Selected Technical Issues in Phono Preamp Design

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The ultimate test of an audio component is its sound, and the final arbiter is the listener. But good sound does not arise by luck or accident. The designer must understand the elements necessary for good sound and have the skill to achieve them. This paper discusses selected issues that arise in the design of phono preamplifiers.

Such an amplifier has two basic functions. First, it must amplify to usable levels the tiny signal that the cartridge generates as it traces the wiggles in the vinyl record. It is no easy task to amplify those minute voltages as much as 1,000 times without also amplifying the surrounding noise to objectionable levels.

The second task that falls to a phono preamp is to compensate for the RIAA curve that the engineers who make vinyl records employ. That curve reduces the level of the low frequencies and boosts the highs of the musical signals that are embedded in the grooves on the discs. This distortion of the musical signal is done to save space on the record and to make it easier for the tone arm and cartridge to track the grooves. But, if one wants listenable music, the effects of this process must be undone and a neutral balance restored to the signal before it reaches the speakers.

Neither of these functions is easy to accomplish well, but both can be achieved by a phono amp that excels in seven elements.

1. **Gain** refers to the amount of amplification a device provides. It can be expressed in terms of how much the amp multiplies the input voltage (an amp that multiplies the input by 50 would be said to have an amplification factor of 50) or on a logarithmic scale in decibels (the amp which multiplies the input voltage by 50 has a gain of 34 dB). The typical 5 mV output of a moving magnet cartridge will need to be amplified by a factor of roughly 100 (40 dB); the much smaller 500 uV signal from a moving coil cartridge, by approximately 1000 times (60 dB).[1] Many cartridges have outputs falling between these extremes. Thus, a good phono stage ought to have gain that is adjustable in several steps from 40 dB to 60 dB. It is relatively easy to design circuits with the needed gain, but doing so while keeping noise and

[1] Those cartridge output levels and gain factors refer to the signal at the midpoint of the audio spectrum (roughly 1K Hz). In specifying the gain of a phono preamp, it customary to state the gain at a single frequency, usually 1 KHz. Often no frequency is specified when stating the gain, it being commonly understood that 1 KHz is meant. But, in fact, the signal level and gain vary across the audio spectrum in accordance with the RIAA curve built into LPs. That curve reduces the lowest frequencies in the musical signal by about a factor of 10 (20 dB) relative to those at 1KHz and boosts the high end by about the same amount. The very lowest frequencies (about 20Hz) must, therefore, be amplified 1000 times for a typical moving magnet cartridge and a whopping 10,000 times for a moving coil pickup.

distortion low and providing adequate headroom can be quite challenging and involves critical trade-offs. Ways to meet these challenges are discussed below.

[Tip for users of phono amps with adjustable gain: *Keep the gain knob on your line level preamp turned up at least to 1 o'clock and use the lowest gain setting on your phono amp that gives you the sound level you want from your speakers. This way you avoid the common mistake of amplifying the signal, then attenuating it, and finally amplifying it again, a process that will surely increase distortion and likely add noise.*]

2. Low noise. The amplification factors discussed above are large and the signals from the cartridges are small. This combination invites excessive noise. If a phono amp is not very quiet, it can bury the musical signal in noise. So, how to make such a quiet amplifier? Feedback, which can be used to solve many design issues as will be discussed below, is ineffective to reduce noise. Consequently, careful attention to the following elements is critical:

a. **Extrinsic noise** exists all around us. It consists of the low level electro-magnetic waves given off by radio stations, appliances, computers, power lines and, transformers, etc., in short all the stuff of modern life. This electro-magnetic radiation can all get into audio cables and equipment where it creates noise that can be amplified to destroy our enjoyment of music. To avoid this result and keep these signals out of the audio chain, a phono preamp should have the following characteristics:

Excellent shielding. All cables running into and out of the amp should be shielded and those shields should be properly grounded. The circuits inside the amp should have effective ground planes in close proximity to the signal circuitry. All transformers should be kept well away from the low level signal circuitry.

Balanced circuitry with differential amplification. Even with the best of shielding, some noise creeps into the input of an amp, riding on the wires that bring in the signal from the cartridge. An ideal differential amplifier has two inputs and amplifies only the *difference* between voltages on the two inputs. When noise creeps into a balanced cable (one that uses two wires to carry opposite phases of the same signal) it usually appears at about the same voltage on both inputs to the amplifier. Because the differential amp only amplifies the difference, it ignores (and effectively cancels) the noise that is common to both inputs. Thus for lowest noise, a phono stage should have balanced inputs feeding a differential amp.

