Mic Preamps #22

Unless you are entirely recording electronic music, everything you record originates with a microphone and a microphone preamplifier.

What is the function of a preamplifier, anyway?

The term came about in the 1930s to describe the circuit that boosted the very low level coming out of a microphone to a level high enough to drive a power amplifier for a PA system, or a broadcast transmitter. Really, a preamplifier is just a specialized amplifier optimized for its function.

The term “preamplifier” also applies to a gain stage that boosts the very small signal from a phono cartridge or from a tape head. But today I want to talk about microphone preamplifiers.

Dynamic and ribbon microphones generate their minute outputs by converting mechanical motion into an electrical signal using an electromagnetic principle. The voltage coming out of a microphone is tiny, but of course it depends on the sound pressure level at the microphone. If you put a microphone very close to a source of loud sound, the output level can be quite high – much higher than you might imagine.

Conversely, a mic placed a distance from a quiet sound source will produce a very low voltage.

The mic preamp has to handle this wide range, which could easily exceed a 50dB range.

Condenser mics work on a different principle and their output level is considerably higher than the level from a dynamic or ribbon mic. My podcast episode on Microphones goes into a lot more detail on all types of microphones, if you want more information.

A condenser mic typically has an output 20dB higher than a ribbon or dynamic microphone. This demands even more from the mic preamp, and helps explain why most mic preamps have a switchable 20dB pad. More on pads coming up.

A mic preamp has to accommodate mic outputs from about -70dB, for a ribbon mic placed, say, 30 feet from a chamber ensemble, to over 0dB for a condenser mic placed an inch from the head of a snare drum. I talk about this challenge in detail in the episode on vacuum tubes.

The mic preamp has to output a line-level signal of about +4dBm regardless of the level coming from the microphone.

That is a wider range of levels than you will encounter at any other stage of the recording process. No wonder mic preamp design is so critical to the sound of a recording.

The truth is, no mic preamp can handle that range without some compromises. The struggle is always between the headroom and the noise floor. If you increase the headroom, you will also increase the noise.

And, of course, there is a significant difference between how solid-state mic preamps handle this compromise, compared to vacuum tubes.

As you probably know by now, I started out in the vacuum-tube era, and I have always preferred that sound. However, the real world forced me into using solid-state recording equipment for good, practical reasons. But when it comes to mic preamps, I feel strongly that tubes are the best way to translate the sound from your mics into a good-sounding recording.

There is a lot of technical detail involved, but before I touch on that, let's back up and take a wider view of what we do as recordists.

Is our job to recreate the sound that would be heard in the studio? Or to provide the listener with the best possible experience? Maybe for classical recording the goal is to re-create the concert hall in the listener's living room – a challenge that is essentially impossible, especially if the listener can't spend hundreds of thousands of dollars on the room and the equipment. Most people listen on much more modest systems, and that drives how we record.

Since we cannot duplicate reality for the listener, what can we do? I would suggest that we can provide them with a recording that is actually better than reality, especially for most non-classical music.

If we like, we can close-mic every instrument, perhaps with multiple mics, and essentially put the listener's ear just inches from every sound. That's not possible in the real world, but we can do that if we wish. It's not real, but it is effective.

The microphone preamplifier has to take the potentially huge output level of a close-mic'd sound and handle it without excessive distortion. And here, I am using the term "distortion" to mean any deviation from the original sound. That includes the typical harmonic distortion we see in the specifications for our equipment, or as part of the sound of an electric guitar. But there are other types of distortion, such as intermodulation distortion where the electronic circuits create sounds that are the sum and difference of two frequencies present in the music. That distortion is non-musical and very annoying.

Then there is frequency response distortion, where the reproduced range of frequencies does not match the original sound. The most obvious of this distortion is a rolloff of high or low frequencies. But it can also be uneven frequency response, for example, a peak at a particular frequency. That can be audible.

Noise is always added to the original sound, and although noise is usually classified as a separate type of audio deficiency, it is still a distortion in our wider definition.

And there are more subtle distortions, such as phase shifts, that can smear the sound when the fundamental tone of an instrument is offset in time from the overtones of that sound.

