

## **Mic Preamps – What You Need To Know**

Mike Rivers

Surely you're no stranger to the term "mic preamp." Perhaps in an on-line forum you've been told that you need one, or you need a better one, or you're in search of a silver bullet that will make your vocals as smooth as Tony Bennett but as punchy as James Brown. Exactly what is a mic preamp, and what's good, bad, and ugly about them?

Back in "the day" (about 35 years ago now!) mic preamps were taken for granted. Every studio had a console and every console had mic preamps which contributed significantly to a particular console's sound. That was then, but in a contemporary studio based around a computer (DAW) there is often no console at all. We still record with microphones, however, so the need for mic preamps hasn't gone away.

In October 1987, the "New Products for 1988" listing in one of the popular trade magazines listed only three stand-alone mic preamps, two from Focusrite (actually the same circuitry packaged two different ways) and one from Innovative Audio Systems, the original name for Demeter Amplification. It's interesting that both companies are still making mic preamps today, but many others have come and gone over the past 25 years. Today we have a huge choice, stand-alone mic preamps ranging from inexpensive but functional to boutique, single-channel and multi-channel boxes of various electronic and sonic flavors, preamps with built-in A/D converters, and, yes, still, mixers with mic inputs.

### **So Why Do I Need a Mic Preamp?**

A microphone converts sound pressure to voltage, but it's a very small voltage - just a few millivolts for a normal conversational voice up to around 100 millivolts when close to a loud voice or instrument. We refer to this voltage range of voltage as "mic level." Most recording devices without a dedicated microphone input - tape decks, professional sound cards, or stand-alone analog-to-digital (A/D) converters - work with voltages in the range which we call "line level," with +4 dBu (about 1.23 volts) being the most common nominal operating level for today's audio gear. A microphone's low voltage must be boosted in order to work with line level devices, and that's what a mic preamp does. The "pre" comes from the fact that it's an amplifier that goes before another amplifier.

Building a good microphone preamplifier is a significant design challenge. It must provide gain ranging from 0 to as much as 75 dB while adding a minimal amount of noise to the low level microphone signal, rejecting electrical noise induced in

the cables, and cleanly amplifying anything from pounding drums to subtle strings without distortion or unwanted coloration.

A computer's built-in sound card or portable recorder usually has a mic level input, but let's set those aside as they're rarely designed to work with studio microphones. Many professional computer audio interfaces (a six dollar word for a sound card costing more than six dollars) have built-in mic preamps, as do stand-alone audio workstations such as those from TASCAM or Akai. Many built-in mic preamps sound fine, but others, while functional, leave much to be desired. Lackluster preamps in budget priced gear sent quality-conscious engineers in search of alternatives, and a burgeoning branch of the audio equipment industry, the outboard mic preamp, was born.

