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Amplification Components

PART 1

__Audio Perfectionist Journal__ #9 explained the importance of source components. We demonstrated that source component quality is critical because the component that retrieves the signal from the recording sets the limit of resolution for the entire system. Remember this fact as presented in __Journal__ #9: Any musical signal that emerges from the speakers must have been retrieved from the recording by the source component.

If the source components fail to retrieve the musical signal from the recording, that signal certainly won't be reproduced by the speakers, no matter how expensive or exotic they may be. But the source component can't drive the speakers directly; the signal must first be processed and amplified. Here's a fact that will be demonstrated in __Journals__ #10 and #11: Any musical signal that emerges from the speakers has passed through the amplification components and you'll hear any alterations that occur in the process.

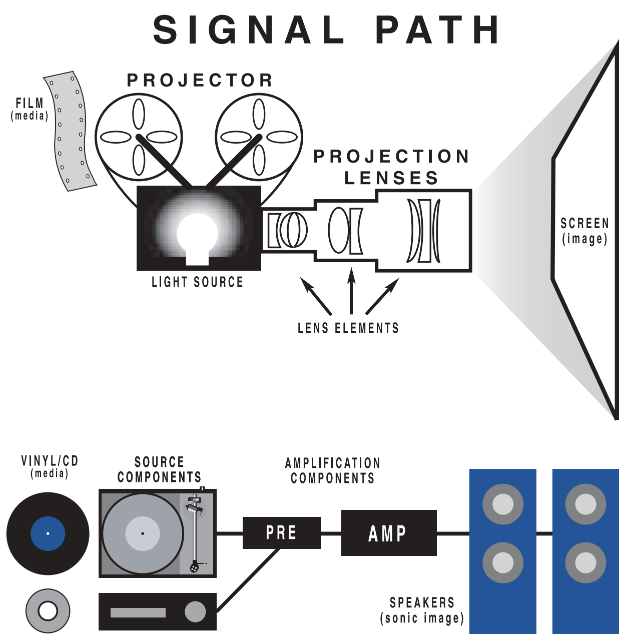
The signal that the source component retrieves from the recording must be processed and amplified to a level sufficient to drive the loudspeakers. The amplification components, which include a preamplifier and a power amplifier, provide this processing and amplification. The quality of the amplification components is critical for obvious and not so obvious reasons. If the source component retrieves all the information stored on the recording and some of that information is lost or altered during amplification, even flawless speakers will reproduce a diminished or colored replica of the original event. The small signal from the source component must be processed, buffered and amplified with minimal loss of signal information and a minimal addition of noise and distortion. This is a far more difficult task than you might think.

Audio Perfectionist Journals #10 and #11 will explain how amplification components work and how they can potentially degrade the signal. **Journals** #10 and #11 will suggest the best ways to choose and assemble these components to minimize this signal degradation in order to get the highest quality sound from the speakers.

Common Myths

You have probably been presented with a lot of false or misleading information about amplification components and we'll have to start by exposing some widespread myths. Regardless of misinformation that would have you believe otherwise, amplification components are critical to the sound you hear from the speakers because they create an amplified replica of the signal from the source component. This will be explained.

You may have been told that amplifiers make little difference in sound quality. False. You may have been told that you can get high-end sound without the use a of a high-end preamplifier and a high-end amplifier. You can't. You may have been told that an expensive surround sound processor can replace a high-end preamplifier with no sacrifice in sound quality. Not true. You may have heard that modern receivers are nearly as good as high quality separate components. Sorry, not even close to true.



In a digital age, preamplifiers may seem to be superfluous. They are not and we're going to tell you why. Some manufacturers would like you to believe that surround sound processors and receivers can offer equivalent performance to high-end separate components. They cannot and we'll show you why.

Modern amplifiers, even those in receivers, have miniscule amounts of measurable distortion. This leads many people to assume that amplifiers are essentially perfect and that most can provide equivalent performance for high-end music reproduction. This is certainly not the case and we'll explain why in this **Journal** and the one that follows. APJ

an introduction to...

Amplification Components

by Richard Hardesty

*Let's start by defining the component names we'll use and explaining some technical terms that you may encounter here and in magazine reviews and manufacturer advertising. This will help you to fully understand the information that will be presented in **Journals** #10 and #11 and to wade through the rhetoric you may encounter elsewhere.*

Head Amp

A head amp is a device that is used to raise the output voltage level from a moving coil cartridge to match the input sensitivity of a phono stage that is designed for moving magnet or moving iron cartridges. Head amps are essentially obsolete because virtually all of today's phono stage preamps can directly accept the signal from a low output moving coil cartridge. Here is some basic information about head amps for those with vintage preamps.

Using a (low output) moving coil cartridge with a phono preamp designed for moving magnet/moving iron cartridges will require the use of a head amp or step-up transformer. Here's why.

A moving magnet or moving iron phono cartridge will typically produce 2mv to 5mv (.002-.005 volts) output at a groove velocity of 5cm per second. A moving coil cartridge may deliver one tenth of this output voltage, or about 200-400µv (.0002-.0004 volts) at the same groove velocity. If you have an older phono stage with input sensitivity suitable for moving iron cartridges, you'll need a head amp or transformer to raise the output voltage from a moving coil cartridge to a level that will drive your phono stage. Impedance matching is also a factor.

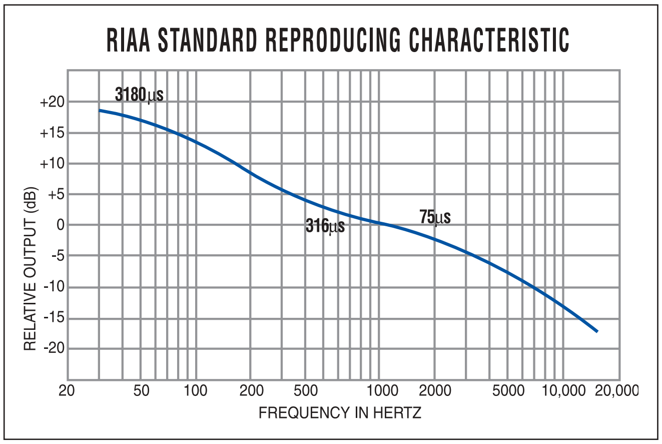
Most phono stage preamplifiers designed for moving magnet/moving iron cartridges have an input impedance of about 47k ohms. Most moving coil cartridges will perform best into an input impedance of 100-500 ohms. A head amp, or transformer, will provide the proper input impedance and electrical damping (loading) for a moving coil cartridge.

The output of a head amp is a phono level voltage (typically 5mv or so). A head amp feeds the phono input of a preamplifier or phono stage. Head amps usually don't provide phono equalization (see next page).

Phono Stage or Phono Preamp

A phono stage is a preamplifier designed to provide the gain and equalization necessary for vinyl record reproduction.

Phono cartridges produce output voltages that are very low compared to line level sources like CD players. Lots of voltage gain is required to raise the cartridge output voltage to a level which is sufficient to drive a line stage preamplifier or an integrated amplifier.



Vinyl records are made with equalization that boosts high frequencies and reduces low frequencies as they are recorded on the disc. This equalization allows the reduction of groove spacing to extend playing time when low frequencies are present in the recording, and raises the level of high frequency information relative to surface noise. A phono stage provides equalization that restores flat response during playback while reducing surface noise. Recording and playback equalization follow the RIAA curve established by the recording industry. See illustration.

The output from a phono stage is a line level voltage. This signal is usually fed to a line level input on a line stage preamplifier that can provide additional gain and attenuation. A phono stage may have provisions for adjusting gain and cartridge loading to make it compatible with a wide range of cartridges but phono stage preamps usually don't have input switching or volume controls which are found on line stage preamps.

Preamplifier or Line Stage

A line stage preamplifier is a control center for a home audio system. It will usually provide volume control and input switching and may provide some voltage gain. Line stage preamplifiers work with line level source components like CD players, and usually don't provide the extra gain and equalization necessary for vinyl record reproduction.

Some preamplifiers incorporate a phono stage, providing the functions of both phono stage and line stage preamplification in one unit.

A preamplifier produces a line level output signal with limited current capability and is designed to drive a high impedance load (10-200k ohms) like a power amplifier.

Power Amplifier

A power amplifier receives a line level input signal and provides output with the voltage and current necessary to drive a loudspeaker. The output from a power amplifier is a speaker level signal with high current capability designed to drive low impedance loads (typically 4-8 ohms) like loudspeakers.

Amplifiers must provide power (the product of voltage and current) to perform work (drive a loudspeaker). See analogies that follow.

Power amplifiers, sometimes referred to as basic amplifiers, don't have volume controls and input switching capability. Amplifiers with volume controls and input switching are called integrated amplifiers.

Integrated Amplifier

An integrated amplifier is a combination of a preamplifier and a power amplifier. An integrated amplifier will usually provide the features of a line stage preamplifier, including input switching and volume control, and will drive loudspeakers directly like a power amplifier. Integrated amplifiers may also include a phono preamplifier.

Receiver

A receiver is a combination of a radio tuner, a preamplifier and a power amplifier on a single chassis. Other components such as a phono preamplifier, a surround sound processor, a video switcher, and a digital-to-analog converter may also be included in the same unit.

Receivers with surround sound capability and video switching are often referred to as audio/video (AV) receivers.

A receiver performs the tasks of several separate components but the performance of each of these components will be compromised. We are aware of no receivers that can provide high-end audio performance so we will generally ignore this product category. In this **Journal** you'll read about the compromised audio performance of surround sound processors. All this information also applies to receivers, which are compromised even more because of the inclusion of power amplifiers.

Surround Sound Processor/Digital Controller

A surround sound processor, or digital controller, is a combination component with no amplifier. These devices are often marketed as a replacement for a preamplifier but, as we'll see, they cannot provide high-end performance for music reproduction and should be thought of instead as an additional component that is placed outside the 2-channel signal path.

Surround sound processors usually include microprocessors to decode compressed digital signals like Dolby Digital and DTS streams, and must have digital-to-analog converters to convert these decoded digital signals to analog for external amplification. A surround sound processor will have controls like a preamp and may also incorporate other components such as a video switcher, a phono preamp, a radio tuner and a multiroom distribution controller. Each year, new features are added to surround sound processors that increase complexity and don't improve sound quality. You can read about this stuff in home theater magazines. We will discuss surround sound processors in terms of music reproduction because many are laboring under the misconception that surround sound processors can provide high-end audio performance.

The output from a surround sound processor is a line level signal that must be amplified for reproduction.

Balanced Amplification Components

An "unbalanced" component has a single signal path (try to say that rapidly). The signal level is referenced to ground and the component ground plane is usually the signal return path. A balanced component has two signal paths that are referenced to each other and both are usually isolated from the component chassis ground plane.

Some components have balanced signal paths through only a portion of the audio circuitry providing some of the benefits of balanced operation.

There is significant debate about the value of balanced amplification components. I am a proponent of balanced amplification because it reduces noise levels and solves many cable problems. There is an audible difference between balanced and unbalanced components of equal quality but the benefits of balanced operation are often subtle.

The biggest advantage of balanced operation is common mode rejection, which cancels extraneous noise common to both legs of the balanced circuit. Cables are a significant source of this noise and complicated systems have lots of cables which are often bunched together and in close proximity to AC lines. The only disadvantage that I can see is cost because all signal path components must be duplicated for true balanced operation.

Bias Current

Some constant current flow (bias) through a tube or transistor allows the designer to utilize the device to conduct signals within the region where its performance is most linear.

The amount of bias current that is conducted by the output devices in a linear amplifier determines class of operation. When the devices are biased so that they conduct continuously, crossover or "notch" distortion in the output stage is eliminated and the component is said to operate in Class A.

Most preamplifiers operate in Class A. Most amplifiers—even those that are advertised as Class A—don't operate in true Class A. There will be more information about this in **Journal** #11.

Negative Feedback

Negative feedback (also called degeneration or error correction) is a technique used to regulate gain and reduce distortion. A feedback loop takes a portion of the output of a circuit and applies it out-of-phase to the input of the same circuit.

The feedback loop in a component with "global" feedback applies a portion of the final output to the input of the component encompassing all the stages in the component. "Local" feedback encompasses one stage within the component and not the whole thing. Each stage of a multistage component may incorporate local feedback.

It may be possible to design an amplifier with no feedback of any kind but I've never encountered one. Amplifiers that are advertised as having zero feedback or no feedback may have no global feedback but you can be pretty sure that some local degeneration is employed.

Ohm's Law

In a direct current (DC) circuit, polarity (positive and negative) remains constant and current flows only in one direction. In an alternating current (AC) circuit, polarity is constantly changing causing the direction of current flow to alternate. Amplifiers produce an AC output signal from an AC input signal representing music by modulation of a DC power supply potential created from an AC power source (the wall socket). This will be explained in **Journal** #11.

The characteristics of an electrical circuit can be described in terms of current, voltage and resistance (or impedance in an AC circuit with reactive components). The relationships between these characteristics are defined by Ohm's law. The symbols (terms) that represent these characteristics in electrical equations are I (current in amperes), E (voltage in volts) and R (resistance in ohms).

P =

$\frac{E^2}{R}$ $\frac{E^2}{Z}$ $E \times R$ $E \times I$ $I^2 \times R$ $I^2 \times Z$

I =

$\frac{E}{R}$ $\frac{E}{Z}$ $\sqrt{\frac{P}{R}}$ $\sqrt{\frac{P}{Z}}$ $\frac{P}{E}$

E =

$I \times R$ $\sqrt{P \times R}$ $\frac{P}{I}$ $I \times Z$ $\sqrt{P \times Z}$

R or Z =

$\frac{E}{I}$ $\frac{E^2}{P}$ $\frac{P}{I^2}$

An AC circuit may contain reactive components (inductors and capacitors). The resistance to current flow (impedance) varies with frequency through reactive components in an AC circuit. The symbol for impedance is Z (impedance in ohms).

Current flow in a circuit is directly proportional to voltage and inversely proportional to resistance or impedance, I=E/R or I=E/Z.

Power, denoted by the symbol P (power in watts), is the product of current and voltage, P=IE. Impedance has a slightly more complex effect on power. See the illustrations.

40dB *Represents a Ratio of About a-hundred-to-one*

60dB *Represents a Ratio of About a-million-to-one*

120dB *Represents a Ratio of About a-trillion-to-one*

The Mysterious Decibel

One of the most confusing terms you'll run across in audio is the decibel (dB). A decibel is a dimensionless measure of the ratio of two powers, equal to ten times the logarithm (to the base 10) of the ratio of the two powers. The decibel scale is logarithmic, not linear.

In simple terms, a decibel doesn't define the amount of something, but rather the logarithmic relationship between two things. If another term is included, like dB SPL or dBV, the relationship is formed between an accepted standard level and the number given.

dB SPL is 20 times the logarithm to the base 10 of the ratio of the pressure of the sound to the reference pressure.

Analogies

Voltage in an electrical circuit is like water pressure behind a dam. Current is like the volume of water flowing through the floodgates of the dam. The gates are like resistance or impedance. Opening the gates increases the flow of water.

Both water pressure and flow volume are required to perform work, like turning a generator. In an electrical circuit, both voltage and current are required to produce power which can perform work like driving a loudspeaker.

Water can be used to generate electricity, which can power a motor that can be used to pump water. Sound waves can be used to generate a signal (in the microphone) that can be amplified and used to drive a loudspeaker that can create sound waves.

Decibels can define the ratio of power-in to power-out in each stage of the process. Perpetual motion will not be achieved in either case. Trust me. APJ

Phono Stage Preamplifiers

by Richard Hardesty

A phono stage must take the tiny output signal from a phono cartridge and raise it to line level while applying substantial equalization. A phono stage must provide high gain and have exceptionally low circuit noise and be well isolated from external sources of noise and interference. These requirements are difficult to achieve.

Any noise, from inside or outside the circuit, will be amplified along with the signal. Low frequency noise, like hum, will be emphasized by equalization. Any error in the equalization curve will profoundly affect the tonal balance and time linearity of the entire system when playing records.

Because of the sensitivity to noise a phono stage must be well shielded from any external source of noise, particularly hum. Radio frequency interference (RFI) can also be a problem because the signal levels at the cartridge are extremely low and the gain is extremely high.

RFI can enter the signal path at the cartridge coils or be picked up by the wires connecting the cartridge to the phono stage. Noise may also originate in the power supply of the phono stage or in another component such as a line stage preamp, especially if the phono stage is mounted inside that component.

Electromagnetic interference (EMI) may affect sensitive phono stage circuits. EMI may originate from a power amplifier or other component positioned near the phono stage.

Amplification

A phono stage amplifies the signal more than any other component and the equalization required to achieve flat response from a vinyl record requires additional gain at low frequencies, which are raised in level during playback.

A moving coil phono cartridge may have output as low as 200µv (.0002 volts). Voltage gain of 60dB or more (at 1kHz) may be required to raise this level to match the input sensitivity of a line stage preamp. That's a lot of gain and it doesn't count the equalization required to boost low frequencies.

Additional gain must be applied (or the insertion loss of a passive network accepted) to raise the low bass frequencies by 18dB or more to compensate for the low frequency roll-off that was applied during the recording process.

Equalization

Phonograph records require substantial equalization for recording and playback. This equalization limits cutter head and stylus excursion at low frequencies and allows the grooves to be placed closer together extending playing time. Equalization raises the level of high frequency signal content relative to surface noise during recording. Surface noise, which remains relatively constant, is then attenuated during playback by equalization that returns the signal to flat response.

The RIAA equalization curve has a turnover frequency of 1kHz where the signal passes without alteration. Below 1kHz, the signal is gradually cut during the recording process and raised during playback. Above 1kHz the opposite occurs: higher frequencies are gradually raised in level for recording and attenuated during playback. The maximum boost and cut at the frequency extremes can exceed 18dB. That's a lot of EQ.

Factors to Consider

By observing some design features you can select the products that are most likely to satisfy your tastes and limit the number of contenders to which you'll have to listen in order to make a final choice. Two factors that consumers can use to classify phono stage preamplifiers for evaluation are the type of device used for gain, and the method of applying equalization and the accuracy of that EQ. Power supplies are important but there is no simple way for the consumer to judge the quality of the power supply or its impact on sound quality.

Each of these factors will impart a sonic characteristic that you may find to be desirable or undesirable. I'll tell you what I hear and you can listen to a few phono stages and develop your own tastes.

The Device Type Used for Gain

Four types of devices are commonly used for gain in phono stage preamps. Some phono stages, like the Emmeline reviewed in this **Journal**, use integrated circuit operational amplifiers (IC op amps). You'll find ICs in most of the inexpensive products (under $500). Better quality components will use discrete devices for gain. These may be bipolar junction transistors (BJTs), field effect transistors (FETs) or vacuum tubes.

IC op amps incorporate substantial negative feedback as part of the design. Phono stages using op amps tend to sound slightly more mechanical and sharp-edged than those using discrete bipolar transistors. Phono stages based on FETs tend to sound slightly smoother and arguably more musical than those using BJTs, but I have heard excellent phono stages based on both BJTs and FETs. Tube phono stages are the most controversial.

Vacuum tubes are excellent voltage amplifiers and, in a tube circuit, electrons flow in an unobstructed path through a vacuum rather than through a semiconductor material. But tubes are inherently noisier than transistors and can be microphonic. Tubes provide a sound that I find to be more musically correct and transparent to the recording but this opinion is not universally shared.