Compact circuitry. The shorter the connections among components in an enclosure or on a circuit board, the less opportunity exists for those wires or traces to pick up noise from each other or from other sources within the enclosure. Thus, components should be kept close together and wires and traces kept short. Using circuit boards with several layers for ground and power distribution planes can greatly reduce the length of interconnecting traces and dramatically attenuate noise.

b. Proper component selection. Virtually all components in an audio circuit add their own noise to the signal. Even if we were able to keep most extrinsic noise out, our circuits themselves will contribute at least a little hiss to what we hear from our speakers. But, through careful selection of both passive and active parts, it is possible to hold noise down to very low levels. For example:

All **resistors** generate an irreducible noise voltage that depends exclusively on the value of the resistor: the greater the value the greater the noise voltage. Thus, noise can be lowered by adjusting a design to use low value resistors. Resistors also contribute an additional noise component based on their construction. Wire wound and metal film resistors are, for example, lower in added noise than carbon composition resistors.

Active devices, transistors, op amps, etc., vary greatly in their noise performance. Good design requires selection of devices that minimize noise in the application for which they are used.

c. Circuit design. Choosing low noise parts is only the first step. Those parts must be used in circuits that allow them to exhibit their noise minimizing characteristics to best advantage. For example, certain types of discrete components (e.g., transistors) can exhibit very low noise, but only if used in a circuit that keeps them in their optimum operating range in terms of current and voltage. If pushed to operate outside that range, their noise contribution increases. In addition, low noise transistors can be made even quieter through various circuit design techniques, for instance by using several of them in parallel at critical points in the circuit. Op Amps too can give very low noise performance, but only if used correctly. The Analog Devices AD797, is a popular, very low noise device, but it will give its best noise performance only if it is fed signal from a low impedance source.

d. Power supply filtering. All amplifying devices use a small signal (low voltage and current) to control a larger current. The small signal is the content (music, voice, etc.) we are trying to amplify. The large current is provided by the power supply. Any ripples, noise or other variations in the current from the power supply are going to find their way at varying levels into the amplified signal as hum and noise. The ideal power supply feeds the amplifier a smooth, steady current with no disturbances.

But the typical, real-world power supply for solid state circuitry takes electricity from the overhead power lines, uses a transformer to step the voltage down to useable levels, and changes the alternating current that the power company sells into direct current using rectifier diodes. This process is a potential source of all kinds of hum, noise and ripple that will degrade the music. The outside power lines make good antennae to gather in radio frequency interference. The transformer is a marvelous source of 60 cycle hum; and the diodes themselves generate interference when they turn on and off in the rectifying process. All of these sources of unwanted electrical "junk" have to be addressed. The RFI should be filtered out of the power lines before it can get into the low level phono circuitry. Diode noise can be addressed both by using very fast, "soft recovery" diodes and by filtering the output of those diodes with film capacitors. The 60 cycle ripple and other noise is filtered by a combination of passive RC (resistor/capacitor) filters augmented by capacitance multiplier circuits and active series and shunt voltage regulator circuits. The challenge for the designer is to achieve the required filtering without using up most of the budget and much of the room inside the amp's enclosure with many large, expensive capacitors.

3. RIAA equalization. As mentioned, low frequency sound is attenuated and high frequencies are boosted in making LP records. These days, virtually all manufacturers use the RIAA curve for this purpose. 1K Hz is the midpoint of the curve, where by convention the level of the signal is said to be neither increased nor decreased. The signal gain at 1K Hz is thus the "zero" level against which the attenuation and boost called for in the rest of the RIAA curve are measured. The gain of the lowest frequencies (20 Hz) is attenuated about 20 dB relative to the 1K Hz zero level, and the gain of the highest (20,000 Hz)is increased by about the same amount. 20 decibels represents a ten-fold increase (or decrease) in signal amplitude. If the signal thus altered were amplified as is, the music would be destroyed. So, the effect of the RIAA curve must be negated and the original balance among the frequencies restored. This can be done with a specially designed network of capacitors and resistors. Such a network must be very accurate, that is it must apply just the right amount of correction to the signal coming from the cartridge to restore a flat frequency response. The network ought to use low distortion, low noise components such as metal film resistors and polypropylene capacitors. Perfect accuracy is of course, impossible in the physical world. But accuracy to + or – a few tenths of a dB is necessary. To preserve good imaging the two channels of a stereo amplifier need to track each other very closely. To achieve the requisite accuracy, the designer must use capacitors and resistors with very tight tolerances (e.g., 1% or better).