And digital audio has its own set of distortions that can be difficult to measure and quantify because they are dynamic in nature and can be pretty subtle. Subtle, at least, until you become aware of those artifacts of digital recording. After that, you can never "un-hear" them. They will always be evident once you train your ear to notice them.

Our hearing is amazingly tolerant of many types of distortion, and some are actually pleasing to us. But distortion is a deviation from reality, and unless you want to use those deviations as part of your creative process, you had better learn how to mitigate them.

In my opinion, many of these distortions are below our level of conscious perception a lot of the time. The more you train your ear, the more obvious they become, which is useful but also a curse.

The average listener will never be able to tell you why some recordings just get on their nerves. But they do perceive this, even if they cannot tell you what is bothering them. How many times have you been enthralled by a song, but then find that you cannot listen to it repeatedly? I think this is fairly common, and listeners don't know why. There could be other explanations, but please consider subtle distortion as one possible reason.

And there are other distortions that we neither have the vocabulary to describe nor the test equipment to measure. I believe this from my own experience.

For example, in developing a new circuit, I rely on my hearing to determine what sounds good. But I also use sophisticated test equipment to quantify my progress. This keeps me from pursuing a dead end, which can easily happen without extensive notes and measurements. I never let the measurements over-rule my hearing, but measurements are useful.

More than once I have had a circuit under test using extremely pure sine wave tones. Although listening to those tones can be very annoying, I find listening useful in some circumstances. And here's what I have found, more than once:

I can control the level going into a circuit in precise one-tenth of a dB steps. While monitoring the output of the circuit with test equipment that measures all the measurable distortions I listed before, and while listening to the pure tone, I can hear a change in the character of the sound with a tenth of a dB increase in level in certain circumstances. There is absolutely no change in the measured harmonic distortion, intermodulation distortion, noise, phase shift, waveform symmetry, or anything else. But I hear the sound deteriorate. This is repeatable, and works even in a blind test when I do not know that the level has been changed.

At first, I thought this was some test equipment anomaly. But even after changing the test equipment, and every other variable in the process, the effect persists.

What am I hearing? I have no idea. It's frustrating, but also kind of reassuring to me that there is still so much about our perception of sound that we do not understand.

The microphone preamplifier was the first challenge I tackled when I was trying to find a way to make recordings sound better.

This process started when I had occasion to listen to a recording I had done in 1968, using all tube mic preamps. I was reluctant to listen to this recording, because I didn't think I could have done a decent job that early in my career. It was true that the recording was not that great in many aspects, but the fundamental sound of the instruments and voices was remarkably good. I quickly realized it was the 1940s era tube mic preamps that made the difference.

In 1992 I wondered if I could re-create that kind of sound, using modern components.

But first I needed to come up with a circuit. This was pre-internet, so research was not as easy as entering a search term and then weeding out the bad information. I had several old textbooks and other books on electronic design, which were published as far back as the 1930s. I also had a few instruction manuals for long-gone vacuum tube equipment I once owned. And I asked friends what schematic diagrams they had, and I went to technical libraries. I ended up with dozens of circuit diagrams for tube mic preamps, and I studied them in detail.

It was obvious that some were the same circuit, perhaps with minor deviations. Designers simply copied an existing circuit. Others were obviously designed to be made as cheaply as possible. Some were somewhat bizarre, with an approach that made little sense to me.

The next step was to build some of these circuits and listen to them. I had a sense of what would sound best, but I wanted to be sure I wasn't missing something that could prove to be useful.

I had many old audio transformers I had collected over the years, and some were suitable as a mic input transformer, and others could be made to work as an output transformer.

The circuit gradually evolved, using the best ideas from a variety of preamps, along with my own ideas. I thought there were compromises made in all those old circuits that limited the sonic potential of those designs. I didn't want any compromises.

It wasn't my goal to copy an old preamp design. What I wanted was something that captured the essence of that glorious sound, but also brought the performance to modern standards.

The original mic preamp circuit, which became the VT-1, was built on a piece of wood in a way that would permit easily changing components, to see what the effect would be.