There will be more noise from a tube circuit, but a truly excellent tube phono stage, like the Aesthetix Rhea reviewed in this **Journal**, can provide the musicality of vacuum tubes while approaching the silence of solid-state circuitry. This combination of attributes will usually cost substantially more than solid-state components of comparable quality.

RIAA Equalization

Equalization can be applied actively or passively and can be either accurate, so that the recording is reproduced faithfully, or manipulated to achieve a certain "sound." How equalization is implemented is less important than how accurately it's done.

There are three ways to apply RIAA equalization: 1) all passive, which requires a huge amount of initial gain because so much will be lost in the network; 2) all active, where the feedback loop is utilized; and 3) a combination of active and passive. Arguments can be made for each approach and I've heard good results from all three methods. Accurate results are more important than method, as stated above.

Carriage trade products are often manipulated to achieve a certain "sound."

Playback equalization can be manipulated to provide a certain sound and this occurs frequently in carriage trade products, which are built to sound different in order to separate them from the real high-end components that are designed for accurate reproduction. How do you tell? Compare vinyl playback to CD.

CDs have many faults but they do have flat frequency response. When you audition a phono stage, compare a vinyl record to a CD copy of the same recording. If the tonal balance of the vinyl sounds radically different from the CD there is probably an error in either the cartridge response or phono EQ. Records should sound better but should not have a radically different tonal character. APJ

Line Stage Preamplifiers

by Richard Hardesty

A line stage preamplifier provides input switching, gain and volume control (attenuation) for line level signals like a CD player or tuner. As digital sources have proliferated, the necessity of this vital component in a high-end music replication system has been underrated and its value sometimes maligned.

A line stage preamplifier should precondition and buffer the audio signal and present a clean facsimile, free from out-of-band additions, to the amplifier. The amplifier should be capable of easily raising the audio signal from the preamplifier to a level that is sufficient to drive the loudspeakers without undue compromise. The preamplifier should prepare the signal and minimize out-of-band noise that may cause the amplifier to oscillate or partially rectify noise components.

A preamplifier with a low and consistent output impedance can minimize the effects that interconnect cables and variations in amplifier input impedance may have on the sound.

Do I Need A Preamp?

Logic might lead an inexperienced listener to believe that preamplifiers are expendable in a digital age. Nothing could be farther from the truth. Minimizing components in the signal path is generally a good idea but eliminating the preamplifier seldom provides a sonic improvement.

In my experience a quality preamp is absolutely necessary for satisfying music reproduction in the home. High-end manufacturers around the world agree and are working to provide new state-of-the-art preamp designs that can keep pace with their latest amplifier creations. I could speculate about the reasons why preamplifiers are vital for musical satisfaction or I can simply share my personal experience.

In an attempt to simplify the signal path, I have tried a surprisingly large number of passive devices over the years and I have driven amplifiers directly from all the major surround processors and several of the available CD players with built-in volume controls.

I have driven amplifiers directly from the Wadia 860 and 861 CD players using their digital volume controls. I have used the output from the Levinson #39 CD player with its analog volume control. I have added high quality attenuators directly to various amplifiers, both single-ended and balanced, and driven these amplifiers with CD players. I can state without hesitation that a high-end active preamp sounds better in virtually every case.

I know how to eliminate the preamp or simplify the signal path and I can construct and install switched attenuators. Auditioning the results of these experiments has motivated me to invest a large sum of money in an active preamp for my personal listening pleasure, and after you try all the alternatives I suspect that you will decide to do the same. Even a digital system that includes an active preamplifier simply sounds better than one that doesn't. A lot better.

Can a Surround Sound Processor Replace an Active Preamplifier?

With today's emphasis on surround sound, many consumers have been led astray. They have been told that an expensive receiver or surround sound processor will replace the separate components of yesteryear. If you're happy with Dolby Digital sound quality and you can be satisfied by MP3 music recordings, this is probably true. If you demand more, you'll find that a surround sound processor, no matter how expensive, is an inadequate replacement for a preamplifier. This is explained in greater detail in the article about surround sound processors in this **Journal**.

Surround sound is an exciting addition to an action movie and we use our high-end audio systems to reproduce the left and right channels of film sound. We feed the front channels from a surround sound processor to the pass-through inputs of a high quality preamplifier to accomplish both stereo music reproduction, which is sourced directly from high quality source components, and surround sound effects, which are sourced from DVD discs. See the illustration.

Narrowing the Field of Contenders

There are many line stage preamplifiers available. Listening comparisons will confirm the necessity of utilizing a preamplifier for satisfying music reproduction. Analyzing the specification sheets will help to narrow the field of contenders for listening evaluation. Observing several of the design choices will allow the user to make preliminary decisions about preamplifier sound quality.

Line stage preamplifiers can be categorized by a number of factors besides price: 1) the type of device used for gain, 2) the type and quality of device used for attenuation, 3) the included features such as a phono stage and balanced or single-ended operation, and 4) the quality and capacity of the power supply.

Gain Device

Gain can be achieved through the use of an integrated circuit (usually an op amp) or by the use of a discrete circuit based on the bipolar transistor (BJT), the field-effect transistor (FET), or the vacuum tube.

You'll find integrated circuits in the least expensive preamps and many surround sound processors. The Emmeline products reviewed in this **Journal** use integrated circuits in excellent designs that sound surprisingly good. Op amps usually include negative feedback as a design feature and the circuit designer has no control over the value or quality of individual components, which may vary from chip to chip. Discrete components are better from a high-end music perspective but usually cost more.

Preamplifiers made from discrete bipolar transistor circuits can be top performers. FETs tend to sound slightly more musical with few negative aspects when used in preamplifier circuits. The Ayre preamps reviewed in this **Journal** use FET circuits exclusively with no global feedback. They warm up very slowly and sound completely different after they've been on for several days or even weeks.

Vacuum tubes, in my opinion, can provide the best combination of accuracy and musicality. Some tube circuits, however, stray far over the line between accuracy and colorful representation so caution is necessary—just because it uses tubes doesn't mean that it's good.

The best preamplifiers that I've heard use vacuum tubes for gain in circuits that are designed for accuracy. This **Journal** contains reviews of some of the finest examples available today.

Attenuation

One of the most important jobs performed by a preamplifier is signal attenuation. Attenuation is a fancy way of saying volume or level control but the signal must pass through an attenuator (sometimes two if there is a balance control) and this device can alter the signal irrevocably, whether it attenuates the volume or not. An attenuator can be a simple volume pot (variable resistor) or it can be a method of inserting fixed resistors into the signal path using some type of switch.

A variable resistor (volume pot) is a device that provides continuously variable resistance to signal flow. Individual resistors, which can be matched more precisely, can be inserted in the signal path with a solid-state FET control, which uses transistor switches to select and insert resistors. These resistors may be included on the silicone chip with the control or be discrete external devices. Mechanical relays may be utilized to switch in external resistors, or a mechanical switch may engage these resistors directly. Individual preamps reviewed in this **Journal** utilize all these methods.

The VTL 5.5 has two Alps pots to provide volume and balance control. The Ayre K-1x uses four Shallco switches to insert precision resistors into the signal path for the positive and negative signals of the left and right channels. The Ayre K-5x has a logic-controlled FET switch that inserts fixed resistors into the signal path. The VTL 7.5 uses a logic-controlled device to activate vacuum relays that switch fixed precision resistors into the signal path. The Audio Research preamps have a digitally-controlled FET device that switches in resistors within the volume control chip.

Features

A line stage preamplifier may contain a phono preamp and it may have balanced or single-ended inputs or a combination of both. Balanced inputs don't guarantee balanced operation, which requires redundant circuitry and volume controls. A balanced component won't necessarily sound better than one that is single-ended but the very best sounding components are balanced. See the reviews.

Balanced operation requires attenuation on each leg of each channel. This attenuation must track very accurately and this is difficult to accomplish with variable resistors. True balanced circuits usually utilize sophisticated attenuators. Refer to the reviews.

Remote control is implemented most easily by installing a Crystal volume control chip like you'll find in many surround sound processors, but you'll pay a big sonic price for this convenience. Carefully designed products can provide the convenience of remote control along with uncompromised sound quality. Read the reviews carefully. A sophisticated method of attenuation is necessary in a good-sounding product.

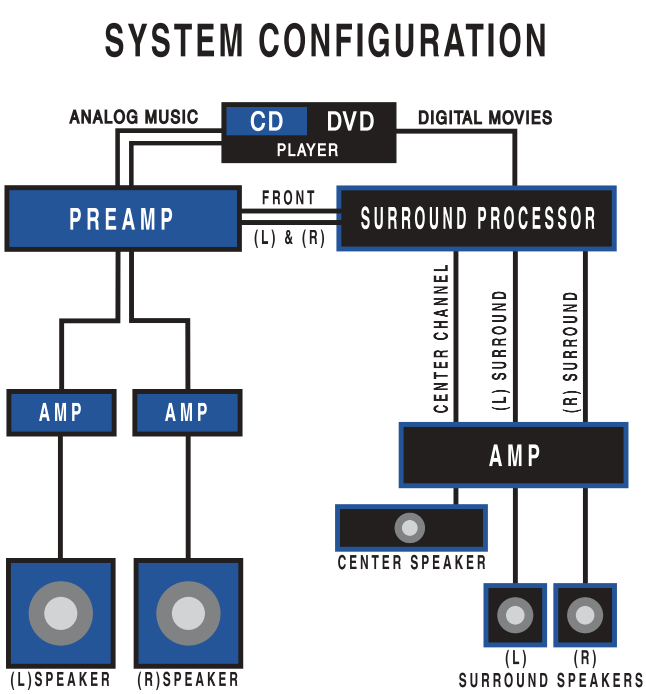
Power Supply

The power supply is vitally important to preamplifier sound but difficult for the technically uninitiated to evaluate. The signal in an amplification device is a modulated version of the power supply so a supply that appears to be much too large is often required in order to provide the purest signal possible. How much is enough? There are no hard and fast rules.

The power supply in the VTL 7.5 is about twice as large as the power supply in the VTL 5.5, for instance. The supply in the 7.5 is mounted in a separate enclosure and connected with multiple umbilical cords. This affects sound quality but I can't offer a simple way for you to evaluate the differences. You'll probably pay more for a better power supply and the added expense will probably provide better sound. This probability is suggested but cannot be guaranteed.

Use Your Ears

If you're trying to achieve musical satisfaction by running your source components through a surround sound processor, borrow a preamplifier and listen to the difference. Shane and I both have outstanding surround sound systems yet we continue to listen to music in stereo through a high-end preamplifier. We connect our systems as described in the block diagram shown in this **Journal**. Stereo music listening assumes the highest priority in our homes and surround sound listening is achieved by including an outboard processor. After some comparison listening, I suspect that you will join us. APJ



Surround Sound Processors

by Richard Hardesty

Surround sound is very popular today and it may provide added excitement for film sound which may increase our involvement with movies. If you are a critical music listener there is a downside to this phenomenon.

The current popularity of surround sound has spawned lots of misinformation. Many have been led to believe that a surround sound processor can become the center of a high-end audio system. Some expect that they can find musical satisfaction when utilizing a surround sound processor instead of a high-end preamplifier to process music signals. I'm sorry to burst this bubble but it's necessary to examine some facts which may paint a different picture. Objective observation will suggest that surround sound is a capability that should be added to an optimized music system, not the other way around.

Many music listeners have been told that a surround sound processor can provide satisfying music reproduction when used as a preamplifier. Reading this article may help to dispel this myth. High-end audio components can be used for many purposes, including surround sound, if the system is configured properly. An optimized surround sound system can't provide satisfying music reproduction but surround sound can be added to an optimized music system without compromise. Keep this distinction in mind as you consider the information provided here.

We at the **Audio Perfectionist Journal** enjoy surround sound and we know how to add that capability to our high-end audio systems without compromising music reproduction. This article should help to explain how an audio system should be configured if the reproduction of music is an important goal.

Processors versus Preamplifiers

A surround sound processor, even a very expensive one, will perform poorly as a preamplifier and with good reason. (See the comparison between the Theta Casablanca, which is perhaps the ultimate surround sound processor, and the line stage preamplifiers reviewed in this issue.) A surround sound processor is actually a combination component that offers highly compromised performance in a number of areas. It is capable of functioning as a preamplifier but is unlikely to provide high-end performance when used instead of a high-end preamplifier. Throwing money at the problem by purchasing a more expensive processor is unlikely to provide a satisfactory solution because a surround sound processor must provide functions and features that are inconsistent with good music reproduction.

> ***"A surround sound processor must provide functions and features that are inconsistent with good music reproduction."***

Placing the processor outside the music signal path is the best course of action but it means that you'll have to include a preamplifier in the system, too. Music should go directly to the preamplifier, and the front channels for surround sound should pass through the preamplifier after they have been decoded by the processor. That's how we do it and that's how every audiophile we know does it. You can use your high performance audio components for other purposes without sacrificing musical satisfaction if you set up the system correctly and refuse to cut corners. I'll try to present that case here.

First I'll describe what a surround sound processor is and then I'll describe what it has to do and what compromises are necessary in order to make it a practical product. When you examine this component realistically you may view its incorporation into a high-end audio system in a different light.

What is a Processor?

A surround sound processor, or digital controller, is a combination device that can, ostensibly, replace many separate devices. A surround sound processor is just like an AV receiver but without the amplifiers which are usually included in a receiver.

A processor could be called the receiver of the new millennium. It is a single component that tries to do the work of many separate components. When an attempt is made to replace several specialized devices with a single device, the replacement may end up being a jack-of-all-trades and master of none. A combination component may perform the basic functions of many separate components but it is unlikely to do a very good job of performing the individual and sometimes conflicting tasks. A surround sound processor is a good example.

A stack of separate components, each optimized for its purpose, would probably provide better performance than a surround sound processor but dealers might perceive a stack of separate components to be a more difficult sale. A processor looks less complicated and intimidating but it's really not.

You've probably been told that you can use a surround sound processor instead of several separate components but discerning individuals will want more. A surround sound processor is often touted as the central or basic component of a system but should actually be used as an auxiliary device if the best music reproduction is desired. A critical look at this device should help to clarify its actual purpose and provide a clearer picture of how it should actually be used.

What's In There

A typical processor includes a powerful digital computer to decode compressed digital audio, an optical character generator to facilitate setup and use, a video switcher to direct various video signals to projectors or monitors, an analog-to-digital converter to digitize analog sources for various processing functions, a digital-to-analog converter to convert digital audio into analog for reproduction, computer-processed bass management to remove bass from smaller speakers and direct it to full-range speakers or subwoofers, delay circuits for each channel to correct for unequal speaker distances, extra attenuators for each channel to correct for improper speaker placement or unequal speaker sensitivities, input attenuators for analog inputs to prevent analog-to-digital overload, a preamplifier of sorts to control audio signals, and probably some multiroom management circuitry to direct various audio signals to different rooms. Many of these components shouldn't be in the same room with a high quality preamplifier let alone in the same box.

In most surround sound processors all these components will share a common power supply and each is likely to provide compromised performance when compared to a high quality separate device that is optimized to perform the same task.

What These Parts Do

You'll need a powerful computer to decode digital formats like Dolby Digital and DTS. The computer can reassemble the data, which is highly compressed and generally has a bit rate similar to MP3, into a semblance of uncompressed multichannel audio. This computer can also provide bass management and various "extras" like adjustable filters and selectable crossover points, additional channels, special surround sound effects like "hall," "stadium" and "nightclub," and adjustable signal delay to each channel.

If the computer is sufficiently powerful to run the Dolby or DTS algorithms—and they all are or they can't get certified—it's powerful enough. More computer power may provide more "features" but it won't provide more accurate decoding of compressed audio. This is a simple concept.

2+2 equals 4 and the world's most powerful and the world's least powerful computers will arrive at the same conclusion (the one that is correct) if they work properly. An audio decompression computer (digital signal processor) must run in "real-time" (it must decode music and dialog as it plays) and they all do. A more powerful computer can't go faster. In fact, a "black box" surround sound processor will decode Dolby Digital or DTS movie soundtracks just as well as an expensive surround sound processor.

An optical character generator makes it possible to choose between a multitude of adjustments and options. Without it, setup and operation would be difficult or impossible. The characters may appear on the front panel of the processor or on the video display; either way they are created inside the component by a device that produces high frequency signals which shouldn't be placed near the audio signal path.

A video signal should never be allowed to corrupt the audio signal, yet surround sound processors frequently have video switching onboard. They usually do a relatively poor job of switching a video signal, which is best accomplished by an outboard, broad bandwidth video switcher designed especially for this purpose, or by the video monitor itself where the video signal will probably have little negative impact on the audio signal.

Most "features" in a surround sound processor are implemented in the digital domain. An analog input could bypass the digital computer that provides most of the features that the customer is paying for. Filters and bass management are usually implemented digitally. Surround sound effects are usually implemented digitally. To utilize these digitally implemented features, an analog input signal must be converted to a digital signal. Analog sources are usually converted to digital with an inexpensive onboard converter in order to provide these conveniences.

Inexpensive analog-to-digital converters can be easily overloaded. This requires that each analog input have a separate attenuator to prevent overloading the converter when a potentially high quality analog signal is converted to what will undoubtedly be a lower quality digital signal.

The device used for attenuation is an important component inside an audio preamplifier while convenience is often the primary requisite for an attenuator in a surround sound processor. A surround sound processor can't have a high quality attenuator like a top preamplifier and it can't have just one per channel—there must be several.

A surround sound processor must have remote control and it must have the ability to balance each channel in level to close tolerances to correct for improper speaker placement or unequal sensitivity. Each channel must have additional trim capability in order to be adjusted within narrow limits. This practically requires a solid-state volume control chip that provides for these conveniences. Each channel must offer signal delay to compensate for speaker positioning and all analog inputs should have signal attenuators to prevent overloading the onboard analog-to-digital converters. This adds up to lots of attenuation and effectively precludes the use of high quality attenuators, important requisites for an audio preamplifier.

An audio preamplifier often has a huge power supply that is dedicated to the reproduction of the audio signal. A surround sound processor usually has a small power supply that is shared between all the components that a processor includes.

> ***"If you use a surround sound processor instead of a preamplifier to reproduce music, you're unlikely to be satisfied with the results."***

If you want to decode Dolby Digital and DTS film soundtracks you need a surround sound processor. If you use a surround sound processor instead of a preamplifier to reproduce music, you're unlikely to be satisfied with the results.

Music and Movies Without Compromise

How can the discerning listener enjoy surround sound for movies while utilizing the sophisticated audio components that reside in a high-end audio system? By first assembling an accurate 2-channel music system and then adding a surround sound processor as an external device, outside the 2-channel signal path which provides satisfying music reproduction. All you have to do is run the front left and right channels from the surround sound processor through a pass-through or unity gain input on the preamplifier. Most of today's better preamplifiers make this very easy. Look at the illustrations to see how we incorporate surround sound into our high-end audio systems without compromising music reproduction. APJ

an interview with...

Charles Hansen

by Shane Buettner

One of the greatest benefits of working in the hi-fi industry is meeting the talented and interesting people who design the products that result in so much musical satisfaction for so many music lovers. High-end audio companies often represent the vision of one person, and that person's personality is ingrained in every component of his/her company's products. Charles Hansen, who started Avalon and is now president of Ayre Acoustics, embodies all of that. Unlike the many designers who come from an electrical engineering background, Charlie is educated in physics. From my first conversations with him I've been struck by his candor as well as his approach and philosophies regarding audio. If something sounds better to him, he goes with it regardless of what the prevailing wisdom or measurements might say. His designs feature unique power supply innovations, truly balanced topologies with discrete circuits used wherever possible, and zero global negative feedback. I first got to know Charlie after getting hooked on Theta's Dreadnaught power amplifier, on which he did the basic design work and that I've been using for a reference for two years. In addition to their highly regarded preamplifiers and power amplifiers, Ayre's D-1x is one of the finest CD/DVD players I've auditioned. I started this interview by asking Charlie how he got interested in this shared passion we call hi-fi...