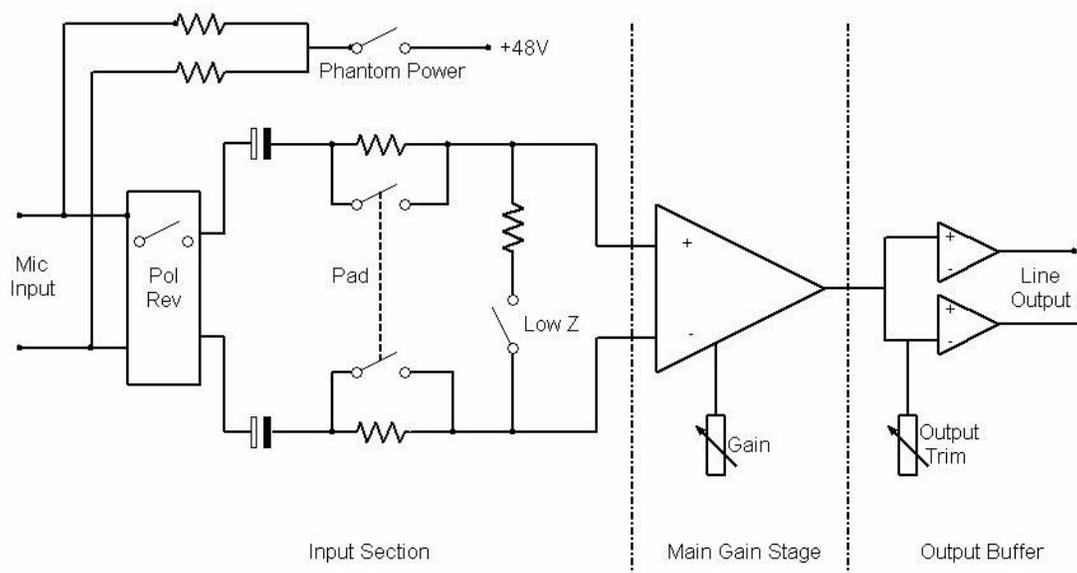
When we use the term "preamp" today, most often we're talking not simply about the amplification function, but about the box, or the product. Preamps come in many shapes and sizes with an assortment of features. Stand-alone mic preamps can be a single channel, dual channel, or multi-channel, with 8 preamps in a box being a common configuration as the front end of a DAW. Most have a pretty basic set of controls with nothing fancy.

We'll talk about variations later, but first we'll take a look at what's inside a basic preamp from a functional standpoint.

## **The Grand Tour**

All preamps have an input and an output for each channel. The input is almost always via a three-pin XLR type connector, the industry standard for studio microphones. It might be located on the front panel, the rear panel, or both. The analog output of a stand-alone preamp is typically on a male XLR connector, a 1/4" phone jack (usually balanced, but not always), or both. Some multi-channel preamps have a single multi-pin connector such as a DB-25 for all the outputs to save space and cost (but don't forget to include a suitable multi-pair "snake" cable in your budget).

Here's a functional diagram of a basic mic preamp, showing the most common switches and controls. Let's start our tour in the middle and work outward:



The main gain stage is what does the real work. I've shown it here as a single operational amplifier (op-amp) but most practical preamp designs utilize several op-amps, a combination of discrete transistors and op-amps, all discrete transistors, vacuum tubes, or any combination of those components. A typical mic preamp provides 45 to 65 dB of gain, either continuously adjustable or switched in discrete steps. Some people prefer continuous adjustment because it lets you fine tune the gain, others prefer discrete steps which allow you to set the gain of multiple channels identically for a stereo or surround mic setup.

The output buffer section provides a low impedance balanced (usually differential) output and may also offer some additional gain. More important, the Output Trim control, if present, allows you tweak the preamp output to match the input sensitivity of next device in the chain. Since many sound cards have no hardware input level control, you can use the preamp's Output Trim to adjust the recording level without compromising the optimum gain setting up front. Use this control to fix "my preamp is too hot and overloads my sound card" or "I can't record hot enough" problems. Don't dismiss a preamp because it doesn't have an Output Trim, but it's a handy feature.

Moving the input section, you'll frequently find a polarity switch, usually inaccurately labeled "phase," but everyone knows what that means. This is nearly always a mechanical switch that simply swaps the wires coming from pins 2 and 3 of the XLR input connector. The polarity switch is usually the first thing tossed when a manufacturer is trying to cut costs, or if the preamp is tightly integrated with a computer interface.

Why is a polarity switch important? It's good practice to maintain acoustic polarity throughout the system – when something pushes air toward the microphone, you

want the loudspeaker to push air toward you. If electrical polarity is reversed somewhere along the line, the speaker cone will move in the opposite direction, pulling air away from you. Sometimes this makes things sound different, sometimes it doesn't, but it's easy enough to get it right. If a sound source is picked up by more than one mic, flipping the polarity of one of the mics can sometimes fix a thin sound resulting from phase cancellation of certain frequencies. This is a "listening" control. Try it if you have it, don't worry if you don't. You can always buy or make an in-line polarity reversing adapter.