When those preamps were designed, there was no recording medium that had better than about a -60dBm noise floor, so the designers were not too concerned about noise. But even in 1991, 16-bit digital recording was capable of better than -90dBm noise, and tape machines with Dolby SR noise reduction were in the same range. The old mic preamp designs were pretty good, but none exceeded -65dBm noise. I had to make my design compatible with the low-noise recording devices. That meant at least a 10dB reduction in noise.

Distortion, too, was pretty high in those old designs. Some of that was due to the type of components they had available at the time, but I suspect that there was not much pressure to reduce distortion since there was so much distortion in all the rest of the audio chain. Back then, it was a challenge to get distortion below about 5%, and a lot of equipment had much more distortion than that. I wanted to get well below 1% distortion.

I also discovered that the old designs did not consider phase shift through the circuit. Or, if they did, it must not have been considered an important issue.

The old transformers I was using were surprisingly good, but I needed parts I could buy in quantity today. When it comes to audio transformers, Jensen is a known leader. There are others that make really good transformers, too, but Jensen was my first choice.

The folks at Jensen were very helpful, and really seemed enthusiastic about what I was trying to do. Bill Whitlock, in particular, was patient with me and educated me about the subtleties of not only transformer design, but also the circuitry around the transformers. He had a sophisticated circuit analysis program he developed for this purpose, and ran my proposed design through the software and came up with some subtle changes that he thought would improve my design. Eventually, Bill gave me a copy of the software so I could experiment with changing component values and see what the results would be.

The main benefit of the circuit analysis software was in minimizing phase shift through the circuit, and predicting what the frequency response at the extremes of the audio range, and beyond, would be like.

Jensen also designed transformers for me for my preamp design – transformers that we still use today.

With the optimized circuit, the custom Jensen transformers, and a lot of listening and measuring, I came up with the final circuit. That circuit has been unchanged since the VT-1 single-channel mic preamp was introduced at the New York AES Show in 1992. And it is the circuit used in the VT-2 dual-channel and VT-24 4-channel mic preamps we make. In fact, a lot of that original design concept is used in the line-level stages in the other products we make, like the VT-4 and VT-5 Equalizers, and the VT-7 Compressor.

That commonality gives all our products a consistency in sound.

I was pleased with the sound of the VT-1, but I was not sure if it could be a viable commercial product. I didn't know if other people would respond to the sound I preferred, and I was concerned that the price of the VT-1 would be considerably higher than other mic preamps.

But there were a couple of things I was sure of: I was not going to build to a price, and I would do nothing to compromise what I thought the preamp should sound like.

This was counter to the prevailing approach for product design, where you start with market research to determine what the potential customers think they want and what they are willing to pay, and then designing the product to match. I did not want to do that. I wanted to make the best possible preamp I could, let the price fall where it must, and see if there were people out there who heard what I heard, appreciated the difference, and were willing to pay a premium for it.

I figured, worst case, I would have some great mic preamps for my own recording.

By the generally applied rules for pricing a product, my products are quite a bit below what the rules say. I could do that because I kept overhead low, and taking a minimal amount out of the sales revenue for my personal use.

But it was also critical that the enterprise was profitable enough to support me and my assemblers, and have enough cash flow to ensure the continued health of the company. I wanted to be able to support the products as long as possible, and that meant staying in business.

Fortunately, the recording world responded very well to the sound of the VT-1, even if people were shocked at the price, which was $2000 at its introduction – a lot of money for a single-channel mic preamp in 1992.

In 1993, I introduced the VT-2 dual channel mic preamp, which is still our best-selling product. And for people who had limited rack space but needed a lot of preamps, I developed the VT-24, a 4-channel version. All of these are 3 rack units high.

At the AES and other shows, I could count on three questions from people: Why is it so big; why is it so expensive; and why is it red?

It was big because I did not want to mount the tubes horizontally, which compromises cooling and thus tube life, and affects performance as the tubes age. It's also big because of the large components that I needed to achieve the quality I wanted.