The RIAA equalization can be applied either "actively" or "passively." It is said to be active when the resistor-capacitor network is included within a feedback loop inside the amplifier. It is passive when it is placed between two gain stages and does not involve feedback. The active approach has the advantage of requiring fewer component parts and can therefore be cheaper. The passive circuit has a small advantage in accuracy and avoids the need for feedback.

4. **Head room** refers to the amount by which a signal can be amplified before clipping occurs. One might imagine the amplifier as having a tunnel through which the signal travels from the amp's input to its output. The tunnel's diameter is more than adequate compared to the small size of the signal present at the input. But as the signal moves through the amp and is amplified, it may grow too large for the tunnel and bump into the tunnel walls. When that happens, hard clipping occurs. The tops of signal waves are cut flat, and the speaker makes a harsh, unpleasant sound. Just before hard clipping, as the tops of the signal waves approach the tunnel walls, the waves are compressed causing a dramatic increase in distortion and loss of the dynamics of the music.

Ironically, even though the input signals from the cartridge are very small, phono amps may lack ample headroom. The possibility that the musical signal will clip is a real one because phono amps greatly amplify the level of the incoming signals and because the use of RIAA equalization means the signal comes off the LP record already boosted 20 dB in the high frequencies. The problem is particularly acute in the first amplification stage. A good design will include adequate of headroom to avoid these problems and to reduce the extent to which tics and pops caused by dust or scratches on the record interfere with the musical experience. These annoying artifacts often have much higher voltages than do the musical signals. The nastiness of their sound is magnified if the electrical signals they generate hit those tunnel walls and clip. They become much less annoying and distracting if the amplifier has enough headroom to reproduce them without clipping.

A good measure of an amp's headroom to avoid clipping on loud musical signals and record pops is its "overload margin", the amount of room it has between the average signal level and the point at which clipping occurs.[2] The greater the room, the louder the transients it can handle without clipping. Overload margin is usually measured in decibels with reference to a particular input level and gain setting. For example, a phono amp might be said to have an overload margin of 20dB at 5mV input with a gain setting of 40dB.

Overload margin can be increased by raising the voltage levels of the power supply rails and adjusting the amplifying circuitry commensurately. But even with high supply rail voltages, the first stage of a phono preamp can be susceptible to overload because that stage amplifies the signal before the boosted high frequencies from the LP record are attenuated by the RIAA network. When we say the average signal level from a typical moving magnet cartridge is 5mV, we are actually speaking of the average level at 1 KHz, the midpoint of the RIAA curve. The signal coming off the LP record at 20 KHz (the high end of the audio spectrum) is actually 50mV, which is 20dB or 10 times greater than the 5mV we usually talk about. If that 50 mV signal is amplified 100 times (40dB) in the first stage it will rise to a level of 5 volts. And, remember that 5 volt level is only an average. Transient peaks can easily be 2 or 3 or more times that average. Even a circuit with up to 15 volts headroom in the first stage may not be sufficient to avoid clipping on signal peaks if that first stage has as much as 40 dB of gain. Because there are practical limits to how high the power supply voltages can be raised, it is incumbent on the designer to set the first stage gain to a level low enough to provide adequate overload margin. [3]

5. **Low Distortion.** One might think of distortion as a bending or deforming of the shape of the electrical signal, be it a simple sine wave or the very complex waves that carry music. The amount of distortion is directly proportional to the amount of deformation. In audio applications it is not, of course, desirable to alter the signal we are amplify; doing so will surely take us further from the sound of the original performance. Thus, low distortion is a characteristic to be sought in phono amp design.

But distortion measurements may not be particularly good guides to excellent sound. Most distortion measurements, including the commonly reported ones for total harmonic distortion (THD) and inter modulation distortion (IMD) , are made using sine waves as the test signal. The problem is that musical signals are far more complex than these sine waves. Music presents a much greater challenge to the amplifier than do the test signals. It is safe to say that an amplifier that shows high distortion in a sine-wave in a distortion test will also distort the more complex musical signal badly. But, the opposite is not necessarily true. An amplifier that has good test results may still distort the more challenging musical signal.