I've been around hi-fi systems as long as I can remember. My father is a real music lover and he had Dyna electronics built from kits in the early '60s. I was fascinated with his system, but he wouldn't let me touch it because I was only about six years old! He gave us kids his old mono record player in a stand-up cabinet with an Eico amplifier.

By junior high school I had taken over the Eico and was always tinkering with it. I would even cut classes and hang out in the school library reading books like *Elements of Radio* by Marcus and Marcus—a great introductory text, if you can still find it. That led to a job in the repair department of the local stereo shop by the time I started high school. I was too young to drive so I had to hitchhike there. The shop had McIntosh and JBL to begin with, and when the high-end began we were right there with Audio Research, Magneplanar, Linn, and Levinson. A group of us were really into the new frontier that high-end represented, and we spent a lot of time listening to music and building home-brew projects. (Tommy Thompson and Charles Roe, if you're still out there, give me a holler!)

Then you studied physics in college?

Studying physics was a good thing and a bad thing. On the one hand, I learned the fundamentals about things and how to analyze things in a rigorous way. On the other hand, we don't really know that much about how the universe works. We have equations that describe gravitational forces, and we can predict the motion of bodies with incredible accuracy, but nobody knows what gravity really is or how it really works. In the physics department, it's easy to develop a certain intellectual arrogance that can be a real impediment when it comes to understanding hi-fi equipment. For example, I doubt many physics professors would accept the fact that interconnect cables have any impact on sound quality!

How did you get from physics back to audio?

After I graduated, I had worked out some seemingly simple theories of the behavior of speaker driver diaphragms that led to a real breakthrough in the ability to achieve transparency and freedom from coloration, from top to bottom. I started Avalon Acoustics with a woodworker friend, Bob Grupp. We didn't know much about the realities of running a business, but luckily there were some key people that kept the company alive along the way, including Richard Broida, Jeff Rowland, and finally Neil Patel. I learned an incredible amount about the processes of both listening and designing from Neil, which helped enormously when I started doing electronics at Ayre.

How did Ayre Acoustics come about?

I started Ayre with a couple of friends, Peter Bohacek and Katie Lehr, in 1993. The goal was to bring that "magic" level of performance we had achieved at Avalon down to a more reasonable price level. You know how it is when your system is really singing, and you play it for other people? Whether they're audiophiles or not, the reaction is usually, "Wow, I didn't know a stereo system could sound like that! I can hear so much more of the music!" And then at some point they ask the price, you tell them, and they just say, "Oh, well. It sounds great but I can't afford that."

While it's great to build an all-out "statement" sort of product—and you learn a hell of a lot when you do that—there are just a few lucky souls that will get to own and enjoy that design. We're going to keep building statement products, but at the same time we're going to keep transferring that knowledge to our less expensive designs. We really want to broaden our audience and bring a different kind of musical experience to more people.

What things have you learned from the state-of-the-art products you've designed that have trickled down to your more affordable products?

We've learned a lot about how to achieve a really musical sound with virtually any type of music. There is a lot of gear out there that can shine with one type of music, but really falls apart with another. For instance, it's tough to beat a good single-ended triode for sheer beauty with chamber orchestra or string quartet. But put on a rocking, dynamic piece and the music tends to just sit there without coming to life. Some gear will sound fantastic with a reference-quality recording, but really nasty with an average recording. Some gear makes everything sound "good" because it colors the sound, sweetening everything.

With Ayre equipment, when you put on a great recording, of course it sounds great. But even when you put on an average recording you'll say, "The recording isn't all that great, but listen to how beautiful those vocal harmonies are!" The equipment literally gets out of the way. This year we launched a new series of products that represent a real breakthrough. In the two-channel lineup, we have the AX-7 integrated amplifier and the CX-7

CD player (reviewed in **APJ** #9, pg. 37), and they're both stunning. They'll hold their own against just about anything at any price. When you talk about sheer musical enjoyment, when the system just gets out of the way and the music comes alive, they're right up there with the best.

Tell me a bit about your design philosophies. Many designers feel that judicious use of global feedback is beneficial, but your products are zero global feedback. Why?

In my experience, the most musically natural sound comes from circuits without feedback. Feedback was invented by Harold Black in the 1920s for the telephone company. There were repeater amplifiers along the telephone lines that would

Ayre K-1x

boost the signal to make up for the losses in the long wires. It was very important to have exact gain matching between these amplifiers, and this was extremely hard to achieve due to the variations in the amplifying tubes. By applying feedback to the amplifier circuits, the gain was now set by resistor values rather than by the tube characteristics, resulting in improved gain matching. This is the real reason negative feedback was developed.

Feedback also happens to improve the measurable characteristics of amplifier circuits by increasing the bandwidth, reducing the distortion (at least for steady-state signals), and lowering the output impedance. And this is why its use became so universal in audio circuits. For decades, it was assumed that there was a direct correlation between measurements and sound quality. But there is something about feedback that we can't really measure that seems to have a detrimental effect on the sound quality of real musical instruments. It seems to be related to time domain performance.

Time domain performance? Can you elaborate?

Well, the simplest way to think about this is that feedback cannot respond to an error until after the error has occurred. And this correlates pretty well with what we hear. Compared to a zero-feedback design, adding feedback seems to overemphasize the leading edge of transients. This can give a more "spectacular" sound in the hi-fi sense, but it is less musically natural than a zero-feedback design. And this sonic "thumbprint" seems to exist under a wide variety of conditions. On the other hand, a zero-feedback design becomes very chameleon-like from a sonic standpoint. In my experience, the zero-feedback designs will "get out of the way" of the music with a wide variety of source material. And with our designs there isn't any reason to use feedback. The Ayre V-5 power amplifier has a bandwidth of around 200kHz, distortion of around 0.1% at 100 watts, and an output impedance of around 0.2 ohms. This level of measured performance is unprecedented for a zero-feedback design. There's really no reason to use feedback when we can achieve such numbers without it.

You also employ discrete circuits wherever possible in your statement products. Tell me more about this.

Using discrete circuits versus integrated circuits is like making a cake from scratch versus making it from a mix. Clearly, it's easier and cheaper to make a cake from a mix, and you don't have to know that much about cooking to do it. But I don't really think Betty Crocker knows more about making cakes than a great French dessert chef! It's a similar situation with audio circuits. Discrete circuits offer a greater opportunity to use the highest quality components, in any conceivable configuration, and adjust all of the parameters to obtain the highest possible performance for a specific application. This is the only way to go if you are making a true state-of-the-art product.

How do you approach that kind of sound quality without the benefit of discrete circuits in your AX-7 integrated amp and CX-7 CD player?

We are now using ICs (integrated circuits) in some of our less expensive products designed by Dr. Barrie Gilbert, one of the "godfathers" of analog circuit design. Now, it's very important to differentiate between the terms "integrated circuit" and "op amp" (operational amplifier). We're using integrated circuits, which simply means that there are many transistors created at once on a single piece of silicon. An op amp, on the other hand, uses negative feedback as its fundamental operating principle. You cannot use an op amp in a zero-feedback circuit. The topology of the IC we use in the less expensive products is very similar to the topology of our discrete circuits. We've figured out how to implement these monolithic ICs with zero feedback. By using two of them together, and modifying their characteristics, we can get within spitting distance of the performance of our discrete circuits at a much lower cost. The critical advantage here is that the monolithic design of the IC means that all of the transistors are matched extremely closely. In contrast, with our discrete circuits we have to spend a lot of time measuring, sorting, and matching transistors, which translates to a more expensive final product. By using an IC in a way that hasn't been done before, we're able to achieve a real breakthrough in performance at a real-world price point. It's great to make an all-out assault on the state of the art, when the only limit is your imagination. But the real challenge is to bring those lessons back to the real world where more people can enjoy the fruits of our labors.

Your power supply technology incorporates inductive filtering, which I'm starting to see on more and more products these days. How does this improve sound quality?

Well, you've got to remember that there are two different ways to use filter inductors in a power supply. The first way is before the initial filter capacitor, immediately after the rectifiers. The second way is after the initial filter capacitor. The first method is called an "inductor-input filter" or a "choke-input filter" (with the emphasis on "input"). The second method is called "choke filtering" or "inductor filtering." If you look closely at the advertisements, you'll see that most companies offering chokes are actually using "choke filtering," which is much less expensive than "choke-input filtering."

Both methods will reduce high-frequency noise coming in on the AC line, which is a good thing from a sonic standpoint. But only the inductor-input filter will produce a continuous charging current to the initial filter capacitor because the energy stored in the magnetic field of the inductor is slowly released into the capacitors. Any other method results in sharp pulses of charging current to the capacitors. An inductor-input filter provides a much purer source of DC than does a conventional supply, which results in improved sound quality. The problem is that the cost of the supply is roughly doubled, which really only makes it applicable to cost-no-object designs.

When we started working on our new product line, we knew from the beginning that we wouldn't include a true inductor-input filter. We decided instead to develop a new circuit that would be immune to any noise induced into the circuitry from the power supply. The result was our "current-mirror amplifier." This circuit has around 1000 times (or 60dB) greater rejection of noise from the power supplies.

Do you use any other techniques to address noise on the AC line?

Yes, there is another extremely important element to achieving the best results. We used to use ferrite-based filters to clean up the incoming AC power. They do a really good job because they don't try to block the RF energy or shunt it to ground, which can't really be done in an effective manner. Instead, they literally absorb RF and turn it into heat. The musical result is a blacker background, reduced grain and hash, and sweeter high frequencies. Unfortunately, ferrite filters become magnetized over time. I don't fully understand the mechanism involved, but you can easily hear the sonic effects—the sound becomes hard and glassy, with squashed dynamics. If you demagnetize the ferrite, it will sound good again, but only for a few days. Then it becomes magnetized again. To solve this problem, we developed a filter—called the Ayre Conditioner—that absorbs RF energy in the same way as a ferrite, but it is completely non-magnetic. Also, it operates in parallel with the AC line, so there is no restriction of current flow whatsoever. Normally with power line conditioners, there are sonic tradeoffs. Some things (like treble purity) improve while other things (typically dynamic impact and bass authority) get worse. The Ayre Conditioner is the only filter we've found that does only good things to the sound.

*Switching gears, let's talk about digital sound. "Upsampling" is being referred to by some in the industry as a "magic bullet" that improves playback of the entire existing CD collection. What would you want **Journal** readers to think about with respect to this topic?*

First, "upsampling" is a marketing term, not a technical term. From a technical standpoint, there is absolutely no difference between upsampling and oversampling. Thirty years ago, the marketing fad was low distortion. Now it is upsampling. When manufacturers talk about upsampling, they are referring to the fact that there are two separate digital filters being used. Since changing anything will change the sound of a component, it is no surprise that adding another digital filter will also change the sound of a component. However, you could achieve the exact same results by using one digital filter that had the characteristics of the composite of the upsampling and existing digital filters.

In other words, from a technical standpoint there is absolutely nothing new about upsampling. Even the idea of cascading digital filters is old hat—virtually all digital filters are a cascade of 2x stages. For example, an 8x interpolation (oversampling) filter comprises three 2x stages in cascade to achieve an overall rate of 8x. There are some "upsamplers" that are realized in external boxes. In this case, you have to remember that a significant portion of the sonic impact of such a device comes from the fact that the jitter spectrum will always be modified. None of this should be taken to mean that upsampling players sound bad. On the contrary, improving a digital filter can improve the sound of a digital player. It's just that there is a lot of misleading marketing going on out there. You should always buy a component because it sounds good to you in your system, and not because of the marketing claims behind it.

Some people have been drawing a distinction between "upsampling" and "oversampling" based on whether the product also interpolates the word length as well as the sample rate, i.e., converting 16-bit signals to 24 bits.

This is total B.S. All digital filters work by multiplying the data by coefficients. The way that you make a 16-bit word into a 24-bit word is simply to multiply it by an 8-bit coefficient. Normally, the coefficients are much longer than 8 bits. For example, if you multiplied 16-bit data by a 24-bit coefficient, you would end up with a 40-bit word. This then must be rounded (or truncated) to get back to 24-bit data at the output of the digital filter. Any currently available digital filter will "interpolate" 16-bit input data to 24 bits at the output. But no matter what you do, when you start with 16-bit data, you will only have 16 bits of precision in the output. Even if you multiplied by 1000-bit coefficients, increasing the actual resolution of the original signal is impossible.

*Charlie, your Ayre D-1x is one of the best sounding CD players I've ever heard. (Reviewed in **APJ** #9, pg. 33.) In addition to the D-1x, you've introduced a new CD player—the CX-7—at a time when many other high-end companies are scrambling to release DVD-A and/or SACD players. Why?*

First of all, thanks for the kind words on the Ayre D-1x. We spent over two years on that project, and we're very happy with the way it turned out. The fact that our first digital product was so well received gave us a lot of credibility in this arena.

As you mentioned, we've just released our second digital product, the Ayre CX-7, and the reaction has been overwhelmingly enthusiastic. We chose to make it a CD-only player simply because that is the most cost-effective route to enjoying music. At this point in time, adding support for SACD or DVD-Audio sharply increases the cost of a player. Given the limited amount of software titles that are available, paying thousands of dollars extra for the capability to play a handful of titles doesn't make a lot of sense for most people. Unfortunately, given the economic realities facing the software producers, this situation isn't about to change anytime soon. Frankly, I have my doubts about the viability of either of the new formats as anything other than audiophile niche products. We're still looking at the possibility of making a universal player, but until we can offer something that makes sense (at least on some level), we're going to hold off.

But even if DVD-A and/or SACD were to survive as a niche format, aren't high-end audio and companies like Ayre Acoustics where that niche will be? In other words, aren't your customers the kinds of people who will be looking for a high-resolution player?

Well, you've got to remember that Ayre has been at the forefront of these new technologies from the very beginning, being one of the very first high-end companies to offer a DVD player. And to tell the truth, we thought there would be a flood of other companies following in our footsteps, but that simply has not happened. Only a handful of high-end companies offer DVD players, and some of them have already dropped out of the game. The reality is that it is extremely difficult to implement these technologies in a meaningful, viable way. It's like they say, "If it were easy, everyone would be doing it!"

So, yeah, our customers would love to buy a "universal" player that sounds great, looks great, and costs $5,000. But when they find out that they would be lucky if a $5,000 universal player would sound as good as a $2,000 CD-only player, then they're not so sure that's what they really want. At the same time, they're afraid to spend any significant amount of money on a player that won't play all of the new so-called formats. (I say "so-called" because it's not clear to me that a few dozen titles, or even a few hundred titles, really constitute a format.) They're afraid that if their new machine won't play all the new "formats" that they'll get stuck with an expensive boat anchor!

A big part of the problem lies with the way the magazines have been reporting on these new "formats." They've created completely false expectations in the minds of consumers—hype about “universal” players and how they will solve all of the consumers' problems. For instance, every time a new multichannel DAC chip is released, there is some article proclaiming how the "universal" players are right around the corner. This is completely erroneous. It's like saying, "I can get a carburetor in the mail from JC Whitney, so it'll be really easy to build a new car from scratch."

Now, if you want to build a true universal player, you've got to include multichannel capabilities. And I have yet to see a single article on multichannel sound that addresses the obvious issue of quantity versus quality. In my personal system, I've got a very enjoyable setup. The speakers cost $8,000 a pair, the amp costs around $4,000, the preamp is another $8,000, and the digital front end is around $8,000 or more. Throw in $6,000 of cabling and the grand total is over $30,000. It sounds great, and I love listening to music through it. But let's say that I wanted to go multichannel. I would have two choices. Either I could sell everything off and spend an equivalent amount on a multichannel system, or I could add on to my existing system. Well, assuming that I want to maintain the same performance level, the latter proposal borders on the preposterous. Right off the bat we're looking at another $30,000 for amps, speakers, and cabling. And for multichannel equipment, there isn't anything made that will even come close to the performance of my preamp and player, regardless of price. And that assumes that I even have space for all of that equipment in my house, which I don't!

The other choice would be to downgrade and keep the overall investment the same. Is there anyone out there that can look me in the eye and, with a straight face, tell me that I'll be happy going from two $4,000 speakers to six $1,200 speakers? I don't think so. I remember when I owned $1,200 speakers and, frankly, I'm much more interested in upgrading my speakers than downgrading them. And it's still not clear if either format will survive. Some people are saying they're already dead. Without thousands, or better yet tens of thousands, of titles, I don't think you've really got much. The problem is there's not really any incentive for the software companies to make new titles. So like I said, we'll keep looking at it. When we can offer something that makes sense, we'll do it. But between the direction the Japanese majors are going with the new formats and the expectations that have been created by the magazines, we manufacturers have got our work cut out for us. But, hey, that's our job, so I can't complain too much! Besides, I've got this idea for a player that might be just the ticket…

*Charlie, thanks for taking the time! I think you've given **Journal** readers a lot of interesting things to think about!* APJ

an interview with…

Luke Manley

by Richard Hardesty

VTL (Vacuum Tube Logic) was founded in the US in 1986 by David and Luke Manley. In 1993 the original company, which was involved in diversified endeavors, was divided and the father-and-son business partnership dissolved.

David Manley, a well-known recording engineer and designer, left VTL to concentrate on music recording and the development of professional studio electronics, which are sold under the Manley brand name. Luke Manley and wife, Beatrice Lam, took over the VTL name and inventory and refocused the company on a single pursuit—developing and efficiently manufacturing vacuum tube amplification components for high-end home music systems.

A decade of innovation and development has enabled the continuous refinement of existing products in the VTL line and produced new products, like the TL 7.5 Reference preamplifier and Siegfried Reference 800-watt monoblock amplifiers, designed to advance the state of the art in consumer audio. The VTL manufacturing plant builds components using techniques which have been developed to produce exceptional reliability and consistent unit-to-unit performance.

I visited Luke and Bea at the VTL factory in Chino, California, on a hot day in early summer 2003 to learn about these new products and see how they're made. Bea showed me the facility and the team that proudly builds each component and Luke and I discussed the high-end audio industry and the future of vacuum tube electronics in general and VTL products in particular. Here's an edited version of our conversation.

Luke, let's start with some background information about you and the history of your company. When and how did you become interested in music and why did you decide to enter the business of manufacturing high-end home audio components?

My father has been a recording engineer as long as I can remember, so music was always part of my life. There was music in our home on the hi-fi system, and he took us to live classical music performances starting when I was young. Piano lessons as a kid made me realize early on that I can't play a lick but, being around studios and equipment doing wiring and speaker projects with my father, I learned how to solder pretty well.

When I went to college I studied commerce, figuring that I would apply my skills to a business that interested me. I'm a good problem solver, and I knew that I wanted to be an entrepreneur—I guess that's the great Chinese candle, as my Dad has always been self-employed.

Anyway, Dad started the original VTL company in the UK, and after I heard the early VTL amplifiers at the Chicago CES in 1986 I joined him in the audio business. We decided to move the manufacture over here to keep VTL competitive in this market, this being by far the largest market in the world for this type of equipment.