It was red because, on a whim, I chose a can of red spray paint at the hardware store for the front panel of the first prototype. I like red, but I planned to use a less exotic color for the production units, probably a bluish-gray. But when people saw the prototype, they loved it. It made a statement in their rack, and they urged me to use that color. It didn't take much to convince me – I liked the way it looked, too.

For a quality piece of gear, the aesthetics had to be quality, too. I decided to use quarter-inch thick machined aluminum for the front panel, and 1/8-inch thick aluminum for much of the rest. These are twice as thick as most products use. The visible controls had to look good, too, which meant expensive knobs and switches, and a real VU meter. And the layout had to please me with a nice sense of rhythm and balance.

The controls and indicators had to have a quality feel and appearance. The ergonomics had to be good, too, for use in the real world of recording. Everything had to be easy to manipulate, easy to understand, and help the engineer in quickly getting the settings needed at the moment. That meant attention to the way the level changed with the rotation of the controls, for example. And a layout that made the most often-used controls dominant.

And the cost? Well, when you do not compromise on parts or construction, it gets expensive to build. I did not go overboard with super-expensive parts that offered no benefit, but I did use the best parts available.

And since it was expensive, and sounded really good, I thought it needed to look really good, too. The paint I decided to use for the front panel is DuPont Imron, a polyurethane finish used on airplanes. In fact, the first 100 or so front panels were painted in the maintenance shop at the airport where my airplane is based. They had extensive experience using that paint, and they knew it had to be perfect. They were used to painting airplanes that cost millions, and those owners would not accept any defects in the finish.

I also wanted the products to last a long time. I have 1950s test equipment made by Hewlett-Packard, back when that was their focus. It still works perfectly today, and some of it has the original tubes from 60 years ago. Well designed and well-built equipment can last for decades, and that is what I wanted my products to do.

I design for a minimum of a 50-year life. Some parts, like tubes and electrolytic capacitors, might not last that long, but everything else will. And for those parts that will eventually need replacement, I made the design so that replacement was easy.

And I used point-to-point wiring for all the audio path. That's a technique from the vacuum tube era that minimizes the length of the wire paths within the circuit, and also allows the circuit to be built in a three-dimensional manner, unlike the two-dimensional limitations of a printed circuit board.

Why is that important? Well, keeping the audio path as short and direct as possible improves the sound by reducing the possible interference from one part of the circuit coupling into another part of the circuit. It also reduces the noise level and improves the high-frequency response.

From my experience as an Amateur Radio operator, and my time spent in radio broadcasting, I had a lot of background in designing devices that operated at much higher frequencies than audio. At those high frequencies, not only would poor layout and excessive wire lengths compromise performance, inattention to those details could make the circuit not work at all – or self-destruct. I use that radio frequency type of construction in my designs, which improves performance.

Point-to-point wiring requires that the assemblers are skilled and experienced. I have always been grateful to have found really excellent craftsmen (and women) who know how to build in this style. They take a lot of pride in their work, which is one reason why the assembler's initials are on every piece of equipment we make.

The assemblers appreciate the feedback that their work is on a particular hit record, or a photo of a unit they built in a studio like Abbey Road.

Attention to detail in the design is vital. If I make a decision that improves the performance by one percent, it is unlikely that the change will be noticeable in most circumstances. But if I can find ten of those improvements, then maybe the product will be 10 percent better. And that is worth it and can be heard.

No compromises. To me, they are just not worth it. If the price becomes higher than any other comparable product, so be it. That's the product I want my name on.

Another design criterion was whether I would use this device in preference to anything else that was available. If it didn't meet that test, I did not want to release it. I have plenty of prototypes of products sitting in storage that just did not make the cut. I usually revisit that design from time to time, and sometimes an idea comes to me that raises the performance enough to continue development.

Before any product is put into production, I use it in my own recording projects for a while, just to see if anything needs to be changed. It could be that the design falls short under certain circumstances. I have to fix that.

I never want any piece I designed to sound bad, under any reasonable circumstances.

That's the design philosophy. Now let's look at how to use one of our mic preamps in actual sessions. This is specific to the VT-1, VT-2, and VT-24 products, but many of the principles can be applied to any microphone preamplifier.