Today's preamps typically provide phantom power for condenser microphones.

We used to be able to say that phantom power was only for condenser mics, but today there are a few ribbon mics that have a built in pre-preamp that operates from phantom power. This provides a little extra gain for the typically low output of the ribbon, and also provides a known load impedance which the mic manufacturer can use to fine-tune the ribbon damping.

There's almost always a switch to turn it on or off, though some preamps or mixers have a single phantom power switch for all or a group of inputs. There's a lot of 'Netlore about phantom power damaging ribbon mics, but the truth is that if the mic and cable are wired properly, a dynamic or ribbon mic will not see the phantom voltage. However, a mic cable with a short between pin 1 and one of the other two pins will put the full phantom voltage across the mic element, which may cause harm. A shorted cable will also keep a mic that requires phantom power from working, but that's a more obvious problem. It's good policy to connect mics with the phantom power switched off. If you forget, a loud pop will remind you for the next time. Individual phantom power switches are nice. If you don't have them, just be careful.

Phantom power is usually 48 volts  $\pm 10\%$ , but 24v and 12v are also included in the official standard, and some preamps use non-standard voltages. Be aware that some preamps offering phantom power provide less than 48v. A few low cost mixers borrow voltage for phantom powering from the mixer's +15v op-amp power supply. Nearly all phantom powered mics are happy with 48v, but many won't work with a lower voltage.

If our preamp had an input transformer, it would come next in the signal chain. This diagram shows capacitors, and alternate way to isolate the preamp's gain stage from the DC phantom voltage. Transformers are like a religion, faithfully embraced by some, rejected by others. They always add some coloration, but sometimes that "transformer sound" is desirable. A well designed transformerless preamp sounds very transparent, but often gets labeled "sterile" because of this. Take your pick. Note that some mics sound better working into a transformer, others sound better without one. It's a preference, but it affects the bottom line, too. Good transformers are expensive, and cheap transformers rarely sound good.

The pad is a switchable attenuator, usually 10 to 20 dB. Mic preamp design is tricky because it has to deal with a wide input dynamic range, from crickets chirping on a quiet night to a roaring guitar amplifier, and do so without excessive noise or distortion. The pad acts as a downward extension of the gain control, reducing the level of a loud input signal so the gain stage can handle it without clipping. This is important because if the preamp clips, nasty distortion will be carried throughout the entire recording chain. If the preamp is clipping when set for minimum gain, you need to pad the input. If the preamp lacks a built-in pad, you can buy or make one that can be plugged in between the mic and the input connector. A properly designed pad won't affect phantom powering.

Back when the standard studio mics were an RCA 77DX ribbon or Neumann U47 condenser, console mic preamps had an input impedance in the 10,000 $\Omega$  ballpark, but for quite some time now, an input impedance of 1,200 $\Omega$  to 2,500 $\Omega$  has been fairly common for mic preamps. With today's huge choice of mics, we've discovered that certain ones are affected in a good way (and others in a bad way) when loaded differently. Several preamp manufacturers have responded by offering an input impedance switch. This, too, is a "listening" control – if your preamp has one, listen to how it changes the sound of your mics. Dynamic mics like the popular Shure SM57 often exhibit a very noticeable change in sound as you adjust their load impedance (but be sure to compensate for the associated change in level), while transformerless condenser mics are relatively immune to load impedance. No rules here, just what sounds right.

In the early days, all amplifiers used tubes which, by nature have a high input impedance. The input transformer was there to step it down. Transistors, by nature have a low input impedance, so a transformer is sometimes used to step it up.