Although I started at VTL by developing and running the business operations, it wasn't long after that I had to develop the QC process for the company, which is when I learned how the VTL designs worked, and what was going wrong when they didn't.

Dad did the design work and taught me about parts buying and finished product sales, and I pretty quickly learned how the audio business worked, as I dealt with customers and handled repairs. I never worked on the assembly line, so I'm definitely not as good at soldering as our production people are.

You've stated that music is an important part of your life. Expand on that.

I have always been very keen on live music, and of course I enjoy my audio system at home. Bea and I are extremely fortunate to be able to have access to live music in both Northern and Southern California. In the LA area where the factory is located we regularly go to Catalina's Jazz Club where, if you sit forward of the sound reinforcement system, there is a real natural sound to the instruments that gives a good perspective on how things should sound. We also often catch performances in Orange County that we wouldn't normally be able to see in the Bay Area, where we live.

When I took over the company in 1993 I moved up to the San Francisco Bay Area, where I met my soul mate Bea, who is an avid music lover and audiophile. When we are at home on the weekends we go to the San Francisco Symphony and opera performances regularly, and we often take a trip out to New York to hear a particularly rare performance that we might not be able to catch on the West Coast.

When you and your father were partners VTL was also involved in music recording and the design of professional components for the recording industry. Did this dilute VTL's commitment to high-end audio?

Yes, definitely. For one thing, with multiple product categories engineering gets spread really thin trying to work with different disciplines and product specs and requirements, and it is difficult to excel at any one thing.

For an example of this one might look back to the early VTL products that came from different models, similar in design, with pretty good sound but quite inadequate user interfaces, and honestly, not super high reliability.

It's hard enough just to keep on top of one engineering discipline, let alone multiple and completely different businesses. The notion of what is acceptable design practice in the pro studio market is totally different from the high-end high performance playback market. There are vastly different requirements for products that are in the sound creation/recording domain, where coloration is often part of the creative process, than in the playback domain, where coloration is generally not considered desirable.

Even analog and digital are two very different engineering disciplines, with somewhat different test equipment and principles.

This is the main reason that VTL does not make digital products, and probably won't in the foreseeable future.

The record business is a totally different business again, with completely different customers and distribution channels, and when Dad and I split up it was a really good decision to spin the pro studio and record divisions off. Otherwise VTL would never have gotten to where we are today, with the level of sophistication that the market has achieved and that is required of products today.

You can look at our new TL 7.5 line preamplifier and Siegfried power amplifiers to see what I mean when I say "sophistication." In my opinion we never could have developed these products if we had continued trying to do so many other different things.

Some people think vacuum tubes represent old-fashioned or obsolete technology. I'm sure that this question has been asked many times but, in this remote-control, networked digital age, why do you continue to champion vacuum tube amplification for home music reproduction?

While it's true that in some circuits tubes would represent old-fashioned technology—in power supplies, for example—it is quite clear to me that for a number of solid technical reasons vacuum tubes simply sound best when used within their capabilities, and when used in the right places. And sound quality should surely be the raison d'etre of any company in the high-end audio business.

The major technical reasons that tubes sound better is that tubes are very linear voltage amplifiers (while transistors are not), and tube components usually use far simpler circuits and need very little error correction, also known as negative feedback. This translates to both purer sound and a closer emotional connection to music.

Most people get this when they listen to our products, but then they ask me, "Why now? How come tubes didn't sound like this in the old days?" I respond that in the old days modern components that allowed the designer to get the most out of tubes were not available.

Examples of the modern components available today that weren't available when tubes were in their heyday are the solid-state power supply components. The rectifiers available now are designed to address a far larger capacitor than the old tube rectifiers could, and they don't age as tube rectifiers do, so we get stiffer and more reliable power supplies, which directly affect the sonic capabilities of the tubes in the amplifier stages.

So with modern components we are now getting far more from tubes, and with modern circuits and modern test equipment we are making tubes even more viable today than ever before, while engineering out the known typical drawbacks of tubes.

I can tell you that there's nothing old-fashioned about our latest products, especially the TL 7.5 preamp and the Siegfried amplifiers. I truly believe that they represent the latest in amp and preamp design thinking.

Some proponents claim that a tube is a more linear device than a transistor. Others say that audiophiles who prefer tubes are seeking euphonic coloration rather than accuracy. Would you care to weigh in on this controversy?

As I mentioned, tubes are known to be superior (more linear) voltage amplifiers and, therefore, need less error correction (negative feedback) than transistors generally do. To further this point, I have found that audiophiles who prefer tubes are typically real music lovers and, in my experience, musically knowledgeable listeners demand accuracy and full frequency response.

However, another reason why tubes sound the way they do is that tubes have quite a different distortion characteristic and tend to go into overload more gracefully. Generally speaking, the onset of distortion when clipping from overdrive is quite different in tubes than in transistors. Transistors typically produce high-order harmonics under such conditions, while tubes produce low-order harmonics, which, even though the acceptable distortion figures are typically higher for tubes, seem to be benign to the human ear.

The gentle onset of overload distortion and the low-order harmonics seem to yield more relaxing listening sessions for tube users.

Having said that though, I wouldn't say that all tubed equipment is necessarily accurate. In my opinion a well-designed circuit does not rely on euphony, unless euphony is intentionally made part of the design for some kind of retro sound. There is plenty of equipment out there being manufactured today that is serving the retro sound of yesteryear, and certainly such products have their following, but VTL has never catered to that crowd. Since Bea and I regularly go to live music performances our goal at VTL is to make our equipment reproduce as accurately as possible what we hear in the live venue.

Do you advocate the use of tubes for all circuits or do you find that solid-state devices perform better for some purposes?

No, we don't use tubes for all of our circuits, as we prefer to be able to make a choice and use the best component for the particular application. For example, we have historically always used solid-state rectifiers rather than any tube components in our power supplies because transistors are very linear current amplifiers, and they don't change much as they age, so we can use them to their best advantage for reliability in power supplies.

We are presently expanding our use of transistors in the power supply area, with the new Siegfried amplifiers coming out this year, and also in the R&D work we are doing currently in power-factor-corrected power supplies.

As far as signal path components go, transistors are also very good as true constant current sources (approaching the ideal of infinite impedance and infinite voltage) and for their capabilities with impedance matching, for which tubes are not as well suited.

However, we always use tubes in all our gain stages because, as I mentioned before, tubes are such linear voltage amplifiers. We have found that we can always get the best sound that way.

In our more recent designs—like our new TL 7.5 Reference line preamplifier—we use transistors in the output stage buffer, which I feel contributes greatly to neutrality and predictable performance into a wide variety of loads.

However, there are other issues that crop up in such designs, and the trick is to figure out how to match MOSFET transistors, with their high gate capacitances, to the high output impedance of tubes, without high frequency roll off, and one has to take care to match these components together carefully.

Those who have never owned tube equipment are often worried that tube amplification components will be unreliable or require lots of maintenance. Are these realistic concerns?

Preamp tubes usually require very little maintenance and, at their low-running currents, they should last well over 10,000 hours. This might be an easy way for users who are new to tubes to get a taste of the tube sound: Start with a tube preamp, which will mean little or no maintenance, and which will bring noticeable sonic improvement to a solid-state amplifier.

Power tubes certainly will eventually require replacement as they do wear out, but how quickly they wear out depends on the equipment design and the usage. For instance, tubes will wear out faster if the amplifier is played consistently at high levels. But the worst that can happen with worn out tubes is that the amplifier will sound like it doesn't have the snap in the top end or the bass punch that it used to when the tubes were new. Replacing the tubes completely rejuvenates the amplifier and the sound quality.

In all the VTL amplifier designs the output tubes are biased for very low current, which extends tube life under idle and at low power. And in our newest designs we are extending tube life by optimizing the operating points of the output tubes and warming up the tubes slowly, as well as providing very low trickle current ever-on positions.

As far as failure modes go, I must point out that any circuit not properly designed can be unreliable, whether designed with tubes or transistors. Unfortunately, when transistors fail they tend to take everything else out on the board, so repair can be very expensive.

This is not necessarily so with tubes. By definition tubes are modular and can be easily replaced!

Seriously, though, tubes are very rugged and can stand adverse conditions far longer and recover far quicker than solid-state components can. Tubes are also very forgiving with circuit tolerances, and so they tend to be in very simple circuits, which then have the benefit of less to go wrong, and the amplifier is far simpler to repair if something inconvenient does happen.

There are some very simple things one can do to get maximum protection should a tube fail, so that the major problem then becomes one of replacement to maintain sound performance when they wear out. For instance, one can keep high voltages and currents only on highly insulated Teflon wiring, rather than running it on circuit traces, and one can use up-rated components on the board to handle surges and high-tension fuses to open before any damage is done.

However, we are cognizant of the fact that people don't want to have to know a lot about maintaining a tube amp, and our new Siegfried amplifier eliminates most maintenance questions, with fully logic-controlled auto bias to keep the tubes in optimal operating condition all the time, fault sensing for protection if a tube does fail and diagnostics for estimating tube life, with service reminders to alert users of the need to check and replace worn tubes.

Your TL 7.5 preamp contains just two inexpensive vacuum tubes. Your Siegfried amplifiers have auto bias and a complete system of fault detection and protection, all of which makes them very user-friendly. Will some of this technology trickle down to the more affordable products in your line?

The main reason for starting with an expensive design is to have the budget to solve problems, and then the challenge becomes figuring out how to make the same technology available in less expensive products.

For instance, both the TL 7.5 and the Siegfried were 5-man 5-year projects—each. Just to give an idea of the R&D costs involved in these types of projects…

Certainly we will leverage as much of the huge investment we have made into lower-priced products in the line, but this will take some time, and it will probably raise the cost of the models that get the new technology somewhat.

Components that combine FETs with vacuum tubes, utilizing microprocessors and numerical displays, must be designed by a team of engineers from various disciplines. How did you bring these advanced product ideas to fruition?

It is pretty different now from the days when Dad did all the engineering. The market for high-performance products has become far more demanding of performance, while consumers expect to interact far less with the equipment. At VTL we have to have five engineers designing new products, each with quite different skill sets.

We have one engineer working for us who has seen and serviced just about every piece of test gear ever made, and who specializes in tube differential circuits. He nuts out the basic audio circuits based upon the design spec that we are starting with.

Another of our engineers specializes in switching power supplies using MOSFETs and complex interleaved magnetics, and from this we have the capability to completely understand the power supplies in all the VTL products and their effect on the sound of the design.

Two other engineers are responsible for all the control systems that keep the circuits operating properly and the software that controls the hardware to do that. And the fifth engineer specializes in PC board layouts and mechanical engineering (metal packaging and industrial design).

However, I have to point out that even with all the engineering and sophisticated test equipment that we use, we still have to rely on our ears to verify that what we see on the bench actually correlates with what we hear in the concert hall, and at VTL it has become our specialty to reconcile the two. As I mentioned, Bea, who is actually also an EE, and I regularly attend live music concerts, and one of the engineers here is also a very good listener, so between the three of us we are able to do the final voicing after a new design leaves the bench.

Between these five members of the VTL design team the products get designed, spec'd and tested so that they are ready for the market, which is quite different from the way we did it in the old days when my father was the sole engineer and critical listener.

Single-ended triode amplifiers and transformerless tube amplifiers are fashionable these days but your company hasn't jumped on either bandwagon. Why?

I would think that with single-ended triode amplifiers it is technically unfeasible to get full frequency response because the output transformer saturates, and I feel that midrange-only tends to sound rather boring after a while.

Regarding transfomerless tube amplifiers, since tubes are voltage amplifiers and are not ideal current amplifiers, and because of the high output impedance of power tubes, OTLs are not equipped to efficiently couple into the low impedances of speakers, which makes it hard to get good bass performance into real-world loads.

VTL is not fashion-oriented, and we require full frequency response into real-world loads, so SETs and OTLs are pretty much ruled out for what our customers demand.

Is it possible to make a really quiet tube preamp? How about a high-gain phono stage?

It's true that, with their higher impedances, tubes can be more susceptible to noise, but in that respect I believe our current line preamps are pretty quiet. The TL 2.5, 5.5 and 7.5 are all about –110dB, and the differential input of 7.5 offers increased CMR and lowered signal-to-noise with the extra 6dB of gain that comes from balanced operation.

There is always some gain noise in high gain circuits, which, depending upon the particular design, can come equally from tubes or transistors, and which shows up as a soft high frequency hiss. But how much of this noise is transmitted into the room depends upon the sensitivity of the amplifiers and the speakers.

Tubes are desirable in phono stages, which are the highest gain circuits, but they have strengths (linear voltage gain) and weaknesses (noise and susceptibility to interference). Even though tubes can be made to be quiet, people have different tolerances for noise level and the noise output depends heavily upon the rest of the system.

One would think that step-up transformers might have much to offer in this application, but we have found that they are not the way to go because of their unpredictable reactance to transmitted capacitances. They also sound artificial. Audiophiles know this and usually aren't very accepting of input coupling transformers.

At VTL we have an entry level TP 2.5 standalone phono stage in our Performance series that has about 60dB of gain, which means it can take moving coil cartridges down to about 0.5mV before the noise level becomes too high. And we have an internal phono stage for the TL 5.5 that can take cartridges down to 0.1mV, and that will likely be made into a Signature standalone phono stage in the near future.

We have been working on a Reference level phono stage, but when we introduce a new product in the Reference series it has to be able to address a need that is currently not being addressed in any other product.

It seems that some sort of hybrid circuit might be the way to go for a phono stage, but we are still looking at the best way to achieve this and still get the best of both worlds. So our big phono stage is still some years away.

Negative feedback has a significant sonic impact on solid-state amplification components. What is your position on negative feedback and how is it used in your tube products?

At VTL our position on negative feedback is that it is useful and even desirable, but in small amounts. It's kind of like spice in cooking—if you add too much the spice overpowers the meal.

The reason solid-state gain stages need so much negative feedback is that they typically have a lot of gain, and they also are not very linear voltage amplifiers, so they need lots of error correction, or NFB. The problem is that, while high NFB circuits yield measurements that are really good, negative feedback tends to cause sonic and emotional disconnects and products with high NFB sound flat and lifeless.

Because tubes are such linear voltage amplifiers and because we use only tubes in all our gain stages, extremely low amounts of NFB are used in VTL components, and the exact amount is determined by listening. VTL designs typically use very low amounts of NFB: under 5dB in preamplifiers and around 10dB in power amplifiers to help handle complex loudspeaker loads.

What are you developing now and what can we expect to see from VTL in the future?

We're working on getting the Siegfried into production, for shipping release September 1st. After that we'll have a stereo version of Siegfried in a tower chassis at 300 watts per channel, as well as a one-box version of the TL 7.5 line preamp, which will be priced between the 5.5 and the 7.5, and called the 6.5. As I mentioned above we'll also eventually do two stand-alone phono stages for the 5.5 and the 7.5.

We're in the research stage on some other projects that I'll be able to talk about in more detail later this year when we're closer to completion, and we'll probably pursue some applications for multichannel products in the not-too-distant future.

Lots of work to be done, and some exciting products!

*Is there anything else you'd like to share that might help **Journal** readers in their search for musically satisfying amplification components?*

From this discussion you know I feel that tubes, with their inherently simple circuits and low NFB, sound the best, but I would think that the key to enjoyment of one's audio system is to buy equipment that makes sense to the listener. In other words, I would choose only the equipment that sounds good to me and suits my system, rather than something that was highly recommended by somebody that might not know my system.

On the technical side, I would say that specs and measurements don't always tell the whole story, but some specs are important. For instance, impedance matching and overload headroom would be conducive to good sound that is predictable under a wide variety of conditions, and balanced differential circuits (without use of a reactive transformer) seem to sound more dynamic, but I would make sure that the balanced circuitry is properly implemented, rather than just something glued on by the marketing department.

In closing, I would like to say that I sincerely hope that all of this information is of help and use to **Journal** readers, and I want to thank you for visiting us today.

Thank you both for allowing me to visit your production facilities and for providing lots of information for our readers. APJ

Processor vs. Preamplifier

by Richard Hardesty

*In order to provide more than rhetorical information we compared several of the audio preamplifiers reviewed in this **Journal** to a high quality surround sound processor. We used a reference-grade processor for this comparison—the Theta Casablanca II with Xtreme DACs.*

The Casablanca II is the finest surround processor we've heard. Some of the sharpest minds in audio worked on its design. It incorporates some of the best parts available and

costs nearly as much as my wife's new Mazda Protegé 5. This product offers the best sound that surround sound processors can produce. We wanted to see how it might compare to an audio preamplifier when used to reproduce music.

Many people have been told that a high quality surround sound processor is also a good music preamplifier. The Theta is advertised as "a music and cinema controller." Our goal was to determine, by actual comparison, whether that sales pitch holds up under scrutiny. Is an expensive surround sound processor really an acceptable music preamp?

This article is not meant to denigrate the Casablanca II. In fact it's a tribute to that excellent product which is arguably the finest component of its type. This article is meant to offer a viewpoint that may counter much of what you've heard and encourage you to perform some listening comparisons.

State-of-the-art DAC & a High-End Preamplifier for Free?

The Theta Casablanca II is presented as a surround sound processor that can also offer high-end digital-to-analog conversion and audiophile musical performance. It's truly an outstanding surround sound processor and we didn't challenge that claim because we can verify that it's true. We evaluated the Casablanca II's performance as a DAC/preamplifier and as an analog preamplifier. Here's what we did.

The dCS Verdi transport was used throughout this test to play CD and SACD discs. We used the digital output from the Verdi transport to test the Casablanca II's performance as a digital-to-analog converter and preamp. The digital output from the dCS Verdi was used to connect the dCS Delius DAC for comparison. We used the analog outputs from the dCS Delius digital-to-analog converter to gauge the performance of the Theta Casablanca II as an analog preamplifier. We compared the Casablanca II's performance as a DAC/preamp to the dCS Delius DAC and to Theta's own Generation VIII DAC feeding a high quality analog preamplifier. We compared the analog performance of the Theta Casablanca II to the performance of the line stage preamplifiers reviewed in this **Journal**.

We connected the digital output from the dCS Verdi transport directly to the digital input of the Theta Casablanca II. In this configuration the Theta Casablanca II acts as a DAC and a line stage preamplifier when music discs are played through the combination.

We connected the analog outputs from the dCS Delius DAC to the "Analog Direct" inputs of the Theta Casablanca II. In this configuration the Theta Casablanca II acts as an analog preamplifier but the digital circuitry, which provides many of the features for which the Casablanca II is known, is bypassed. Here's what we heard.

Casablanca II as a DAC/Preamp

The Theta Casablanca II is a good, but not excellent, DAC/preamp. The music signal retained good detail but the sound from the Casablanca II lost dimension and musicality when compared to the sound from high-end separate components. Using the digital input on the Casablanca II produced sound from the Theta that was more mechanical and less three-dimensional than that delivered by Theta's own Generation VIII DAC, which can best be described as excellent, or by the superb dCS Verdi DAC, when played through a high-end audio preamplifier. The sound from the Casablanca II was more than acceptable for film soundtracks but was markedly inferior to the sound from the other components we tried when music was the source. How did the Casablanca II fair as a line stage preamplifier?

The Casablanca II performed poorly as an analog preamplifier when compared to the preamps reviewed in this **Journal**. A modestly-priced preamplifier, the Ayre K-5x, easily outperformed the Theta Casablanca II when used for music reproduction. The better preamps simply left the Theta in the dust.