I'll also answer some questions that I get asked fairly frequently.

Let's start with the installation of your new preamp in your rack. Heat is the enemy of all electronic equipment, whether it is tube or solid-state. Heat dries out electrolytic capacitors. In fact, their life is directly proportional to their operating temperature. The cooler the equipment, the longer those capacitors will last.

This applies to digital equipment as well, which often runs very hot. Most of those boxes will be obsolete in a few years, so designers have no reason to build them for a long life.

To prolong equipment life, my recommendation is to use a vented 1-rack unit panel above and below the mic preamp. In fact, do that with all your rack gear if you can. Your equipment will be more reliable, stay in proper alignment longer, and last longer.

We make a vented rack panel that matches our products, but you can use any vented panel if you like.

Audio wiring should be done with high-quality cable. It does not have to be expensive, esoteric cable that costs a lot of money. Good quality balanced twisted pair audio cable from any reputable manufacturer will be fine. I use various cables made by Gotham Audio, but there are several other that make excellent cable.

Keep your audio cables and digital cables separated. You may get away with audio and digital cabling place together, as long as both the audio and digital circuits are balanced and there is no chance of the equipment being connected to a device with an unbalanced input or output. But why take a chance? Just keep them separate, and if they have to cross, have them cross at right angles.

I am strongly opposed to typical TRS patch panels for routing microphone lines. There are a variety of reasons for this, including potential damage to your ribbon mics, but also because patch panels are an insidious source of noise and distortion. This happens when the plugs and jacks become even slightly oxidized. You can tell because they are no longer a bright, shiny color. This actually creates a semiconductor junction where the patch plug and jack meet, which is essentially a distortion generator. This is how many guitar pedals work, increasing the distortion. Great for a guitar sound, but terrible for your recording.

And this noise and distortion is insidious because it will slowly increase over a period of months or years until one day you realize everything doesn't sound as good as it used to.

Mic level patching is particularly vulnerable to this problem. But line-level patching can also suffer.

In my studio, I use only XLR connectors to make a patch panel. Sure, it takes up a huge amount of space comparted to the jackfield typically found on a console patch bay, but XLR connectors, by their design, are practically immune to the oxidation problem.

AC power? This is an area frought with misunderstanding and sometimes even charlatanism. Use good quality outlet strips and the power cords that came with your expensive gear and you will be fine.

Power conditioning? Unless you work in a building with really antiquated wiring, your AC power is probably just fine. Your studio should run on a dedicated AC circuit that powers only your equipment, installed by a skilled electrician. The details of this are beyond the scope of this episode.

All the connectors in the microphone path should be highest quality XLRs. I use gold-plated Neutrik connectors on all our products, and everywhere in my studio. Swtichcraft is another dependable manufacturer of good XLR connectors.

Keep the number of connectors to a minimum. It is impossible to avoid at least a few connectors in the path, but try to keep the number as low as possible.

Preamps in the studio? In most studios with a separate control room, the mic wiring is at least fifty feet in a small studio, and perhaps hundreds of feet in a large room. Does this matter?

The theory is that the very low level signal on a mic line will deteriorate as the cabling gets longer. There is some basis for this, but I have done experiments with spools of 1000 feet of quality cable inserted in the mic path and I can neither measure nor hear any change. This is because of the low impedance, balanced circuit involved. That's the simplistic answer, without getting too technical.

The idea of putting the preamps closer to the mics would seem to be preferable, because the signal going back to the control room over 100 feet of cable will be at a higher level and thus less susceptible to noise and other degradation.

I've tried it both ways, and if there is a difference, it is very subtle.

Having the preamp close at hand probably outweighs any other benefit.

On our preamps, there is an Input control with several positions. The first one, labeled 0, means that the mic signal goes directly into the mic preamp circuitry. This is the setting you will use most of the time.

But sometimes the level from your mic is so high that it overloads the mic preamp in a way you do not want. In that case, the second position, labeled -20, may be desirable. This inserts a 20dB pad before the input transformer. Saturation of the input transformer is possible with extreme levels, and that sound is not particularly useful for most recording. The pad eliminates this.