## Theme and Variations

With today's more compact studios, mic preamps are often combined with other functions. A "channel strip" adds an equalizer, a compressor, or both to the mic preamp. Preamps are now available with integrated analog-to-digital converters, providing a digital output which can be connected directly to a digital recorder or sound card with a digital input. Digital outputs come in several flavors – optical or coax S/PDIF or AES/EBU for two channels or pairs of channels on a multi-channel preamp, with the ADAT™ optical interface being a popular choice for 8 channel preamps. A relatively new development uses the IEEE 1394 (Firewire) or USB protocol to connect a mic preamp directly to a computer, eliminating the need for a dedicated sound card.

One must ask if the A/D converters in these digital-output preamps are as good as you'd choose if you were assembling components (or as good as the A/D converters that you may already own). The answer is that today, most are really quite good. They're well matched in quality to the preamp's analog circuitry,

making this implementation a good buy for a studio on a budget. Because of certain timing concerns, the jury is still out as to whether the USB or Firewire computer interfaces, which aren't really designed for real-time data transfer, can really keep multiple tracks in phase accurately, but they seem to be working out OK, and they're really convenient.

One problem interfacing a preamp digitally is that there's no place to connect your favorite vintage analog compressor or equalizer. Manufacturers have recognized this and analog Insert jacks have begun to appear on preamps that have a digital output. The Insert jack (or occasionally a pair of jacks per channel to allow for a balanced connection) provides an accessible connection point between the analog output of the mic preamp stage and the input of the A/D converter where outboard processors can be patched into the signal chain.

**Inserts:** While it's useful to have a line level analog insert point ahead of an integral A/D converter, they've started popping up on analog In/Out preamps, too. I never understood the value of such an insert on a single or dual channel preamp, particularly since the insert is nearly always an unbalanced connection. You can just connect your line level processor between the output of the preamp and whatever it goes to next. But they're useful on a multi-channel preamp where the only output connector might be a DB25 and you don't have a patchbay.

Another function that's frequently found in a contemporary mic preamp is a direct instrument input. This is a very high impedance input on a 1/4" phone jack that takes the place of the direct box (DI) that you'd use to connect an instrument pickup to the microphone input. It's a convenient feature, particularly when everything, including the player, is in the same room. Curiously, some preamps are particularly well known for their instrument inputs and rarely get used for microphones.

### **Specs – What to look for when comparing preamps on paper**

If you compare the specs for different preamps, you'll find that they all read pretty much alike. Here are a few things to look for which might influence what goes on to your short list, but it's always a very good idea to choose a preamp by listening to it with both the mics and sources with which you plan to use it.

Gain is the amount of amplification that the preamp can provide. How much gain you need depends on the mics you use and what you record. In my work, which is primarily recording acoustic string instruments ("back porch music") I'll often need 55 to 60 dB of gain. A preamp offering 65 dB of gain is an advantage to me, and one with only 45 dB gain would get little use. If you only use your mics for drums and screamers, maximum gain is not as great a concern, but look for one with an input pad, or be prepared to buy one (they're cheap - \$25 or less). It's difficult to make a quiet, high gain amplifier, so noise or price are sometimes the tradeoffs for high gain.

Gain is calculated by dividing the output voltage by the input voltage. Hybrid preamps with analog input and digital output leave us, when attempting to express their gain, comparing apples to oranges. The gain of the preamp (analog) section of the box can be measured in the lab, but its output is tied internally to the input of an A/D converter, the output of which is a digital word, not volts. While most manufacturers publish a gain specification for devices of this design, you really can't tell for sure what it means in practice. Note that if there's an insert jack or, on a mixer, a direct output, that's a place where the analog gain can be measured, but that may or may not be the measurement that the engineering department sent to the marketing department.

What's important in a preamp with digital output is what level at the mic input, with the gain set to maximum, is required in order to produce full scale digital (0 dBFS) output. That's will determine whether the mic preamp (the box) is sensitive enough for your application. That specification is just starting to appear in published literature (I always measure it when reviewing such a product), but you may just have to try it out.