The Theta Casablanca II is a high quality product with open architecture that can be easily upgraded. Many solid arguments can be offered in support of this product and as a surround sound processor the Casablanca II can't be faulted. If you have the funds, buy a Casablanca II and an SOTA preamplifier. Those with budget constraints should remember that any computer approved by Dolby and DTS can successfully run the algorithms necessary to decode film soundtracks and even a very expensive surround sound processor simply cannot compete with a dedicated music preamplifier.

Buying an inexpensive processor along with a high quality audio preamp would probably be the most cost-effective way of constructing a satisfying audio system that can be used for both music and movies. Assembling a no-compromise music system first and adding surround sound capability by running the front channels from the processor through the preamplifier will provide the best sound for the least cost. APJ

PREAMP Preamble

by Shane Buettner

We've got something really special for you here with this preamp survey. Our reviews are different. Here's why.

If you pick up any of the mainstream magazines you're going to see preamp reviews, maybe even a few per issue, performed by several different reviewers with different rooms, different systems, and different biases. The magazines' year-end recommended systems features are compiled largely on the editorial staff's opinions of the reviewers' opinions. In some circumstances the editors have heard the products themselves and go by that, in other cases they haven't.

The magazines won't provide what we're giving you here—two guys, two sets of ears, and two opinions on several of the best preamps currently available with the evaluations performed almost entirely in one room with one system with the products evaluated entirely head-to-head. And I need to blow my own horn here a little bit.

I performed much of the evaluation myself, with Richard Hardesty making two trips to my house where we listened together for three days each, and me making a three-day trip down to Richard's with more gear. One of the reasons this issue is late is that I've been meticulous in keeping all these preamps plugged in, with signals running through them as much as possible, for weeks on end. Some of these products have solid-state components that require long break-in periods to perform their best, which is how I want to hear a product.

I had disc players all over the house plugged into these things and playing for weeks. As I write these words my wife is happy as hell that it's over and I don't have the heart to tell her that the power amps for **Journal** #11 are about to start showing up on the doorstep, soon to be strewn about the house in the same fashion! My point is to sell what we're offering here—an attention to detail and a format you can't get elsewhere. Viva **APJ**!

Before getting started, let me talk about the system in which these products were evaluated. I've recently moved into a new home with a dedicated theater/listening room that's approximately 25 x 17 feet. The component racks are toward the back of the right sidewall, with the exception of the power amplifiers, which are up at the front of the room right next to the loudspeakers.

I run 10-meter balanced interconnects from my preamp to the power amps, and use 4-ft. speaker wires. In every instance possible with these preamps I used balanced cables, and when the preamp had only single-ended outputs (as with the ARC SP 16L, Ray Samuels Emmeline CA-2, and Rogue Audio Magnum Ninety-Nine) I used a pair of Cardas XLR adapters.

Some of my gear has changed, and the Vandersteen Model 5A speakers are the most notable. They came in at the beginning of the summer for review in *The Absolute Sound*, so I was able to use them throughout this entire survey. And they aren't going back—I purchased the review samples from Vandersteen Audio.

I used the preamps you'll find reviewed below but my long-term reference has been the BAT VK-51SE. Source components include the Musical Fidelity Tri-Vista CD/SACD player which I've bought and own, and for the bulk of these reviews I had the privilege of listening to the dCS Verdi/Delius transport/DAC combo reviewed in **Journal** #9 (that gear went back in late August, and that was a sad day!).

My Linn LP12 has also undergone some changes and now features a Lingo power supply, Cirkus bearing, Trampolin base, Ekos tonearm, and the Akiva cartridge. Perhaps most significant, I've been using Audioquest's new DBS (Dielectric-Bias System) cables: Cheetah interconnect and Kilimanjaro speaker wire. Both are battery-biased silver wires that use the battery voltage to bias the cable dielectrics. You'll read more about these cables in an upcoming **Journal**, but man, these things are on a whole other level. It's the closest thing I've heard to eliminating the wire from a system entirely—breathtaking transparency. Richard Gray's Power Company 1200S and 400S devices are the only power conditioning product I use. My racks are by Lovan.

Richard Hardesty adds: at my house we listen to Vandersteen Model 3A speakers with 2Wq subwoofers in a vertical biamp configuration. We use my LP-12/Ekos/Arkiv turntable with Lingo power supply along with my Wadia 861 CD player and Vacuum State Electronics modified Sony VDP-9000ES SACD player. I have the same fabulous Audioquest cables mentioned by Shane in slightly different lengths, and I'm just as enthusiastic about them. APJ

CALYPSO Line Stage & RHEA Phono Stage

by Shane Buettner

Aesthetix was started nearly 10 years ago by president and designer Jim White, an experienced industry veteran. Notable among Jim's achievements is that he worked with Theta Digital for many years on design teams that developed their landmark Casablanca digital controllers up through the current Xtreme DAC version (Dude!) of the Casablanca II. Jim also worked on the design of Theta's Dreadnaught power amplifiers, which Richard Hardesty and I both purchased and still use as our reference power amplifiers. How's that for a resume?

Aesthetix hit the ground running with the ambitious Jupiter series preamps—the Callisto line stage and Io phono stage. These highly regarded products are two-box designs, with each component having its own separate power supply and massive numbers of tubes in each chassis and they sell for $6,500-$14,000, depending on unit and options. By virtually all accounts the Jupiter series lives up to its celestial name and indeed challenges the state of the art in preamp design and performance.

The challenge Jim White undertook with the Saturn series components we're looking at here was to approach the lofty performance benchmarks established by the Jupiter series for a fraction of the price in a pair of single-chassis designs. The results are the Calypso line stage and Rhea phono stage, which sell for $4,500 and $4,000, respectively. (The series also includes the Janus, which is a line stage with integral phono for $6500.) As you'll find out these products pack an amazing amount of build quality, features and performance into two gorgeous boxes that indeed give the higher priced products a run for your money.

Calypso Line Stage

Just to make sure you're tuned in, remember as I list Calypso's attributes that this is a $4,000 line stage!

Calypso is a single-box tube line stage with a fully differential circuit topology—single-ended signals are converted to balanced in the gain stage and remain balanced through the output. An additional conversion takes place when the single-ended outputs are used. Balanced input signals stay balanced from input to output. No global feedback is employed.

There are five inputs plus a tape monitor loop and two sets of single-ended and balanced outputs per channel. Calypso uses a solid-state power supply built with E&I core transformers rather than toroids. There's an inductor choke on the high voltage power supply to quench noise.

The input switching is solid-state with FETs while four tubes are used in the gain and output amplification stages (Sovtek 12AX7s for gain and 6922s in the output/buffer stage). The volume control uses FET switches that switch discrete resistors with 66 steps of 1dB. A bypass mode can be engaged easily (two pushes of the bypass button within 5 seconds) on any input to establish unity gain for surround processor pass through. The remote is fully featured—in addition to engaging the bypass mode you can invert phase, shut off the front panel display and make input selection, mute the signal, and control the volume and balance (balance is available only on the remote).

On the tactile front, Calypso is heavy in the way that makes you feel like you've really bought something substantial. It's pretty in a tough way with a brushed anodized aluminum chassis, which is non-ferrous and resists magnetic artifacts. But Calypso is more than weighty. Every aspect of it that you'll touch has a rugged, substantial feel to it from the input connectors to the front panel buttons and the embossed A (for

Aesthetix) logo. The "A" logo and the front panel are cut separately with a laser, and then the two pieces are pressed together. Pretty cool.

The front and back panels and bottom chassis are silver while the top cover and side panels are black. Volume can be adjusted on the front panel by pressing either side of the blue LED display, which I hadn't seen before. The tubes require installation and, quite surprisingly, the top cover lifts off with a good tug to separate the patches of what look like Velcro™ that hold it to the chassis. I say the material *looks* like Velcro™ because Aesthetix has informed me that it's actually an "advanced polymer interlocking material." Geez, even the stuff that looks like Velcro™ is actually some high-grade "unobtanium" on these things! Whatever it is, it works just like Velcro and so long as it makes for a good and ridgid coupling (and Calypso's performance suggests it does), I'm all for avoiding 30 access screws to pop the cover.

As impressive as Calypso is outside, unlike most of today's supermodels it's got inner beauty to match. (That's actually speculation on my part—I haven't dated a supermodel in a long time.)

According to Aesthetix, Calypso and Rhea both use power supply and dual-mono audio circuit designs derived from the Jupiter series products. Opening the top cover you'll see the transformers are up at the front of the chassis in a stainless steel can, and set behind them on either side are mirrored, physically isolated circuit boards for the left and right channels. High quality parts are contained throughout such as Neutrik XLR connectors, Roederstein resistors, and WIMA capacitors (the lit doesn't brag about it but there are some Relcaps in there too).

The board layout is clean and uncluttered in appearance, and true to Aesthetix's literature you don't see much wire. The wire that runs from the AC receptacle on the back panel to the can with the transformers passes through a shielded conduit. Overall, Calypso appears to offer a staggering amount of build and parts quality for the money.

Now, there are two more features that set apart Calypso from the other preamps we've reviewed here. Many preamps with remote control, especially one as fully functioned as Calypso's, utilize a microprocessor to execute remote commands. As with Ayre's K-5x, Calypso's microprocessor is narcoleptic—it stays in "sleep" mode, waking only to execute a command before nodding off again so that it's not constantly generating noise. For additional noise reduction, the display can be shut down via the front panel or remote.

More impressive is Calypso's standby mode, which keeps all of the solid-state circuits heated up and the capacitors charged with higher voltage than is used in normal operation—apparently everything but the tube heaters and front panel display stays up. This should dramatically reduce the time it takes for your system to get up to speed when you turn it on and maximize the tubes' life span. That's hot, and it's something I'd sure like to see in the products that cost 2 to 3 times Calypso's price!

Last, but certainly not least, I should go into a little more detail on the remote that ships with Calypso and Rhea. If you bought both you'd have two remotes, but you only need one. The diminutive remote is plastic—the build quality is incorporated elsewhere, which is fine by me.

Three little buttons at the bottom switch its operation mode from Calypso to Rhea, or even to Janus if that's the way you've gone. I found operating Calypso and Rhea with this single remote a real snap, if not an outright treat. It even has direct access to the input sources. If you've been even halfway around the hi-fi block you know that high-end performance and good ergonomics seldom meet. The Saturn series Aesthetix gear is a refreshing step in another direction, and I've never used anything that's as complex in function yet simple to use.

Rhea Phono Stage

Rhea appears to be a phono twin of Calypso in most ways. From the outside looking in, only a slightly different set of front panel buttons and rear panel connections differentiate them. Rhea's made from the same chassis material, and finished in exactly the same strikingly attractive and heavy-duty fashion (including the space-age polymer that looks and behaves like Velcro™).

Although all three of Rhea's inputs are single-ended, signals are balanced in the gain stage and the audio circuitry from that point on is differential (balanced) through the output so long as one of the two balanced outputs is put to use. So, even with single-ended inputs many of the advantages of a balanced circuit topology are still realized.

As with Calypso, Rhea uses zero global feedback. There are two single-ended and two balanced outputs. Each input will "remember" discrete gain and cartridge loading settings so you can accommodate separate table/arm setups. Speaking of which, there are seven gain settings for moving coil cartridges in (mostly) 6dB increments from 38dB-75dB (off is the 8th option). Nine cartridge loading settings are available: 75, 125, 250, and 500 ohms, as well as 1k, 2.5k, 5k, 10k, and 47k.

I've never used anything that's as complex in function yet simple to use.

Rhea's solid state power supply and dual-mono audio circuit boards are also derived from the Jupiter series products, and, although the power supply is solid-state, all gain and amplification is performed by tubes. Rhea uses more than twice as many tubes as Calypso with a total of ten tubes for its three amplification stages and output buffer stage (12AX7s for gain and 6922s in the buffer stage). Rhea's RIAA network is passive with polypropylene caps and Roederstein resistors

In general, with Rhea you see the same board layout and high quality parts you see in Calypso—only here, there're more of them. Rhea also includes the same nifty standby mode as Calypso.

As mentioned above, the remote control and user interface are outstanding. When Rhea's remote mode is operational you can switch inputs, mute the signal or switch load and gain settings from the remote. This is just killer for dialing in the right cartridge loading value on the fly with familiar material. In addition to all this, you can run Aesthetix's renowned cartridge demagnetizer from the remote or front panel.

I have to say I was looking forward to taking Rhea for a spin (literally!) after using a Linn Linto for over two years. The Linto is a terrific phono stage that I've heard dump other phono stages that cost multiples of its price. But it is rather rigid in gain and loading configurations, which Rhea certainly isn't.

Performance

Let me just say it: the Aesthetix gear is the stuff that throws off the curve for everyone else. Calypso provides the kind of performance that you just won't hear with anything else anywhere near its price point, and it's better than many things that cost twice as much. In fact, it's my opinion that of the preamps reviewed in these pages only the VTL 7.5 eclipses the Calypso's overall performance.

Calypso doesn't have quite the same level of see-through transparency and layered dimensionality that the VTL 7.5 has, nor does it have quite the jump factor or vividness in the midrange that the ARC Ref 2 has. But Calypso comes shockingly close in overall performance to what's accomplished by its much pricier competitors.

Calypso has clarity and depth with focused, coherent images layered from front to back with musicians occupying distinct planes. Dynamics didn't jump from the speakers so much as they just flowed, rising and falling, waxing and waning in a startlingly natural fashion.

As with its more expensive brethren, Calypso could be turned up louder, and will rock harder, without strain or congestion. This preamp sounds closer to the neutrality of the VTL 7.5 than any of the others auditioned, and has much of the harmonically rich tube sound that's undeniably right for me.

The big VTL is the only preamp in this survey that exceeded Calypso's inner detail and resolution of fine musical texture. The Aesthetix isn't quite as open and extended on top as the big VTL or ARC preamps, but it extends very deep on the bottom with authority and no bloat whatsoever—just taut, articulate, detailed bass.

While Calypso doesn't quite match the mid-bass speed of the ARC, neither does it have that preamp's slight leanness in that area. I think you're getting my drift by now. The $4,500 Aesthetix is not quite everything that the VTL 7.5 and ARC Ref 2MKII are, but it's damned close for much less than half the price. I believe that's the definition of a benchmark and notice that there's a new sheriff in town.

Rhea has many of the same sonic characteristics as Calypso, which is to say it too is fabulous. Although you'll hear a fair amount of tube rush before the music starts, once you drop the needle you hear detailed, dimensional sound with a black, quiet background.

Rhea's ability to layer and separate images on the stage is extraordinary. An excellent example is found on the Classic LP *Come Away with Me* by Norah Jones.

The last track on the record is *The Nearness of You* with Norah singing at the piano. The piano is clearly out in front, with the vocal layered in space behind it—just like it would be if she were sitting behind the piano and singing to you in your room. Excellent sense of depth, space and focus along with outstanding texture and shading with the vocal.

Tonally, while Rhea captures all instruments with veracity, I was particularly struck with its rendition of piano. It showed just the right amount of foundation and weight with light, airy and natural decay. Extension at the low end is authoritative and powerful, but also detailed and quick. Rhea seems just as neutral and high in resolution as Calypso, and also has enough tube life in it to let you connect to it. I love this phono stage.

Comparing Rhea with the Linn Linto is an interesting exercise. The Linto is in some ways more spectacular as a result of lightness in the bass that emphasizes its rhythmic speed and midrange clarity—and also makes the Linto seem like it's got a little more jump at times.

I'm convinced that's a bit of coloration, but it's seductive to be sure. Rhea sounds fuller and warmer, but not out of proportion—it just has a low end foundation and depth that the Linto doesn't. Rhea strikes me as more tonally balanced.

The Linto seems livelier on top, but also emphasizes surface noise to a higher degree. On super quiet vinyl Linto's quick and punchy sound can be very appealing. On less pristine vinyl, Rhea lets you get more of the music and fewer playback artifacts. The fact remains that these two phono stages are close in overall quality on my Linn LP12 rig, in spite of a nearly 3:1 price difference.

Rhea is more expensive, and it does sound better to my ears, always pulling me into its deeper and more dimensional musical landscape. And Rhea is far more flexible in terms of inputs and outputs, as well as loading and gain settings.

The Linto, good as it is, has fixed gain and loading, which may be an issue depending on your current turntable setup, not to mention potentially limiting some move you might want to make with your LP playback in the future. Rhea can pretty much accommodate any about-face you might make at any time and, hey, for $4,500, it should!

Conclusion

The Aesthetix tandem is a lot more than a pair of "budget" products. Calypso, in particular, sets a standard. I would rank this preamp's performance in this survey where I did regardless of its price. That it costs only $4,500 is cause for celebration among audiophiles. It's not really a budget product at that price by any stretch, but it simply performs far beyond the level you'd expect in every way, including its ease of use and scope of features. Nothing I've seen is as ergonomically sound as the Aesthetix gear.

Rhea's in the same performance league, although it is expensive for a phono stage. But it has unsurpassed flexibility and blends seamlessly with Calypso in your system, both components effortlessly controlled by a single remote. And it too is simply beyond reproach in terms of pure performance. Jim White may have done his job too well. Even his more expensive gear is looking like a hard sell in comparison to Calypso and Rhea!

Richard Hardesty comments on...

CALYPSO Line Stage & **RHEA** Phono Stage
PREAMPLIFIERS

The Aesthetix Calypso and Rhea are among the most versatile and aesthetically pleasing products in this group. The Calypso not only looks beautiful, it is easy and convenient to use. It provides audio performance that rivals the most expensive components yet it costs comparatively little.

It's difficult to criticize a product this good, but if pressed I'd say that the Calypso may offer slightly less resolution of subtle details than the more costly offerings in this group and it leans ever-so-slightly towards the richer, darker (one could say more musical) side of neutrality.

The price of the Aesthetix Calypso line stage preamplifier will undoubtedly rise above its current bargain basement cost of just $4,500. At the current price this is a value leader but I would still consider it to be a top performer at twice the cost.

The Rhea offers lifelike vinyl reproduction with very little tube noise. The musical presentation and three-dimensional imaging set it apart from the solid-state competition. The Rhea leans slightly towards darkness and sounds rich and warm. It mates well with a variety of line stages. We have reviewed only a few phono stages but this is the best one auditioned so far. It's not cheap at $4,000 yet it provides accurate performance rivaling prestige components that cost far more. APJ

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Reference 2MKII

audio research

by Shane Buettner

No survey of high-end preamps could be started, let alone considered complete, without Audio Research Corporation's top dog, which these days is the Reference 2MKII.

Audio Research has been one of the companies that matter in hi-fi since 1970—the very year this reviewer was born (I'm not sure if that says more about them or me, but there it is). They've had far more hits than misses over the decades, and anyone who knows which end of the plug goes into the wall will tell you that ARC's enviable stature in the industry has been earned.

The Ref 2MKII uses tubes like ARC's other Reference series components. This latest iteration of the ARC flagship preamp is priced at $9,995 MSRP, same as its predecessor.

Owners of the original Ref 2 can upgrade to MKII status for $2,995 and Ref 1 owners can step up to the big time for $4,995. The former receive a two-year warranty extension while the latter get bumped up for three years, and new tubes are included in both cases. When you invest in ARC they invest in you too. Let's talk about what the consumer can expect as a return on that investment.