You may wonder why there are not several levels of padding available. Well, to maintain the proper load on your mic, and provide the preamplifier with the impedance it is designed for, the minimum loss that a pad can be is about 20dB. This degree of attenuation is perfect for most circumstances, fortunately.

Some consoles have adjustable preamp gain, which might be continuous or stepped. Usually this control not only switches in a pad, but may also adjust the gain of the first stage of the mic preamp. This is also a viable approach, and works well with the typical op-amp mic preamp stage.

Does the pad change the sound? Well, it has to, to some degree. By reducing the input level, the tube stages are dealing with a lower signal level, and depending on many factors, that may subtly change the sound.

My experience with our mic preamps is that the pad can change the sound, especially with certain mics. The technical reasons are complicated and won't be covered here, but it may not be your imagination if you find the sound changes with the pad. This, of course, presumes that you have compensated for the changed level elsewhere in the chain.

For me, I almost always prefer the sound without the pad. Maybe I like how the sound is fattened by the subtle overload. Whatever, I use the pad only when absolutely necessary. That may mean running the Attenuation control very near its minimum setting, but this is not a problem, as I will explain in a minute.

The third position on the Input control is labeled Lo-Z, and it is used with microphones that have a non-standard, below normal output impedance. This includes some of the Schoeps and Neumann condenser mics that use a transformerless output.

To understand this, we have to get slightly technical, and see how this evolved over time.

Soon after the Audio Engineering Society was founded in the late 1940s, they set up standards committees to establish best practice guidelines for manufacturers and users. This is still a core mission of AES, and covers way more than the audio we normally use in the recording studio.

Microphones in the 1930s and 1940s had a wide range of output impedances. Every mic designer seemed to have his own preference.

AES studied this and came up with a simple recommendation to address this problem. The microphone should have an output impedance of 150 ohms and the mic preamp should have an input impedance of 1500 ohms.

This 1-to-10 ratio is common in many audio impedance compatibility circumstances. I won't go into detail here.

Ribbon mics generally have a higher output impedance than 150 ohms. 300 ohms is common, but some are even higher. And those mics work best into a higher preamp input impedance – as high as 10,000 ohms. But that is a separate topic that I won't cover now.

Dynamic and condenser mics are easier to deal with, and the majority of manufacturers conform to the 150-ohm standard. That's easy to do when the mic has an output transformer.

But quality transformers are expensive, and take up a lot of space in a microphone, so mic designers have reduced the cost by using electronic circuits to provide some of the functions of a transformer. In general, this works pretty well. It only works with solid-state mic electronics, like you would find in a phantom-powered condenser.

There are claims of superior performance by getting rid of the transformer, but I am not convinced. I think it just makes the mic cheaper to make.

But we are stuck with many mics that have a transformerless output. They don't sound bad at all, so it's not a big loss. But those transformerless circuits have an output impedance that is lower than 150 ohms. The actual impedance may range from 30 to 100 ohms.

Not a disaster, but the sound of those non-standard mics can often be improved by designing the preamp input to accommodate the lower impedance.

By the way, "Z" is the electronic abbreviation for impedance.

My design is derived from research done by Jensen Transformers, who designed an input circuit, before the preamp input transformer, that provided a better match for those low impedance mics. It cannot optimally match all the mics, since their impedances can be all over the place, but it does help most of them.

The difference in sound can be subtle, and you may prefer the sound of the 0 position or the Lo-Z position on some mics on some sounds. It's a tool you can experiment with.

Generally, using the Lo-Z position tends to tame some of the annoying extremely high audio frequencies and make them sound more natural. Try it and see what you think.

Soon after the VT-1 mic preamp was introduced, I ran into people who raved about how a pair of them improved their mixes when they put them on the mix buss. I was initially appalled by this, because the line level was far too high for the mic preamp, and the impedances were wrong, too.

But I had to agree that the preamps on the mix buss did improve the sound for many projects.