I recently ran across the specs for a computer audio interface which stated "Gain Range" of 73 dB." This might suggest (as it did to the magazine reader who called my attention to it) that it has up to 73 dB of gain. What it actually means is that the difference between the gain with the pad engaged and the gain control set to minimum and the pad switched out and gain at maximum is 73 dB. Since the pad has 20 dB of attenuation, its actual maximum gain is 53 dB. You have to watch the wording of published specifications, as some of these marketing folks can get pretty creative (or, to give them the benefit of the doubt, don't fully understand what they've been told by the engineers).

Maximum output level tells you how many volts the preamp can put out without clipping. It's important to know this because you'll be connecting the preamp's analog output to the input of another analog device. If the preamp is connected to an A/D converter that requires +24 dBu for full scale (0 dBFS) digital output and your preamp has a maximum output level of +22 dBu, your recordings will never reach full scale. In reality, this is a good thing, but not if you force the preamp into clipping while you're trying to get the meter to hit the top of its range. You won't find a maximum output level specification for a preamp with only a digital output. The assumption (usually valid) is that you can reach 0 dBFS without the preamp clipping, perhaps with judicious use of the pad.

People often complain about a preamp not having sufficient headroom. What they mean is that the preamp clipping level is lower than the clipping level of the next stage. A preamp with a high maximum output will allow you to set your system up with more headroom than one that breaks out in a sweat trying to drive your converters to full level.

Noise – the amount of hiss you hear with the gain cranked up pretty high - is of prime concern. Many manufacturers specify Equivalent Input Noise (EIN). This is a calculated figure derived from the noise present at the output (generally at maximum gain, because that's where the EIN number usually looks best) with no input, and with the gain subtracted from the measured noise level. If the EIN is -127 dBu at 60 dB gain (a typical number), this means the quiescent noise at the output is -67 dBu. (-67 dBu noise minus 60dB gain = -127 dBu) That's not bad, but it's enough to hear if you crank the listening volume way up. Since the maximum output level is around +24 dBu, you still have a signal-to-noise ratio of 91 dB, which is pretty respectable.

Specs are tricky. There are a number of different ways that the same measurements can be made, and they'll get you different results. Guess which result gets published – the one with the best looking number, of course.

## **Class A is Classy**

Any advertising copy writer worth his keyboard will surely use “Class A” in the preamp's description if he can. The term has become so common in this context that for many, it's lost its true meaning. When talking about amplifiers, Class A doesn't mean “A-number-one-best-quality”, it describes the electrical operating point of the device providing the amplification, be it tube or solid state. Here's a primer on amplifier classes.

An amplifier works by controlling a large voltage or current source (the power supply) with a small signal (the input). This led the British to descriptively name vacuum tubes “valves”. The higher the input signal, the more electricity the tube or transistor conducts, allowing a greater portion of the power supply current to flow to the output, sort of like a voltage-controlled variable resistor.

Audio signals normally alternate between positive and negative polarity, but a single tube or transistor works for one polarity only. During the negative portion of the input cycle, the amplifier is “cut off” since the positive power supply, a positive voltage, cannot go lower than 0 volts. By adding a fixed voltage (or current, in the case of a transistor) to the input signal, we can keep the input signal to the gain element positive even during the part of the cycle where the signal goes negative. This added voltage or current is called “bias”. When correctly biased, the amplifier can perform linearly over the full cycle.

Early on, designers designated amplifier classes based on the portion of the alternating cycle over which the gain stage conducts, which is in turn determined by the amount of bias applied. A Class A amplifier is one where the tube or transistor is biased so that it conducts throughout the full cycle. For voltage or current swings centered within the conducting range, amplification is highly linear. Class A operation is quite inefficient, however, since with no signal

present, the tube or transistor is still drawing about half its maximum current, wasting power that's not contributing to the output.

