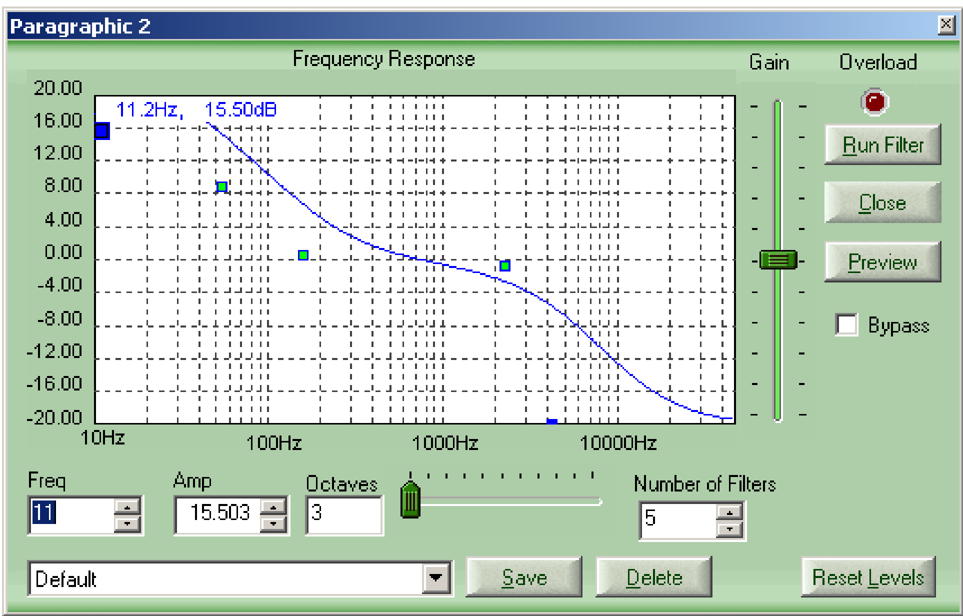
To see the problem, let's travel back to the early days of LPs. Don't worry, we won't be there for long. Now let's assume you are going to make a record and you are familiar with the "wide groove" problem at low frequencies. After some experimentation, you decide that dropping the volume of a 30hz signal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

That's all well and good, but what about 300hz audio? 300hz is a higher frequency than 30hz, so it won't make as wide a groove on the record to begin with. If we drop it by a whopping 20db, we'll end up with relatively narrow grooves. Not a good idea, you think. Again you experiment and find that at 300hz you should really only decrease the signal volume by about 2db 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 10,000hz you need to increase this frequency by a gigantic 16db. This makes the good audio at 10khz 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 10khz by 16db, thus making our good audio sound right while reducing the noise by 16db. But, like in the low frequencies, you discover that at 2000hz the amount of boost while recording, and cut while playing, needs to be only 2db.

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. Here's a picture of the RIAA curve right out of DC Six.



Here's a screen shot from DC Six. Notice how the equalization is represented by a curved line. This is a playback curve, so the low frequencies are boosted while the highs are attenuated. This curve is exactly the opposite of the one used to make an LP. Notice how the amount of boost or cut changes with frequency in a non-linear manner.

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 mixdown 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”, 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 3db 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.)

1. Analog EQ's produce fairly appreciable levels of Harmonic and Intermodulation distortion because of the nonlinearities of the op amps used.
2. Analog EQ's produce Noise because of the op amps used.
3. Analog EQ's have sloppy frequency and phase calibration because of resistor and capacitor tolerances.
4. The left and right channels track poorly because of component variations in terms of tolerance.
5. 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.
6. Analog circuits pick up some hum because of the physical loop areas which can not be avoided in the circuit layout.
7. Analog circuits display crosstalk due to stray capacitance which can not be eliminated between the channels.
8. 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?
9. 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.
10. 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 parasitics cause them to behave in a less than 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.

This is kind of like looking at your saliva under a microscope and seeing lots of things swimming around, isn't it? Yuck.

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. 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?