So, I decided to address this with an accessory box that provided the proper level and impedance when using a VT-1 or VT-2 in this application. That was our LP-1 line pad, which drops the level down to something the preamp can easily handle, and avoids excessive loading of the mixer stereo buss. We have since sold thousands of the LP-1 boxes, and not just to VT-2 owners.

The LP-1 drops the level about 43dB, which still provides a fairly hot level into the mic preamp. Often that gives you the sound you want. But by using the VT-2 20dB pad in conjunction with the LP-1, the level is reduced even more, which may sound better to you. Experiment with both 0 and -20 on the VT-2 to see which gives you the effect you seek.

We plan to incorporate the function of the LP-1 into our VT-2 preamp soon, so that the separate box will not be required.

The control of level through the mic preamp is adjusted with the Attenuation knob. Why is it called Attenuation and not gain? Functionally it is the same thing no matter what you call it, but I chose to label it Attenuation because of the way the circuit is designed. By the way, the VT-24 labels say Gain, mainly because of the space available. The circuit is the same as the VT-2.

In all our preamps there are essentially two gain stages. The first stage is actually two triodes, but connected in such a way that they function as one amplifier stage.

The second stage provides more gain, and the last tube is used simply as an impedance transformation, with no additional gain.

The Attenuation control is in between the first and second stages.

In other words, the first stage (two tubes) always operates at full gain. The only way to change that is with the -20 pad. This was a conscious design decision after a lot of analysis and listening, and time has proven it to be the right choice.

The Attenuation control does not affect the gain of any stage. It only affects the level somewhere in the middle of the path. Attenuation seemed to be the best descriptor of that, although if you want to call it "gain," that's fine with me. I tend to do that myself.

The Attenuation control is designed to provide an easy-to-adjust range of gain. The control is never touchy or critical, but rather it smoothly adjusts the gain throughout its range.

Also included in all our preamps is a Phase switch. This reverses the polarity of the mic signal, which can be critical in many situations, particularly when multiple mics pick up the same sound. An example would be a top and bottom snare mic.

Technically, this is a polarity reversal, not really a phase change. But consoles have usually labeled this control as phase, or used the electronic symbol for phase which is a zero with a backslash through it. For consistency with practice at the time I designed the preamps, I used the term Phase. However, on the VT-24, I changed that to Polarity. It's the same thing, but Polarity is more technically accurate.

Microphones should be standardized in their output, so that, in theory, all mics in a session should have the same phase relationship. But not all mics conform to this standard. And any mis-wiring in your cables can also introduce a polarity change.

There are times when you need to reverse the polarity of a mic either because of the way multiple mics are picking up the same sound, or because of an error in the mic or the wiring.

Normally with one mic picking up one isolated sound, the polarity is irrelevant.However, I discovered early in my career that when I hear my own voice live in headphones, the polarity does make a difference. The sound reaching your ears from both the electronic path and the mechanical path through the bones of your head, can either conflict or emphasize the level and character of the sound you hear.

Maybe I am unusually sensitive to this. I have asked vocalists which position of the phase switch they prefer in their headphones, and many of them say they do not hear any difference.

To me, the “out of phase” position provides a louder sound in the headphones, with a lot more bass. I prefer that. But everyone is different and I may be an outlier in this. It could be worth a try, however.

Note that in digital recording there is almost always some latency heard in the headphones. Ideally, this is below 5mS, a delay that most musicians can ignore. But even that short delay changes the perception of one’s voice in the headphones. Changing the polarity may make this better or worse for the performer. Pure analog recording does not have any latency.

All our mic preamps have true VU meters, which means they conform to the standards established long ago when VU meters were the predominant way of measuring levels. Actually, the VU standard is still an excellent way to rapidly get an idea of not only the level, but also the apparent loudness of a sound. I won’t go into detail, but the VU presentation is still quite useful. Digital requires a presentation that shows the true peak level, which is critical since digital overload can ruin a recording, but that’s a different issue.

The VU specifications call for a predictable response of the meter needle to sound, which an experienced engineer can interpret easily. The standards also specify the range of the meter markings, and even the colors used.