We rarely see Class A power amplifiers outside of the audiophile world since it's expensive to throw away 50 watts much of the time in order to get linear operation over about a 10 watt range. For a small-signal application such as a microphone preamp, however, the power involved is small enough so that we can afford to take a hit in efficiency in exchange for linear operation and simple design. Typically the first stage of a mic preamp is required to deliver only a couple of milliwatts of power, so unless we're specifically trying to maximize efficiency (for example in battery powered equipment) Class A operation is a sound design principle.

You'd think that if there's an A, there should be a B, and there is. A Class B amplifier is biased so that it conducts over half the cycle, and is cut off for the other half. By itself, this is a sure recipe for distortion, but by operating a second amplifier fed by the same input signal with the opposite polarity and combining the two outputs, (this is called push-pull operation) we can make a much more efficient amplifier since there's no wasted power at low levels.

In reality, gain devices don't operate very linearly near zero, so it's a good idea to add some bias so that non-linear operating area is avoided. A Class AB amplifier is biased so that it draws less current than Class A, but more than Class B. There's a Class C also, biased so that the amplifier conducts for less than half a cycle. Class C is common in radio transmitters where it's necessary to amplify only a single carrier frequency. The distortion caused by the extreme clipping is easily filtered by a resonant circuit. Since I'm this far into the alphabet I'll mention that there's Class D and beyond, used to describe pulse width modulated amplifiers (often called "digital"), where the output voltage switches between fully on or fully off well above the audio frequency range, and the length of time it's on determines the amount of energy sent to the output.

## **The Opamp**

An operational amplifier, or opamp for short, is an amplifier that performs a mathematical operation. That's a fancy way of stating our primary goal - to multiply the input voltage by the desired gain. A specific configuration of opamp which is almost universally used in mic preamps is called a differential amplifier. It has two input terminals and operates on the voltage difference between those two inputs. In some applications, one terminal is connected to ground (zero volts), but in other applications, the signal is applied between the two terminals.

Since microphones almost always have a balanced output, with the signal appearing as a voltage between two "hot" terminals rather than between one

terminal and ground, a differential amplifier works very well as the input stage of a transformerless preamp.

## Discrete is Both Neat and Discreet

“Discrete Class A” is a descriptor often applied to a mic preamp. Discrete simply means that the amplifier is built from individual transistors or vacuum tubes, resistors, and capacitors as opposed to being an integrated circuit. That’s how everything was built before semiconductor manufacturers learned how to build more than one component on to a piece of silicon, integrating an entire circuit on to one “chip”. IC (integrated circuit) opamps opened up a whole new area of electronic applications, but after an initial burst of enthusiasm, the audio industry became somewhat disenchanted, and for good reason. Early IC opamps just didn’t sound as good as “the real thing”, either a tube or good discrete semiconductor design.

Opamps are usually designated by number, sometimes prefixed by letters. The first commercially available opamp that was inexpensive enough for audio applications (there were earlier ones) was the type 741, but it was hampered by a low slew rate, the maximum rate at which its output voltage can follow a (theoretically infinitely) rapid change in input voltage. Typical slew rate for a 741 is 0.5 volts per microsecond which, at higher gains, (take out your slide rules) makes it impossible for a 20 kHz signal to reach peak amplitude in the cycle before it has to start heading in the opposite direction. The higher the level, the greater the slew rate distortion.

This seems like sluggish performance when compared to today’s gigahertz computers on a chip, but that was the best the industry could build at the time. Until about 1977 with the introduction of the TL072 and 5532 opamps (which are still common in today’s audio designs), the best sounding solid state amplifiers employed discrete transistor Class A circuitry. Recent opamps developed specifically for audio applications such as the Burr Brown OPA604 series and the SSM2017 (and its extinct but arguably better predecessors) are giving discrete circuitry good run for the money, but there’s still a lot of marketing cachet (and good performance, too) which can be gained from a discrete design.