A wise designer will seek low measured distortion but will recognize that is no guarantee of great sound. Negative loop feedback is one often used route to produce very low measured distortion. Section 7 below discusses some of the issues pertinent to feedback. If the designer decides to eschew feedback, it is still possible to achieve quite low measured distortion. All transistors have a range of

[2] One useful definition of clipping in audio amplifiers is the signal level where total harmonic distortion reaches 1%.
[3] This is one of the many places where a balance must be struck. To reduce noise, it is desirable to have as much gain as possible in the first stage. But too much will reduce headroom to inadequate levels.

voltage and a range of current within which they can operate. Within these ranges are smaller ranges where the transistor operates close to linearly. Operation within these nearly linear ranges produces the least distortion. So the designer's first task is to arrange the circuitry so that the voltage across the transistor and the current through it are such that the device operates in its most nearly linear range. This process is referred to as "biasing" the transistor. Distortion is further reduced if the circuit is designed to hold the bias voltage and current stable even as the signal input and output change. One way to hold the current stable is to use active current sources to set the bias current. A current source is a sub-circuit that keeps the bias current at a fixed level even as voltages change. Similarly, a cascode arrangement will hold the bias voltage constant. Techniques such as these, together with selection of low distortion devices will produce good measured distortion figures, easily to 0.02% THD or less in a phono preamp.

6. **Cartridge loading** refers to the amount of resistance which the signal from the cartridge "sees" at the input of the phono preamplifier. For moving magnet cartridges, 47 K Ohms (47,000 ohms) has become the standard and works well. But moving coil cartridges present a different situation. For them, the amount of load resistance affects the amplitude of the musical signals at the high frequency end of the audio spectrum. In effect, it tunes the signal from the cartridge. The optimum resistance varies from cartridge to cartridge, system to system and listener to listener. To be suitable for use with a variety of cartridges and systems, a phono preamp should have a 47 K Ohm load for moving magnet cartridges and be configured to allow the load resistance to be adjusted in several steps from 100 to 1000 ohms for the moving coil cartridges.

7. **Simple discrete circuitry that does not rely on feedback.** Here we enter the subjective arena. Very good measured results can be obtained using negative feedback. It controls gain precisely, can produce astonishingly low levels of total harmonic distortion (THD), and lowers output impedance. Op Amps are, by their very nature, feedback devices; good designers can achieve some very good sound using them.

But, use of negative feedback always reduces the gain of an amplifier. The more feedback, the more gain is reduced. In effect the price of using feedback is loss of gain. If feedback is to be used, the amplifier must generate a great deal of "excess gain" so that it can afford to give up some to the feedback. Creating this excess is not difficult; the designer simply adds additional gain stages to the amplifier until a sufficient excess is created. Herein lies the problem, however. Each of those stages necessarily adds distortion and noise. The feedback can greatly reduce THD. But, as noted above, low measured THD does not guarantee good sound. Whether or not this distortion cancelling effect of feedback well serves the music remains a controversial topic in both the engineering and audiophile communities.

Another cause of concern regarding feedback in audio amplifiers arises from the high frequency roll off present in every amplification stage. Each such stage adds its own roll off. The cumulative effect can be a phase shift in the high frequency output of the amplifier that is sufficient to cause oscillation and instability. Such problems can be addressed by good engineering techniques, and the engineers who advocate feedback argue that they are fully capable of doing so.

We do not propose here to resolve these long standing disputes. We do, however, believe that the very best sound is achieved in designs that avoid feedback and keep circuits as simple as feasible. The use of negative feedback requires one to use a relatively complicated, multi-stage design that necessarily creates a distorted signal and then depends on the subsequent action of the feedback loop to "repair" that signal. Doing so risks creating a circuit that cannot keep up with the musical signal because of its complexity and the time delay, however miniscule, built into its "distort and repair" topology. Moreover, it is aesthetically unsatisfying to design a circuit that will first generate a defective signal and then repair it. We prefer to build it right the first time, with low distortion as described above. Our listening supports this choice.

Conclusion

The above discusses the first half of the process of designing a good preamp. Once application of engineering principles and methods has achieved good performance in each of the elements described, the second stage of the design process begins: repeated cycles of listening, modifying and more listening. The end result of the effort is the sound of music played through the amplifier being designed. No measurements will identify the best balance of attributes to create sound that will be realistic to human ears. Only the ears of careful listeners can do that.