Getting to Know You

The Ref 2MKII is a single-box chassis with integral power supply. As for appearance, the newest Ref 2 is classic ARC, more than a little retro with industrial-looking handles, cool-looking knobs and toggle switches. (I have a long history admiring toggle switches. A buddy of mine once installed toggle switches for the electrical controls of his '73 Toyota LandCruiser, including the starter, and I thought that was just too slick!) The top and sides of the chassis are vented to let the tubes get as much air as possible and, as you can imagine, it would be a good idea to give this thing plenty of breathing room in the rack.

The Ref 2MKII shows up in a box with its eight vacuum tubes packed in foam inside the front two boards of the chassis. The toughest thing about getting the tubes in and firing it up is the time it takes to get all the screws off the top cover, although it's likely your dealer would do that for you. The user manual's instructions are thorough and simple to execute.

While the Ref 1 used a solid-state power supply, the Ref 2MKII is even more of a tube preamp—it uses tubes even in the power supply. Changes to the Ref 2MKII are mostly to the power supply and the analog input, driver/gain and output stages.

According to ARC the input board and digital circuits are the same as the Ref 2. Power regulation has been beefed up to match the kind of dynamic response that ARC believes its analog circuitry is capable of delivering, and different toroids are used. The solid-state input stage now uses JFETs, and the gain/driver stage uses two 6H30 tubes, the Russian so-called "Super Tube" that is also used in Balanced Audio Technology's VK-50SE and VK-51SE preamps, among others. Two more 6H30s are direct-coupled to the gain tubes to drive the output stage.

The Ref 2MKII's volume control is a digitally controlled chip-based solution with the resistors on the chip of differing values 100k and 10k. The former value resistor operates at higher volume levels, the latter at lower levels. The volume control features 125 steps of 0.5dB until the volume level gets down to about -40dB, at which point the increments grow to 1 and then 2dB. Each channel uses four mono volume control chips—two for each leg of a balanced signal, per channel, for a total of eight chips for balanced stereo operation. The Ref 2MKII is indeed fully differential, balanced from input to output.

Features and Functions

The Ref 2MKII has eight inputs that can be used with either balanced or single-ended connections. One of those cool toggle switches on the front panel switches between balanced or single-ended for the selected input and the Ref 2MKII remembers which mode was last selected. One of the eight inputs is labeled “Processor” and as the name suggests this input is configured for unity gain so that the front channels of a surround processor/controller can be connected and the controller’s volume control is used. As you’ve read in this issue of the **Journal**, this is critical for high-end music listeners interested in cobbling surround sound around their music playback rigs.

The aforementioned, cool-looking knobs are inset in the front panel. The volume, balance, input and record source selection controls all operate in the same fashion—crank them once in either direction and you move in a single increment; hold them down in either direction and you move continuously.

In addition to toggles for power on/off and balanced/single-ended mode operation there are switches for inverting phase and muting. Virtually all of these controls are also accessible from the small, slim line remote. This fully-featured preamp is very simple and straightforward to use.

The only feature I would like to see in a preamp at this price that’s not here is a standby mode (the ARC is either all the way on or off) like the one in Aesthetix’s Calypso that keeps the solid-state circuits heated up when not in use (only the tubes sleep). This reduces warm-up time and also increases the tubes’ longevity. Aside from that, the ARC has everything.

Performance

The Ref 2MKII that ARC sent for audition had been in use in their listening room and so was a well broken in unit. After just a few hours of listening it was up and in its performance envelope.

The first things that are going to jump up and grab you about the ARC are its incredibly vivid and lifelike midrange and striking dynamic contrast. No matter what the program material, the ARC has that magic. It’s not a Conrad-Johnson style tube glow by any stretch—that’s not ARC’s tube signature. It’s more of a sheer vibrancy that makes vocalists in particular sound like they’re standing right there in the room with you.

Once you’ve heard the Ref 2MKII’s midrange that could very well be it for you—there may be no going back. Along with this midrange resolution and allure, the dynamic snap and speed of this preamp are also hard for the other products in its category to match. While VTL’s 5.5 is the only preamp reviewed that matched the ARC’s jump, it couldn’t resolve nearly as much fine detail and musical texture.

Now, the midrange magic is undoubtedly the Ref 2MKII’s strength, and paradoxically it’s the most controversial aspect of its performance. None of the other preamps reviewed have the Ref 2MKII’s splashy and spellbinding midrange magic. Does ARC know something about midrange reproduction that none of the other manufacturers know or is there a little Technicolor in that frequency range? I think there might be some truth on both sides of this question.

It does seem a little more human than human, if you get me. In spite of the midrange’s grab-you-by-the-lapels vividness, I never felt it push far enough forward to be bothersome, nor did I detect any glare or grain. I never felt like any particular instruments or piano keys jumped out any farther than any other.

On the other hand, I did come to believe that, in comparison with the other preamps I reviewed, the ARC is slightly leaner in the bass, especially the mid-bass. Loads of speed and texture make up for that, but I think there’s a possibility that the slightly lighter bass makes the midrange and dynamics stand out just a bit.

The treble doesn’t sound closed down at all in comparison, just very airy and extended—very natural with cymbals and piano decay that’s extremely engaging. If you enjoy listening to it as much as I did (and I think you will) then who cares why?

But there are other things going on here that I need to talk about. In reviewing products that span the price spectrum from $1,000 to $12,500, there are some things I noticed about what you get when you spend more money for a preamp and the Ref 2MKII exemplifies some of these attributes.

The Ref 2MKII and some of the other more ambitious designs have an ability to play loud, with contrast, without sounding congested or any less refined. The Ref 2MKII can crank up loud enough to do heavy rock or orchestra and it just holds together in a way that the less expensive preamps can't approach.

There is also a difference between preamps like the Ref 2MKII and the lower priced products in wringing the nth degree of detail and spatial refinement out of the music. While the lower priced preamps get the music right, the ARC gets much farther into the recording space and draws out more nuance, texture and shading.

Conclusion

Simply put, the ARC Ref 2MKII is a blast to listen to, and is unquestionably a must listen for anyone shopping for a high-end preamp. It holds its own with any preamp I've heard at any price in virtually every regard and its combination of dynamics, speed, texture, and vibrant life in the midrange is unparalleled.

More than anything, this is a preamp that will let you sit down and listen to any kind of music—and get off on it—for hours on end. The Ref 2MKII is another compelling product from ARC, another product that matters. APJ

LINE STAGE **SP16L**

audio research

by Shane Buettner

The SP16 is ARC's answer for a cost effective, single-boxed, tubed preamp solution available with a phono stage for $2,495 (SP16) or without for just $1,995 (SP16L).

The phono section is rated at 54dB gain, which is borderline for a low output moving-coil cartridge like the Linn Arkiv I use, so I stuck with the SP16L line stage-only version. In spite of its diminutive price tag the SP16L delivers a full suite of the features users are looking for, albeit in a more utilitarian and less splashy fashion than its big brother, the Ref 2MKII.

Looks and Features

The SP16L has a boxy look to it with none of the retro-looking knobs or toggles that adorn other ARC gear. On the left side of the front panel is a section housing a row of LEDs that indicate volume level, with another row of LEDs above it showing the input source, as well as indicating if the signal is muted, if the processor loop is being used, or if mono playback has been selected. Next to that section on the front panel is another with several black buttons—power, mute, processor, and up and down for volume.

The SP16L is pretty light and the front panel buttons have a plastic feel to them. Rogue Audio's Magnum Ninety-Nine costs a little more at $2,395, but has an aluminum chassis with solid-feeling knobs and an external power supply. All of the $3k-and-up preamps feel more substantial as well.

The back panel reveals six single-ended inputs including a processor input that's set for unity gain. I don't have to say at this point how cool that is. There are two sets of main outputs and one tape/record loop output, all single-ended. Hey, it's a single-ended preamp! The input connectors have the same solid and substantial feel as those on the Ref 2MKII. The AC cord is removable of course.

The remote control is a small wand and fully functioned although there is no provision for inverting phase from the remote or front panel. Can't say I personally miss it—that's not a feature I use.

Inside

The SP16L uses a solid-state power supply and it's all tubes beyond that. The input/gain and output stages use 12AX7s—three in the line stage only version, six for the phono version.

The volume control is a less sophisticated version of the Dallas chip solution used in the balanced Ref 2MKII—the resistors that attenuate the signal are on the chips in both cases. There are 70 precisely tracking steps of attenuation.

Performance

This review is quite literally a tale of two preamps. I reviewed the initial SP16L and found a number of undesirable characteristics in its performance. It didn't sound like a tube line stage—it sounded very solid state, and in all the wrong ways.

It was very fast, very whitish, and had a very transistorlike sound that is not what I think people expect when they buy a tube line stage. It reminded me of a TV with the sharpness cranked up too high. In all honesty, if I hadn't installed the tubes myself when the SP16L arrived I wouldn't have believed they were in there.

The soundstage was extremely compressed front to back, and very flat. The expected family resemblance between it and the Ref 2MKII simply wasn't there. I described the initial review sample's sound as being like a diamond—almost transparent and among the hardest substances known to man!

ARC was taken aback by my findings when they received my review for fact-checking. They listened to the returned sample and agreed that something was amiss, attributing what I heard to the Electro Harmonix 12AX7s having gone bad. They promptly sent a replacement with new tubes that did indeed sound different, mitigating or eliminating many of the issues cited with the first preamp.

The re-tubed SP16L is nowhere near as hard and white as the first sample. It has a respectable semblance of depth in the soundstage and good clarity overall.

Products in this price range simply aren't going to be as truly revealing and transparent as something like the Ref 2MKII, but the SP16L provides an illusion of this clarity to a degree. Through the midrange and the vocals the sound is still a bit dry for my taste.

The bass is lean and quite taut, but fast and rhythmic as all get-out. You may be so enthralled with this aspect of its performance that you won't notice its lack of fine texture, inner detail and extension. On an acoustic bass for example you'll hear the notes themselves, but not so much of the fingers on the strings. And let's say Ray Brown plucks a big, fat, open string—it doesn't bloom and boom as much as it does on the more expensive, higher resolution preamps.

Similarly, the SP16L doesn't fully open up on top. Piano decay, for example, sounds a little truncated and lacking in air. Now, all of this is pretty much what one would expect from a preamp in this price range. In other words, something has to give at $2K.

But you'd also expect from this description that a tube line stage that's rolled off on top and bottom would have a warm, tubelike "midrangey" sound, but even with the new tubes that isn't the case. I wouldn't describe this as "laid back" or "forgiving" under any circumstances. If the Ayre K-5x is a solid-state preamp that some tube guys should look at, then the SP16L is a tube preamp that some solid-state guys are going to love.

The plus side of these characteristics is that fans of this kind of sound will get off on the speed and apparent clarity the SP16L provides. And it is fast and surprisingly dynamic. If it were paired with an amplifier—tube or otherwise—that was softer and forgiving in the midrange it could make rhythmically involving music that you could tap your foot to with no apologies. The only caveat I'd throw out there is that such a system would be one of mutually beneficial coloration, so when and if you replace those components you may want to replace both at the same time.

Conclusion

For just $2K ARC's SP16L is as fully featured as one could expect. Its sound is not a tube-lover's sound in my opinion. People who prefer its sonic characteristics will admire its clarity and transient speed, while others might want to live with a more relaxed and classic brand of tube sound in this price range. I confess to being in this latter camp, but I encourage you to listen for yourself and see if you agree. APJ

Richard Hardesty comments on...

audio research Ref 2MKII & SP16L

LINE STAGE PREAMPLIFIERS

The ARC Reference 2MKII is an outstanding preamplifier. It is characterized by a slightly lean yet rhythmically satisfying bass and a minor emphasis, or "spotlight," on the midrange, particularly the vocal region. I won't pretend to end the argument about whether this vocal range clarity is an artifact. I suspect *that the ARC is slightly less transparent rather than slightly better than the others, and some are likely to be enthralled by the lifelike midrange presented by the ARC, whether it's better or just different. I liked it.*

I agree completely with Shane's assessment of the ARC SP16L. The only characteristic shared between the SP16L and the Reference 2MKII is the nameplate. I didn't hear the second review sample. APJ

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Ayre K-5x

LINE STAGE PREAMPLIFIER

by Shane Buettner

Ayre's Charlie Hansen isn't the first product designer to tell me that the greatest challenges in audio aren't in the megabucks, all-out assaults on the state of the art. Far trickier, it's said, is to design an audio component that challenges its pricier competitors in quality for a lot less money. Just how close can Ayre get to the performance of their reference quality K-1x ($7,000-$8,600) preamp for a mere $3,000? The answer to that question is the K-5x.

Design, Form and Function

The K-5x is a fully differential (balanced) solid-state line stage preamp with remote control. Like the K-1x (and per Ayre design philosophy) it incorporates no global feedback.

The chip-based, digitally controlled volume controls that are common at this price point weren't good enough for Hansen and company, so the K-5x features a proprietary volume control with FET switches that switch discrete metal-film resistors. Hansen is extremely proud of the performance of this volume control versus the off-the-shelf solutions that are available. It sports 66 levels in 1dB increments.

Where the K-1x features a separate outboard power supply and is built entirely with discrete circuits, the K-5x is a single box (with a removable power cord) that relies on monolithic circuits in the signal path similar to those used in Ayre's CX-7 CD player and AX-7 integrated amp. The K-5x cannot be outfitted with a phono section, and it's no wonder—there's no room in the box! That sucker's packed!

The K-5x has a cool looking silver aluminum case and, overall, it matches the industrial-style CX-7 and AX-7. Good looking in a spare way, the K-5x's look is all business with one exception—the mystical touch of using celestial symbols instead of names or numbers for input selection.

A row of buttons on the front panel features these lucky charms: a star, a planet, a comet and a moon. Just above these celestial input symbols is a display with blue characters. A single knob for volume and a mute button are the only other features on the front panel; the K-5x is on all the time so no power button is needed.

The K-5x features two balanced and two single-ended inputs. Outputs are also one each balanced and single-ended, in addition to a pair of single-ended tape loop outputs. Any input can easily be set to Processor Pass-Through mode, which means the input is set to unity gain, allowing the external source component to control the volume for that input. As the name indicates this allows you to connect a surround sound controller/processor to your hi-fi without the hassle of swapping wires.

As you'll read elsewhere in this issue of **APJ**, it's critical for avid music listeners and audiophiles to have an alternative to using a surround controller as a line stage. To set up an input for Processor Pass-Through you simply unplug the K-5x for ten seconds and then plug it back in while holding down the input selector for the desired input. Now the K-5x's display will show "PP" when that input is selected.

According to Ayre the microprocessor that commands all of this impressive functionality in the K-5x stays in "sleep" mode until it receives a command, which it executes before returning to sleep mode again to negate noise contamination from the microprocessor.

The Ayre K-5x is fully featured, and yet it's also easier to use than just about any I've experienced. You can control everything from the remote, and its user interface is so clean and simple I just don't see how it could be any better.

Performance

The K-5x user manual states that it needs 100-500 hours of music played through it before it breaks into its full performance envelope. While I found this to be true I'd also note that it does make very good sound right out of the box. It sounds just a hair cold and a little mechanical, maladies that disappear after a few hundred hours are on it.

As for listening, the K-5x right away struck me in much the same way as the other Ayre components I've heard. Sonically it's very much in tune with the house sound over at Ayre. I immediately took notice of the detail I was hearing.

Right out of the gate I was somewhat stunned by how revealing and detailed the K-5x is. It just doesn't sound like a $3K preamp. As with other Ayre components, the background is velvety black and quiet. Words like "purity" and "clarity" come to mind.

The focus is exceptionally sharp, and the soundstage has depth and dimension. Is it a baby K-1x? Yes, I think you could say that. The K-5x does many of the things that the K-1x does, just not to the same degree.

The most noticeable areas of performance in which the K-5x falls short compared to its big brother are that it doesn't match the K-1x's staggering amount of dynamic contrast, nor does it focus to quite the same razor-sharp degree, and its resolution of inner detail isn't up to the same level of "you are there" transparency. And for $4-5K less, it damned well shouldn't!

There are small criticizms. This preamp sounds killer and, like the K-1x, it has as little of its own signature as I can imagine. In other words, I couldn't detect any nip or tuck from neutrality. This thing just lets you hear the music and the components connected to it without imposing its own signature on the music. You can absolutely plug this thing in and just dive into your record collection.

Comparing this product to the other preamps in its price range is interesting fun. Although the ARC SP-16L is a tube preamp, it sounds more solid-state than Ayre's brand of solid-state! The Ayre is far more liquid sounding and yet is noticeably more resolved.

By comparison, if the K-5x has a polar oposite, the Rogue Audio Magnum Ninety-Nine is it—not as extended at the frequency extremes, not as spatially resolved or as dimensional or defined. And the VTL 5.5 isn't as quiet or refined as the Ayre, but is in some ways more spectacular to listen to with its light and airy tube glow.

The Ayre is more ergonomically refined than the VTL or the Rogue, has a more sophisticated volume control, and it's the only preamp we looked at below $4K that's fully balanced. The Ayre K-5x is a very compelling package to be sure.

Conclusion

The Ayre K-5x is a killer preamp and a hell of a deal for $3K. Its performance truly transcends its price point, making it the choice for fans of solid-state sound at that price. I can unequivocally state that you should NOT buy a solid-state preamp anywhere near this price before looking up an Ayre dealer.

What's more, the K-5x is so liquid and musical that those who prefer the less golden-sounding tube gear may want to give it a listen to see if Ayre's brand of solid-state sound is something they could live with. My experience with preamps to date indicates that high-end performance starts at $3K with the Ayre K-5x. APJ

Richard Hardesty comments on...

Ayre K-5x LINE STAGE PREAMPLIFIER

I agree. This is where the high-end starts for solid-state preamplifiers. The K-5x sounds crisp and clear with lots of resolution and it provides three-dimensional imaging that's not often encountered in solid-state components. It's a very attractive piece that's easy and convenient to use. It's made with high quality parts like other Ayre products.

The K-5x compares favorably with any solid-state preamplifier, except its big brother, the K-1x, which stands alone and establishes the performance standards for the class. Compare these preamplifiers with any surround sound processor for an ear-opening experience. APJ

Ayre K-1x LINE STAGE PREAMPLIFIER

by Richard Hardesty

In the course of my career in audio I have listened to many, many solid-state preamplifiers, including those marketed primarily to the carriage trade with list prices that regular people like us never actually pay. Although some were beautifully built—and beautiful—I have been disappointed by the performance of *nearly all of them. Solid-state preamplifiers tend to deliver music that has a slightly sterile sound and an image that is somewhat two-dimensional. This sound may seem clear and detailed to some but the results are often musically unsatisfying for me. The Ayre K-1x is an exception.*

The K-1x is an all-transistor component that offers all the clarity of the best solid-state products along with exceptional dynamic contrast and remarkably three-dimensional imaging. This results in a more natural presentation that allows the listener to become more involved with the music.

Outside

The Ayre K-1x is sturdy and functional in appearance. The inductor-filtered power supply is housed in a separate enclosure with a removable power cord. This power supply enclosure is attached to the main chassis with an umbilical cord that can be disconnected only at the supply end.

The thick, nicely finished front panel of the main chassis is rather spartan in appearance. The large volume control knob sits in a milled recess towards the upper left. Two knobs, roughly centered, are used to select the input and direct that signal, or another, to the record outputs. You can listen to one signal and record another at the same time if desired.

There are three indentations near the right side of the front panel that contain the IR (infrared) receiver to accept the remote control signals, a mute/operate switch, and an LED that displays operational status. There is no power switch. When the power supply is plugged in the circuit is hot. If power is interrupted, there is a delay to allow circuit stabilization.