Getting that ballistic response right is not an easy task, and VU meters are quite complex mechanically and electrically, and that makes them expensive. I thought a quality piece of professional equipment should have that level of quality consistently throughout the design, so I chose to use a real VU meter.

Many manufacturers use cheaper meters, which is probably not an issue for most recording these days. But the response of those meters to the audio signal just never looks right to me. I find the information from a real VU meter to be much more useful.

There are times when you may wish to drive the preamp very hard, in order to achieve a particular sound. That may mean that the meter needle will be slamming the stops, which is not only annoying (you can hear it), but eventually it will damage the meter. That’s why I put a meter on/off switch on the front panel.

Due to panel space, there is not a meter switch on the VT-24.

When solid-state condenser mics were introduced in the 1960s, the need for the big power supply for tube condensers went away. But the solid-state mics still needed power, although the requirements were much simpler than for tube mics.

An easy way to power condenser mics with existing XLR cables was proposed.

To understand how this works, we have to go back to the early days of the telegraph. Wire lines were expensive to make and maintain, and as telegram traffic increased, the system soon ran out of capacity. Smart engineers figured out a way to put two telegraph signals on one pair of wires. One went the usual way, and the other was called the "phantom" signal, since it was invisible to the main circuit.

That invention went on to be useful in other applications.

When I worked at WPEN in Philadelphia in the 1960s, we used those phantom circuits in several ways. One was for a remote broadcast that came from a church service every Sunday morning. The equipment was installed at the church, connected to the studios by a dedicated broadcast telephone line.

Normally this would require an engineer to be at the church to turn on the equipment and make sure it was working properly.

But the remote broadcast used only one microphone, and there wasn't much for an engineer to do. And, needless to say, sitting there monitoring the RCA OP-6 was not everyone's idea of an ideal way to spend a Sunday morning.

So engineering devised a scheme to send a DC voltage down the telephone line to operate a relay that would turn on the equipment when it was needed. Normally a DC voltage on the broadcast line would cause all sorts of problems, but this system used a phantom circuit to isolate the audio and the DC. It worked great.

Well, the solid-state condenser mics needed voltage to run, too. The phantom circuit was a simple way to do this. It did not require a special cable and connectors like a tube condenser mic. Brilliant! You could use your normal cables.

In theory, non-phantom powered mics would simply ignore the phantom voltage, so the power could be on all mic lines all the time but only the condenser mics would detect it and use it.

That is true for perfect wiring. In fact, some consoles from that period had the phantom power on all the time, to all inputs. You couldn't switch it off.

But if there was a fault in a cable, the 48 volts used to power the mics could suddenly become non-phantom and cause damage to ribbon and dynamic mics.

The classic fault was in the patch bay, however. As the patch cord was inserted, it would momentarily unbalance the mic line and put the full 48 volts into the mic. That's really destructive to ribbon mics, and not great for any type of mic.

Consoles, and later standalone mic preamps, put switches on each input so that the phantom power could be turned on only when needed.

These switches were usually labeled 48V, to indicate the phantom voltage level.

Without getting into too much detail on the phantom powering system, let me just say that the scheme requires two tightly matched resistors across the mic line. If the resistors are not matched, a voltage will appear across the mic, which can cause problems and potential damage. Fortunately, precision resistors are easy to find these days, so matching is simplified.

But to be sure, we measure every resistor used in this phantom power circuit to make sure they truly match. We start with 1% resistors and then match them to one-tenth of a percent maximum.

Also, when not needed, those extra resistors across your mic lines can have adverse effect on the performance of your mics. So in our mic preamps, turning off the 48V switch not only removes the phantom power, but it also switches the resistors out of the circuit entirely. Just one of those “one-percent” improvements that add up.

To learn more about the sound of tube mic preamps, I suggest you listen to my podcast episode, “Vacuum Tubes: Why They Sound Better for Audio.” And for more detail on mics, listen to the “Microphones” episode.

That’s the story of the D.W. Fearn mic preamps. I hope you will find this information useful to you the next time you set up a session. Deeper understanding of what goes on inside all those expensive boxes in your studio can only help you do a better job.

This is My take On Music Recording. I’m Doug Fearn. See you next time.