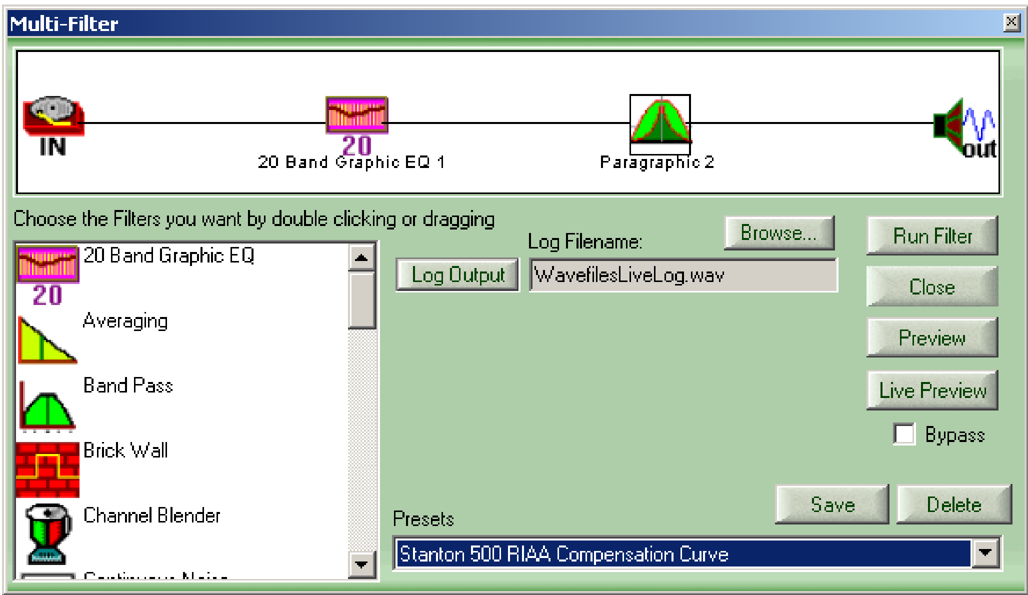
The answer is yes, we would, but wouldn't it be great to apply an RIAA EQ curve that was just ever so slightly shy of perfect? You either already have one or can get one for much less than a high end preamp. It's the RIAA preset in the Paragraphic EQ in DC Six. This RIAA EQ curve is amazingly accurate 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 one would be off by perhaps as little as a few dB while most will be off by as much as 9db at places along the curve. There is simply nothing out there that is more accurate.

One More Thing

So, if we apply a great RIAA eq curve, we are getting better audio. But that's not all. Your phono cartridge also has it's 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.

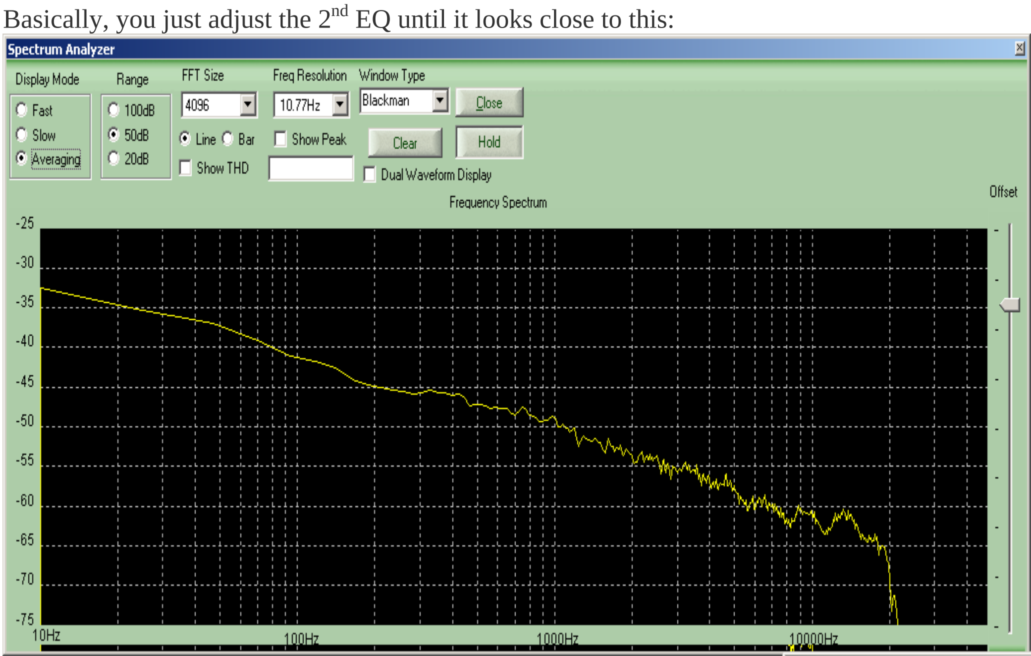
The way we are going to do that is by using our test record. We are going to play a pink noise signal into our flat preamp and we are going to record it into the computer. Next, we look at the signal with our spectrum analyzer in DC Six and adjust the signal until it is what it should be. Since we have 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 TWO Eqs – one for RIAA and another to compensate for the problems in your system. Your multifilter will look something like this:



Now, you just leave the RIAA one alone and adjust the other one for full system flatness. You'll only have to do this once as it won't change unless you change some piece of hardware. Notice we used the 20 band EQ to flatten out our system, but you could use a second Paragraphic if you want.

Basically, you just adjust the 2nd EQ until it looks close to this:



So why is the line sloping to down and to the right? This is because pink noise should drop by 3db per octave. Trust us, this is the way it is.

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 EQ 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 make it reset and then wait for a minute or so as the line settles.

That's all there is to it.