The rear panel contains a mirror image of connectors. The left channel inputs and outputs are duplicated by those for the right channel, and the left and right channel connector groups are inverted to provide the shortest possible signal path internally. There are 3 balanced, and 3 unbalanced inputs for each channel, with the optional phono stage input placed at Balanced 2.

There are unbalanced tape-outs for each channel and a centrally located ground connector. There are two pairs of balanced outputs and one pair of single-ended outputs positioned near the outer edges, at each side of the back panel.

With the exception of remote-controlled mute and volume, this preamp is operated manually. There are no visible frills and no mechanical gadgets to fail. This is high-end audio with few concessions to ergonomics.

The lack of fancy automation is likely to render a product that is both quiet and bulletproof in operation, which is confirmed by my personal experience.

The optional remote control is housed in an aluminum extrusion and there are few buttons. You can raise and lower the volume and mute the output from the remote control. The input must be manually selected and manually directed to the tape outputs. I'm old-fashioned and this suits me fine.

Inside

The K-1x incorporates all the features that an audiophile could imagine in a solid-state preamplifier, and implements these features with the finest parts available.

The power supply includes inductor-input filtering, and multiple "Ayre Conditioners" (power-line RFI filters) are applied at strategic points. The signal path is incredibly short and almost no wire is used. Attenuation is accomplished by mechanical switches with solid silver contacts that place precision resistors in the signal path.

The preamplifier is a discrete, all-FET design with no negative feedback and the attenuators utilize Shallco solid silver contact switches. The signal path is remarkably short with all input and output connectors, along with the active circuitry for the line stage, placed on a single circuit board that is mounted near the back panel. The circuit boards are made from special "high-speed" material. The preamplifier features balanced circuitry throughout.

The dual-mono, balanced phono boards (optional) reside along the sides of the enclosure and have adjustable gain and loading. Adjustments are accomplished by inserting precision resistors in sockets and tightening setscrews, which firmly attach these components to the circuit boards. It's not convenient but it's very effective.

You can select the exact amount of gain necessary to match virtually any cartridge and that cartridge can be loaded with exactly the impedance required for the best sound and damping. Many resistor values are furnished with the preamplifier and you can add anything that you desire.

Much of the enclosure is filled with air that is interrupted only by control shafts that run from the front panel knobs to the switches located on the board at the back of the chassis. The chassis is rather tall to accommodate the Shallco switches, which are linked together by an ingenious system utilizing a cogged belt and a stepper motor to facilitate exact level tracking and remote volume control.

The Shallco switches are solid silver contact mechanical devices that act as stepped attenuators by selecting fixed, precision resistors to control volume. Four units are necessary to provide attenuation on each leg of each channel in order to provide completely balanced, dual-mono operation.

The mechanical arrangement is functional and beautiful and represents a real engineering achievement, in my opinion. When volume is adjusted, either manually or remotely, a slight "clunk, clunk" sound announces your accomplishment. A real man enjoys some audible confirmation.

Sound and Performance

I have used the Ayre K-1x as a reference preamplifier for about two years. During that time it has performed flawlessly while providing me with musical enjoyment and serving as a reference tool and as the centerpiece for **Audio Perfectionist** seminars.

The sound from thispreamplifier is almost completely transparent with virtually no audible flaws. I'm a very experienced listener and I was unaware of any shortcomings and probably would be still if I had not directly compared this preamplifier to state-of-the-art tube units that sell for about fifty percent more. Among solid-state units there are none better and, in my experience, no all-transistor device sounds as good.

The K-1x is virtually free from additive coloration. It offers a transparent window to the performance, which is presented with dynamic authority and an excellent rendition of rhythm and pace. SOTA tube products, which are usually somewhat less reliable and generally cost much more, can provide slight but perceptible improvements in the resolution of subtle details in the recording along with a sense of greater extension at both frequency extremes. That's the only criticism that I can muster.

Conclusion

The K-1x is the finest solid-state preamplifier that I've heard. I have lived with it as a reference component for an extended period and I still have immense respect for it. You can pay a lot more but you can't buy a component that is made with better parts and you can't aspire to a transistor device that is quieter or sounds better, in my humble opinion.

The optional phono stage in this preamplifier is the quietest and most functional phono preamp that I've experienced. It has provisions for adjusting gain and loading, making it one of the most adaptable phono stages available. Oh, and it sounds great, too.

Phono cartridges are inherently balanced and the Ayre phono stage has completely balanced circuitry. You'll have to alter your phono cables slightly to take full advantage of this capability (see the Ayre web site), but you'll probably be rewarded with the blackest background and the widest dynamic range you've ever heard from vinyl.

The line stage version of the K-1x costs $7,000 and the fully-loaded version that I have with the optional phono and remote sells for $8,600. I've listened to products selling for triple the price that can't equal the performance of the Ayre K-1x. I never thought I'd call a product that costs more than $8K a bargain but that's probably the best description for this audiophile component. It's the best solid-state preamp I've ever heard and it's far from the most expensive. APJ

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Balanced Audio Technology VK-51SE

by Shane Buettner

BAT is just seven years old as a company and yet they've already created an enviable reputation with high-end enthusiasts in that relatively short period of time. Their product line is ambitious, including source and amplification components (both pre and power), vacuum tube and solid-state gear, as well as multichannel power amplifiers with a statement digital controller for surround sound on the way. But the reputation I spoke of was created primarily by the company's tube preamps and power amps, and the subject of this review is the VK-51SE line stage, BAT's flagship preamp.

The "SE" in VK-51SE stands for "Special Edition" and is undoubtedly a statement product implementing designer Victor Khomenko's utmost philosophy on circuit design. It costs $8,500 without a remote ($9,000 with) and is a line stage-only with no provision for a phono stage. (BAT sells three stand-alone phono stages.)

Before I jump in, let me say that I've been using BAT's preamps as a reference for the last two-plus years, starting with the VK-50SE then upgrading to the VK-51SE. This review will contrast the improvements made in the new model, and another point that shouldn't be missed is that BAT offers upgrade paths throughout their line to support those consumers who've supported BAT. VK-50 and VK-50SE owners can send their units in for upgrades at incremental costs.

Outside

The styling of the VK-51SE, like a lot of BAT gear, is masculine by any standard if not quite Krell-like in that regard. Very guy to be sure. The VK-51SE's styling changes what was a pretty plain black faceplate to a more refined, very thick silver and black that I admire for reasons that have almost nothing to do with being a die-hard Raiders fan.

The volume knob is silver, with a blue vacuum tube fluorescent display that's reputed to be less noisy. Below the display are buttons to select the five inputs. In addition to a power/standby button, users can access the menus, mute the signal, invert phase and switch between mono and stereo operations. As you'll read later the VK-51SE is an exceptionally full-featured preamp, and both the front panel and the remote harness its capabilities.

All of the VK-51SE inputs and outputs are balanced, so adapters are necessary to accommodate single-ended sources. I bought a few pairs from BAT at $90/pair, both for inputs and for outputs.

There are five inputs, a tape output and two sets of main outputs. The AC cord is removable, although the VK-51SE ships with a heavy duty job that is just fine.

Inside—Construction and Design

The VK-51SE is built like a tank with high quality parts like Vishay resistors, WIMA capacitors, and a whole slew of those custom-made oil-filled caps used in Six-Paks and Super-Paks, which you'll read more about below. There are lots of parts, they are by name and repute high quality parts and, really, this thing looks like it costs $9K.

It features a dual mono design with separate solid-state power supplies with dedicated toroids for each channel. The transformers are way up in the front of the box to physically separate them from the audio circuits as much as possible.

The VK-51SE uses eight 6H30 "super tubes" in the input stage, gain stage and output stage. By contrast, the VK-50 uses twelve 6922s. Let's compare some of the other differences between the VK-50 and its two Special Edition successors, the VK-50SE and VK-51SE.

In addition to using fewer 6H30s instead of more 6922s, the VK-50SE added a "Six-Pak" of custom-made oil-filled capacitors to the output stage, and a "Super-Pak" of the same capacitors to the power supply, effectively doubling the energy storage of the

VK-50. The VK-51SE goes farther still, swapping out the power supply bypass caps with the custom oil-filled jobs, a switch Khomenko and company found improves sonics even more while increasing the capacitance to an eye-popping 350 joules.

BAT believes this translates directly to increasing the preamp's freedom from compression, and lack of strain at all volume levels across all frequencies. As you'll read the VK-51SE certainly has that, whether Six-Paks or Super-Paks are the reason. And as you'll read, dynamics isn't the only place the VK-51SE outshines its predecessor.

Volume control on the VK-51SE is interesting and inventive because Khomenko, like a lot of other designers, simply wasn't content to use an off-the-shelf solution.

A conundrum for a lot of designers is that chip volume control solutions with resistors on the chip offer a lot of very precise steps, which is desirable for obvious reasons, but is generally considered to be sonically inferior. Yet you want to have that kind of fine control, and remote control, but don't want the signal going through a chip.

Khomenko's solution is a shunt attenuator with a single resistor (per phase) per channel in series with the signal path. A microprocessor uses analog switches that change the resistor's value, which determines how much signal passes through the volume control and how much is attenuated by shunting, or bleeding it off to ground. As a result the VK-51SE has 140 discrete steps of 0.5dB.

Features and Functions

The BAT is loaded with features, including a remarkable ability to name inputs with a wide variety of characters including numbers and hyphens. Any input can be set for unity gain making the integration of a digital controller for surround sound quite easy. You can also configure a "maximum gain" setting for any input to protect the uninitiated from blasting the sound so loud as to damage your gear. Come to think of it, that's really to protect you.

The VK-51SE has a standby mode, but the tubes are heated up too, which means their life is being used up to some degree while it's just idling. It comes up to speed quickly when it's fired up from standby, but the tubes' longevity won't be that of the Calypso, which has the nifty standby that heats up everything except the tubes, which come up to speed fast anyway.

The optional remote (with which my unit is equipped) controls all of the front panel functionality, and is heavy aluminum finished in matching black and gray. It's as simple and easy to use as can be, and I simply couldn't ask for more.

Performance

I began my listening with the VK-50SE and the first striking thing about it is that break-in takes a while. It sounded pretty fat and opaque for several days, which BAT's Geoff Poor had conditioned me to expect. It took major leaps in transparency and dynamic life as it approached 100 and 200 hours, respectively, and another leap somewhere in the vicinity of 300 hours (I lost track by that point).

Normally I'm leery of manufacturers who tell me their gear only sounds good after hundreds of hours, as that sometimes means that's when you've listened long enough to forget what your reference gear sounds like. In the VK-50SE's case, however, the improvements were distinct at each point in time and made an impression. Beyond that point it settled into its performance envelope, and was always quick to come up to speed unless unplugged or completely turned off for a time.

The VK-50SE was a very lush, full-bodied tubelike sound. The low end was heavy, making things sound slightly slower in pace, gauzy in the middle and a little soft on top. Things sounded natural and accessible with a dense kind of image focus that was really quite dimensional. Dynamically it could be driven hard without losing its footing, and overall I would say it was a fun preamp to listen to if just a hair on the dark side if the rest of the system wasn't very much on the open and transparent side.

The VK-51SE changes virtually everything about the sonic signature, and all for good. If you've got a VK-50 or a VK-50SE I believe the upgrade is a no-brainer.

The VK-51SE is faster and much more transparent. Its sense of rhythm and timing is especially improved, and gone is the low end bloat that put some gauze in the midrange. The top is more extended, and the overall picture is more revealing and transparent. The VK-51SE is a substantially improved component over its predecessor.

The positives held over from the previous incarnation are the rich tonality, spooky image focus and density, all of which are more prominent and exciting as a result of the improved airiness and clarity the VK-51SE imparts.

The VK-51SE, according to BAT, uses no global feedback in the circuit topology, and it has enough cohesiveness and natural image focus to suggest that's true. Images are very much "there" in space, and layered convincingly front-to-back on the soundstage. In fact, only the VTL 7.5 surpassed the VK-51SE in depth and focused dimensionality by a small but discernible margin.

The VK-51SE is also a full-range preamp, with a very powerful and extended low end. Bass is deep but articulate with involving texture. The VK-51SE renders acoustic bass with authority and details, and drum kits are produced with punch and veracity. Listening to this preamp I was often floored by how each skin in a drum kit had a distinctive texture, a sonic signature of its own (old Fleetwood Mac records were great for that). To me that says good speed in the bass and mid-bass. In any case, the low end of this preamp is stirring, and it's more likely that your other gear will give out before it does.

As I've mentioned, one area in which the more expensive, complex and ambitious preamps separate themselves is in their ability to play louder without strain or grain, along with their ability to maintain dynamic range no matter how hard you're already pushing. The VK-51SE will drive your system (and I do mean drive it) to high volume levels and hold itself together as gracefully as anything I've heard while doing so.

It's an interesting phenomenon and I'm not sure what the mechanism is. Is it distortion that gets amplified along with the signal becoming audible at higher levels? Power supply? Who knows? What I do know is that if you like to blast Metallica at 11 or like trying to reproduce the full measure of a symphony orchestra's sonic output, the VK-51SE is your huckleberry.

While I typically aim to do neither of those things with my system, I do occasionally let it fly with some heavy rock, and even when I listen to small jazz ensembles I appreciate the dynamic swing the VK-51SE has along with its ability to deliver delicacy and inner detail at more realistic volume levels.

Comparatively speaking, the VK-51SE is right in there with the upper echelon of the preamps reviewed in this issue. In my review of this preamp for *The Absolute Sound*, I mentioned that the Ayre K-1x has what I perceived to be slightly better dynamic contrast, but that I've consistently preferred the VK-51SE's accessibility and musicality, which lets me forget I'm listening to a system and gets me more connected to the music.

I stand by that, and reiterate that this is a preamp that I could live with, and indeed have lived with and made excellent sound that I could turn on and listen to night after night. And what's more, it's revealing enough that I've used it to determine the sound quality of virtually all the audio components I've reviewed for the last couple of years.

Perhaps the most interesting comparison, however, is with the Audio Research Ref 2MKII, a preamp that's very similar in price and scope. How does the VK-51SE hold up against the Old Guard of audiophilia? "Very well" is the short answer. I don't think the differences between these two products are qualitative as much as matters of taste.

The VK-51SE is more full-bodied and more extended in the bottom end; the Ref 2MKII is leaner, a little more forward, and in some ways more exciting. As noted in the ARC review, the Ref 2MKII has a more dynamic, engaging sound than just about anything out there, but part of that sound comes from a slightly lean character that highlights the midrange just enough to emphasize leading edge transients. It's not that that's false, just a little technicolor.

I think the VK-51SE images in a more cohesive, holographic fashion. The Ref 2MKII, on the other hand, has more snap and jump and is spooky-good with vocals, and is a bit airier and more extended on top. You have to wait for the BAT to come to you a little bit, while the ARC just jumps up and grabs you.

The BAT's proponents would say the ARC is lean and whitish, while the ARC's proponents would say the BAT is dark. I think that's a less sophisticated way of describing how they both sound and, as much of a cop-out as it is, I think there's some truth in both arguments. I could honestly live with either one, and I encourage you to listen on your own and decide which floats your boat.

The BAT has never so much as hiccupped during the entire time I've used it. No glitches, no burps, no trouble of any kind for a solitary second. No blown tubes, nothing but fun and great sound.

I'm not suggesting this isn't true of the other products reviewed here but, while the other products were used for a typically short review period (three months or so), I've experienced the BAT for a much longer period. If my experience is any indicator, this is a tube product with solid-state reliability.

Conclusion

The BAT VK-51SE is an excellent preamp by any measure. It has grip and drive with dynamics that seemingly have no limit. I know from experience that this is a preamp that I can listen to day in and day out, and it's a product that can connect you to the music in a profound way. APJ

Richard Hardesty comments on...

Balanced Audio Technology VK-50SE VK-51SE LINE STAGE PREAMPLIFIERS

I felt that the VK-50 had compromised bass performance, which seemed to slow the pace of the musicians and make it more difficult to follow the rhythm of the music. I found this characteristic to be unacceptable.

The VK-51SE is much improved in this regard. It's an excellent preamplifier with deep and focused imaging that many will find to be completely gratifying. The BAT failed to provide complete musical satisfaction for me. APJ

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LINN LINTO PHONO STAGE

by Shane Buettner

It's no secret that Linn practically invented the concept of a high-end turntable. To this day their LP12 is available as a thoroughly updated high-end table, and one of the LP playback accessories you can buy for use with it today is the $1,600 Linto phono stage.

Both Richard Hardesty and I have been using this mysterious little black box with our LP12 rigs for the last few years. It's still available, and our experience with it indicates that it's a product you should know about and, ironically, one that we'd like to know a little more about, as you'll read.

Outside

It's a classic black box with a steel chassis and aluminum top cover. It weighs just a few pounds with a black power button on the front. On the back panel there are two single-ended inputs and a single-ended input with a ground. That's it!

Inside

Wish I knew! As an unwise man (Sgt. Schultz) once said, "We know nothing!" Richard contacted the people at Linn, who were unwilling (or unable) to provide any information about how this product actually works, referring us back to the owner's manual.

Here's what we know from that: Linto uses the Linn Brilliant switch-mode power suppl; it has 64dB of gain; and the input impedance (analogous to cartridge loading value in this case) is 150 ohms. It's made for moving-coil cartridges only.

The manual states that high amounts of feedback are bad, but doesn't really say if the Linto uses any or, if so, how much. It also talks about typical cartridge loading networks being wasteful of signal power and noise-inducing, and says that the input stage direct-couples the cartridge's signal to the amplifying transistors without discussing exactly how that's done.

Inside the box there are two cans—one that obviously houses the Brilliant power supply and another that's placed over about 2/3 of the audio circuit board. But I don't know what's in there.

Performance

The Linto is a very "Linn-sounding" phono stage, which is to say it's got musical and rhythmic drive and rightness like you can't believe. It's as detailed and lively as can be without veering into brightness. It throws a deep and wide soundstage with exceptional focus, extending beyond the far sides of the speakers. Sound overall is very clean and musical, but also high in resolution and revealing without losing sight of the musical message in its resolving power.

Although the Linto's got an extended, airy and open top end, it does tend to emphasize surface noise too. It's tonally pretty neutral with one exception: the bass is tucked back just enough to let the midrange take on some extra speed and clarity. The Linto is just plain exciting and fun to listen to and is exceptional by any standard.

Comparing the Linto further with the Aesthetix piece, it's clear that Rhea has more natural and extended bass. Piano's lower registers have more natural weight and body with Rhea, which suggests to me that it's the truthful one in that regard. Rhea isn't quite as spectacular on top, but has more spatial dimension and focus and less propensity to emphasize audible surface noise. Rhea sounds a bit thicker and less exciting in some ways, but overall I think it's more truthful. While the Linn is liquid enough, tubes are a different story. That's strictly a matter of taste, of course.

I've lived with the Linto for a few years, and in that time I've never enjoyed listening to music more, especially LPs played through it. And the Rhea isn't the only higher priced competitor that the Linn has held its own with. An industry acquaintance brought over a phono stage of high-end name and reputation that is said to have once carried a retail price of nearly $6K. This was not an apples-to-apples comparison—I don't know how the "Brand X" stage was configured with respect to its loading network, or if it was even adjustable.