### **21<sup>st</sup> Century Update:**

Two hot op-amp based IC mic preamps today that are becoming popular, for good reason, are the THAT Corp. TA1512 series and the Texas Instruments (now the home of Burr-Brown designs) PGA2500 series. Both sound very good, and the PGA2500 offers digital gain control which integrates nicely in software controlled preamps.

## Transformers – The Sound of Iron

Input transformers – people either love ‘em or hate ‘em. On the positive side, a transformer is the ideal balanced input, and a preamp with an input transformer is often preferred when operating in EMI-prone environments. A transformer also completely isolates the unit’s ground from the outside world, eliminating hum and noise due to current flowing in the microphone cable shield. A transformer also presents an ideal impedance load for some, but not all microphones.

On the other hand, there’s no such thing as a sonically neutral transformer. A transformer will always change the sound, and whether those changes are good or bad is subjective. Unlike with an output transformer in a mic, the preamp designer knows the exact load (the amplifier circuit) seen by his input transformer, so he can design ringing out of the picture (or in, if he chooses to make that a part of his “sound”), but what he doesn’t know is how the input side of the transformer will affect the mic to which it’s connected. By eliminating the input transformer, one source of distortion is removed, but we might like that distortion and miss it when it goes away.

An input transformer is required to match the relatively low output impedance of a microphone to the high input impedance of a vacuum tube, but with a solid state preamp, which typically has a fairly low impedance input, to use a transformer or not is the designer’s choice. Some designers are committed to the transformer sound and it becomes a characteristic of their products. Others have chosen to do away with the transformer, and at least one manufacturer’s design is basically transformerless, but with a mysterious switch that offers “transformer emulation.” Still others have offered an input and/or output transformer as an option. As a broad generalization, preamps known for their “color” usually have input transformers, while preamps known for their “transparency” can go either way.

Transformer or not, the interface between the microphone and the preamp is the probably the most significant influence on the sound of the mic-plus-preamp system. Predicting combinations that work well together is difficult. There’s no clear pattern – transformerless mics don’t consistently sound better with transformer inputs or vice versa. About the only generality is that modern transformerless condenser mics seem to be less sensitive to input characteristics than dynamic mics.

Good transformers are expensive, so going transformerless is an effective cost-cutting measure which, when good design principles are applied, doesn’t in itself degrade the sound quality of a mic preamp. On the other hand, a step-up transformer can provide noise-free gain, reducing the amount of gain required to be squeezed out of the first amplifier stage.

The output of the preamp can also be either transformer coupled or transformerless. A transformer provides a perfectly balanced output that will interface seamlessly with any input, balanced or unbalanced. It can also provide some nice coloration.

### **Tube or not Tube**

The cherished "tube sound" is often coveted by sound engineers. We've learned that sometimes listeners prefer hearing their music with a small but easily measurable amount of second-harmonic distortion, and most tube amplifiers provide just that. In fact, second harmonic distortion of up to 1% or so can sound quite pleasant even though as little as 0.01% 7<sup>th</sup> harmonic distortion can be quite grating. While tube-induced distortion may sound euphoric, high quality audio is supposed to be about accuracy. Using a tube preamp to "warm up" a digital recording is one of those production decisions that should be made with care, not as a rule.

Of course there are many components inside the box, and their choice, as well as the circuits employed, can affect the sound of a preamp. Resistors contribute noise but have little other effect on sound. Capacitors, on the other hand, can add distortion (explanations why tend to be pretty complicated, but golden ears can tell the difference), and certain types of capacitors age rather quickly; "recapping" a 20 year old piece of audio gear often give it a new life. Good design is good design, and there are few shortcuts that can't be heard if you listen for them.

### **Unplugging**

Preamps cover a lot of ground, ranging from basic amplification to including just about everything but the recorder, and they vary just as widely in both price and quality. This is one piece of gear where sometimes you get more than you pay for, depending on what you're seeking. Read the reviews, look at the specs, but listen closely before making your choice.