The listening was brief but, for what it's worth, the Linto killed it and danced on its grave. Even if I like the Rhea a bit more, the Linn is damned good and it's only $1,600. Non-Linn owners will have to figure out if Linto's rigid loading configuration works for them, and if it does, and it's what you can afford, you can buy a Linto without looking back.

Conclusion

The Linn Linto at $1,600 is more than competitive with products that cost substantially more. In some cases it will ruthlessly expose the pretenders. It's not the most flexible thing out there, but if its configuration works in your system you'd better check it out before you spend more. APJ

Richard Hardesty comments on...
LINTO PHONO STAGE

The Linto is quiet, dynamic and rhythmically involving. It leans slightly towards the bright side and raises the prominence of surface noise a little. I have used this product as a benchmark and compared it to prestige components at ridiculous prices. The Linn always holds its own and often wins.

The Linto is best matched to preamplifiers with no tendency towards brightness and cartridges that are suited to its unalterable input impedance and gain. Works great with my Linn Arkiv. APJ

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MAGNUM NINETY-NINE

by Shane Buettner

For the last two years Rogue Audio has been making sound that stands out at CES and other industry shows with their Magnum Ninety-Nine preamp and Zeus power amp—in conjunction with Meadowlark loudspeakers and Acoustic Zen cables.

I was shocked to find out that the preamplifier at the front end of these systems costs just $2,395 as a line stage, and just $2,895 with an integral phono stage. This is a product that got my attention, so just who are these self-proclaimed "Rogues" of the audio industry?

Rogue Audio was founded in 1994 and, although it began worldwide distribution as recently as 1999, it boasts a sales network of over sixty dealers. Mark O'Brien is president, general manager and primary designer of all Rogue Audio electronics. Mark has a degree in science and an MBA as well.

All manufacturing is performed by "skilled technicians" right here in the US of A. And Rogue is very upfront and focused on the mission to design and build tube amplification products. No solid-state components, no source components, no multichannel, just two-channel with tubes. And that's pretty much right up my alley, so let's take a closer look.

Outside

The Magnum Ninety-Nine will get your attention right away—you won't find many two-box preamps at this price point. The external box, connected by two umbilical cords, is the power supply. The Ninety-Nine's look is decidedly spartan, but heavy with a very solid look and feel to the aluminum chassis and buttons that adorn the front panel.

You won't mistake the Magnum Ninety-Nine for a toy by any stretch. It's much more substantial than the ARC SP16L, which is plastic in feel and weighs almost 1/3 what the Rogue does. And Rogue throws in some machined arches for aesthetic effect.

All of the Rogue's ins and outs are single-ended—five inputs, plus a record loop and a single set of outputs. On the front panel you'll see a trio of knobs smack in the center of the unit. The largest middle knob is volume, flanked on one side by a gain adjuster and an input source selector on the other side. Complementing the power button are push-ins for mute, record loop, and mono operation.

The look and overall feel of this preamp is very straight-ahead and businesslike. Keeping with that theme, the silver aluminum-finished remote is volume control only. At this price point it seems that there's a choice to be made between convenience features and functions and trying to eke out as much performance as you can, and Rogue, as their outsider company name suggests, has taken the latter road, and good for them!

Inside

The outboard power supply is solid-state using a toroidal transformer. The stock Ninety-Nine's input stage and gain stages are populated by four electro-harmonix 6SN7 tubes. The volume control is an Alps pot, and other high quality parts used in the Ninety-Nine include Vishay resistors, and Mundorf coupling caps from Germany. And according to Rogue, silver wiring is used throughout. Another point of interest is that the Magnum Ninety-Nine utilizes a zero global feedback design.

There are two significant options that can be purchased with this preamp. The first is a unity gain home theater bypass circuit that can be ordered for just $75. That's a no-brainer if there ever was one. In addition, Rogue offers "new old stock" RCA and Sylvania tubes that can be purchased at the same time with the preamp for $100 for a set of four, or later for $150. I enjoyed the sound of those tubes enough to recommend this option as well.

Performance

The stock Magnum Ninety-Nine, with its Electro Harmonix tubes, has a retro, chewy brand of tube sound that could be considered "classic" tubes. In other words, there's a lot of midrange with little extension at the top or bottom. It sounded musical as hell, but certainly soft and round.

Swapping out the Electro Harmonix tubes for the Sylvanias opened up the top end quite a bit, but also resulted in a midrange character that I found a little on the dry side with flat perspective front to back. It was preferable, but still not quite where I wanted to be. With the RCA tubes the Magnum Ninety-Nine sounds much more to my liking with the added sparkle increasing top end air but also adding some depth to the soundstage without any of the slightly dry, granular sound I heard with the Sylvanias.

The fact that you can change out the tubes to tailor the sound of the Magnum Ninety-Nine is kind of cool for a hobbyist. Once the RCAs are in, its ability to focus at the sides of the soundstage is strong, and the front-to-back depth and layering are improved as well, to a degree that's very respectable, if not quite the equal of the more expensive preamps. The bass is still full and round, and not the last word in extension, but midrange in particular takes on a new and more vibrant life.

Images are solid, focused and dimensional—that's this preamp's strength, and a trait that speaks to me a lot. The sound is well integrated and coherent, and the dynamics are very punchy and authoritative. At this price point there are compromises that must be made, but Rogue has made the choices shrewdly. This preamp is eminently listenable, with none of the analytical colorations that impose on the sound of the SP16L or Emmeline.

Vocals sound wonderful, if perhaps just a hair warm, and this preamp is very easy to listen to. If there's any issue with it, it's that you'll probably find yourself mating this preamp with components that are a little light on bass to maintain the airiness the RCA tubes impart. You definitely wouldn't want to mate it with anything that's warm or has a big bottom itself, because that would surely close things down some.

I've already given some indication of how the Rogue's performance compares to the SP16L, but let's talk about how it compares to the other preamps here. First, listening to the Rogue on its own, you'll want for very little. When directly compared to the Ayre K-5x and the VTL 5.5, the Rogue sounds a little rolled-off on top, with less extension at the low end. In the midrange the Rogue holds its own.

The VTL is a bit more vibrant, and has a little more snap, but then the VTL has more of that than even the more expensive preamps, and it costs a full $1,000 more than the Rogue with the RCA tubes. There's no shame in that whatsoever. The Ayre is just a different animal. It's more ergonomically refined than the VTL or the Rogue, and it is an excellent solid-state preamp. Your preference in that regard is likely to come down to your response to tubes versus solid-state—the Rogue and the Ayre are both solid performing examples of each school of design.

If subjected to torture I'd admit that the Ayre has more overall transparency, but I'd also say there are listeners who'd rather live with the Rogue's inviting and natural tube sound.

Conclusion

The Rogue Audio Magnum Ninety-Nine is an unqualified recommendation at $2,495 with the RCA tubes, which is to say I'm not quite as sold at $2,395 with the stock Electro Harmonix tubes.

The "retro" tube sound I heard in the base configuration softens things up enough that my chief concern would be that you'd need to mate it with an amp that's colored in the other direction—lean and rather mean.

Rogue spent the money wisely on this preamp—it's in rugged construction quality and in components and design choices that enhance sound quality. Even at $2,495 you'd have to spend more money to get better performance, making this preamp a very solid value and a great place to start your journey into hi-fi. APJ

Richard Hardesty comments on... MAGNUM NINETY-NINE

The Rogue preamp is very sensitive to the type of vacuum tubes that are installed. It is furnished with tubes that provide classic tube sound, reminiscent of the Audible Illusions products of the 1980s. The optional tubes move the sound substantially towards greater accuracy but the Rogue still sounds slightly more "tubey" than neutral.

The Rogue preamp could effectively soften a system using an amplifier tending to sound slightly sterile, like an ATI or Bryston. APJ

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EMMELINE CA-2 & XR-2

Ray Samuels Audio

by Shane Buettner

I actually hadn't heard of this gear until Dick informed me that it had been acquired for review in the ***Journal****. Since I'm the review guy, I said, Sure, send it on over!*

All I knew going in was that these were products sold direct on the internet, and that they retailed for somewhere in the neighborhood of $1K each ($995 for the CA-2 and $1,050 for the

XR-2). When the box from Dick arrived I found four black boxes from Ray Samuels inside—both the CA-2 and XR-2 come with separate dedicated power supplies. In both cases it's the Emmeline power supply, which is also sold separately as an aftermarket device for $325. So, what can $2K buy you in a line stage/phono stage combo?

Outside

Not a lot to report here, just small black boxes. The CA-2 has two single-ended inputs, a ground, and one single-ended output. The front panel has a single button that switches between the two inputs and a volume knob. The XR-2 phono stage has one single-ended input with a ground and a single-ended output. No remote control here, this ain't a frills product.

Inside

The CA-2 and XR-2 are both built on Analog Devices' AD 797 IC op amps. All the connectors appear to be of high quality and, according to Samuels' web site, all the components are soldered by hand. The CA-2's quality parts lineup includes strict tolerance (1%) film resistors, polypropylene caps and both the CA-2 and XR-2 use mil spec PC boards. Obviously the power supply and all gain devices are solid-state.

The Emmeline power supply uses a toroid with separate windings and bridge rectifiers for each channel. RIAA is implemented with active circuitry and the CA-2 uses a Noble pot for volume control. One thing is for sure, this is a very short and simple signal path without a lot of places for things to get tripped up.

The XR-2 has virtually no controls or knobs, but there is some hidden flexibility. Opening the box and flipping some DIP switches allows a slick and easy change of cartridge loading to any of the following settings: 47k, 470 ohm, 100 ohm, 80 ohm, 50 ohm, and 30 ohm. If you're a real fancy pants there's an open socket for a resistor of your desired value.

Performance

I started with the XR-2 phono stage with my Linn LP12 rig. I had recently installed Linn's new Akiva cartridge in place of the Arkiv I've been using for the last two or three years.

I'd determined that the 250 ohm setting worked best with the Aesthetix Rhea, and I started with the loading at 470 ohms with the XR-2 only to find it a little hot and flat. It sounded whitish in tone and lacking in dimensionality, so I moved to the 100 ohm setting, which remedied those issues right away, increasing front-to-back depth and mellowing out the sound a bit.

The XR-2 performance is very good overall, and gives up surprisingly little in comparison to the Linn Linto that's my reference. It's very quiet both in terms of background and surface noise, and has a very fast sound. Extension at the frequency extremes is very good and the deep bass is especially robust.

Although the deepest low end is big and heavy, the mid-bass is a little lean and/or the midrange is just a little on the forward side. The lower notes of a piano, for instance, sound just a little light in foundation in spite of the impression made by the low bass.

Overall resolution is solid, and the soundstage has good front-to-back depth and focus at the sides. My biggest issue with the XR-2 is that its sound is just slightly cold and not as musically gripping compared to the Linn Linto. Images aren't as dimensional, and the players aren't quite as focused and separated on the stage.

So, the Linn Linto is better, but at $1,600 is it $600 better? Actually, yes, although I think that says more about the Linto than the XR-2. The Linto more than holds its own against competitors that cost several times its already higher price tag, and is truly a product that transcends its price point by a good distance.

The XR-2 isn't quite as quiet with good vinyl, but the Linn emphasizes surface noise more and is more open and extended on top. On top of that the Linn is simply second to none at communicating the emotion of the music. Relatively small nits, but significant improvements in all for $600. Nevertheless, if $1K is your price cap, the XR-2 is a very solid choice.

At first pass the CA-2 line stage gets your attention. It has a tonal signature that's very similar to the XR-2, which is to say respectable, but with a couple of key differences.

First, the low end is leaner in comparison, but sounds just about right to me, although it slightly emphasizes the dry character in the midrange. Recordings that are lean already can sound downright bleached of tonal color. The CA-2's only other apparent coloration is a top end that's just a little reticent. Piano decay and cymbal splashes sound a little truncated with less air than I'm used to from more expensive products.

This isn't surprising—the other preamp I've reviewed in this price range, the Parasound Halo P3 ($800), had similar characteristics. The key difference is that the Parasound had a fat bottom end that warmed up the midrange just a little bit. But it too rolled off on top and didn't quite open up like you'd ultimately want.

Where the Emmeline falls short of my memory of the Parasound is that it displays more of the same analytical character I heard in the XR-2, along with imaging that's just not coherent and focused in space. Piano notes and fundamentals sound disassociated in an unnatural way. Soundstage width and depth are fine, but the instruments and vocalists themselves just aren't cohesive in a convincing fashion.

And then there's this. My memory indicates that the Emmeline and Parasound Halo P3 are quite comparable in sound quality. The Halo costs less, and is decked out with some features that matter, including six sets of inputs and remote control. In this case I don't feel like you get more buying direct.

Conclusion

The Ray Samuels Audio Emmeline XR-2 and CA-2 are both solid performers. In both cases competing products from established manufacturers with traditional dealer networks standing behind them are right in there with them, and beyond them in some key aspects of performance.

The XR-2 stands on its own even in light of such a comparison and can be recommended at $1,000. The CA-2 sounds a little too much like op amps in a box for my taste. Most of its sonic characteristics sound fine taken into account separately, but that's part of the problem. The pieces of the musical puzzle never quite fall into a musical whole enough to make me forget the gear and get emotionally involved. APJ

Richard Hardesty comments on... EMMELINE CA-2 & XR-2

Ray Samuels Audio

The CA-2 is a bare-bones preamp that costs little and performs well for the money. The XR-2 is a very good phono stage that costs very little. The Linto is better, features discrete components and costs very little more. APJ

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VTL 7.5 REFERENCE LINE STAGE PREAMPLIFIER

by Shane Buettner

The first in VTL's new "Reference" line of products, the VTL 7.5 is a preamp I've had an eye on since I saw a preproduction model with a clear plastic lid at CES a couple of years back.

Looking inside and learning how the 7.5 is made, it's hard not to let the gear head side of me take over. The proof is always in the listening, but knowing that a piece is clearly a smart run on the state of the art gives me a charge.

At a hefty $12,500 with no phono stage, the 7.5 is a two-box hybrid design that uses mostly solid-state components and just two tubes in the gain stage to provide what VTL refers to as "harmonic opulence."

This preamp keeps all the "dirty" (read noisy) stuff, like the control microprocessor and power supply, in a separate box from the audio circuits. As you'll read below you'll also get as feature-laden a preamp as there is out there. The very idea of reviewing a component like the VTL 7.5 is an opiate to a reviewer like me.

The Two Boxes

First, don't stack the 7.5's two boxes—they need ventilation. Make room for them somehow, some way. The smaller of the two boxes that comprise the 7.5 is the control box with the display and volume control knob and buttons on the outside. In addition to selections for input sources, tape and processor loops, there are also buttons for inverting phase, balance and muting. The mute button also doubles when held down to disable an unused input entirely.

The solid-state power supply and control electronics are on the inside of this box, with two toroids that look like they'd be just as comfy in a power amplifier. An umbilical carries power to the audio circuit box, and two control cables, one for each channel, with SCSI (physical) interfaces carry control signals that switch inputs and shift relays that switch discrete resistors for the volume controls of each channel. Only two relays and two resistors per phase, per channel, are in signal path at all times.

The audio box is fully balanced from input to output in a mirrored, dual-mono topology. There are four labeled inputs and two tape loops and two sets of outputs, all with balanced and single-ended connections. The only tubes in the box are two 12AX7s used in the input/gain stage, which is direct-coupled to a MOSFET output/buffer stage. This makes for longer tube life and less expense when they do finally need replacement—very user-friendly, as the VTL motto says.

The output stage employs high-output MOSFETs creating super-low output impedance (rated at 25 ohms with a maximum of 150 ohms at 10Hz) making it a match for virtually any amplifier or cable run imaginable. The volume control has 95 steps of approximately 0.7dB performed by discrete relay attenuators. The overall topology is short and simple, with a very low 1.5dB of negative global feedback.

As you'd expect with a product of this caliber, the parts and construction quality are rock solid and of the highest order. The only thing I'd interject, although I feel bad about mentioning something so seemingly superficial, is that for this amount of money some consumers might expect a little more tactile sensation on the fit & finish front. Although the 7.5's two boxes are finely finished in aluminum (especially in black), at least one big, bitchin volume knob would be good—the 7.5's volume control is pretty much a nondescript dial.

Features and Functions

The VTL 7.5 is incredibly well thought out and is one of the most versatile products I've ever encountered in any category. For starters, VTL is clearly aware that two-channel audio isn't the only game in town and that many listeners (like me) have built, or may want to build, a home theater system around their music playback rigs. The 7.5 is ready. Any of its inputs can be configured for unity gain for processor pass-through.

There are four programmable trigger outputs for amplifiers, controllers, screens, et cetera. And, on top of that, the remote commands have been designed to operate in a discrete fashion for each command—separate on and off commands instead of a single button that switches between the two states, for example—so that the 7.5 fits right into a Crestron/AMX-style automated control system.

Beyond that, holding down any of the input selection buttons on the front panel switches that input between balanced and single-ended operation and, of course, the 7.5 remembers the selection. An LED at each input glows blue for balanced and green for single-ended. Additionally, holding down the mute button with a particular input selected will disable the input, and holding down the input and the mute button will reenable.

Volume offsets can be established to compensate for varying gain levels between source components with a simple combination of button pushes, and another combination of button pushes can lock out the programming features so you don't walk into your listening room one day to find that someone's messed with your setup once you have it all dialed in.

The 7.5's remote is a heavy aluminum job and the only issue I have with it is that the buttons are all the same color and size and the font explaining each button's function is tiny black scrawl. Beyond that, all of the inputs can be directly accessed and, in addition to balance, mute, and phase inversion, there is a "fade" button that gently drops the volume down from its current level to 20dB. Also, a "soft" start-up sequence fires up the 7.5 over 90 seconds with a countdown on the front panel displayed for the duration. Among other things, this is supposed to be easier on the tubes for increased longevity.

Performance

The 7.5 has more neutrality, clarity, and inner detail than I've ever heard from any preamplifier component. It's free of distortion and color of any kind and yet is so relaxed and at ease with the way it goes about its business that you almost take it for granted.

When I had the 7.5 in my system it simply wasn't making the impression on me in the same way that the ARC or even the little VTL 5.5 were. I kept noticing astonishing little pockets of detail and things that weren't quite apparent with any of the other preamps, but it's so completely neutral that none of these things hit me over the head. And hey, sometimes reviewers need a product to stand and wave its hands in the air and say, "Look at me." The 7.5 doesn't do that. It's confident waiting for you to figure out how damned good it is.

The first thing that started creeping up on me is the sheer delicacy of its resolving power. Details are so remarkably free of any form of grain or glare or anything resembling stridence. It's completely nonmechanical and antithetical of any form of "hi-fi" sound. It's smooth, clear and easy, and yet you can look more deeply into the music, which hangs together in between and around the speakers (never at them) in an entirely convincing fashion.

Spatially, none of the other preamps reviewed here were in its league. The 7.5 pushed vocals and instruments way back in the soundstage, with layers of music in between in clearly delineated spatial planes. Musical events were focused in space in a more precise and holographic manner than any of the other preamps could deliver.

While what's been said already applies mostly to the midrange, let me assure you that extension at both frequency extremes is nothing short of remarkable. It's unbelievably clean in both directions and it just keeps on going. Piano keys and cymbals splash and light up the soundstage, and then the decay just floats onward and upward naturally and convincingly.

With the deepest bass the 7.5 seemed to have an extra octave of extension that the other preamps didn't. And yet, it was clear that this was not an emphasis of any kind, just clear authoritative and articulate bass that will go down as far as your speakers allow.

Most of these revelations came to me only after the 7.5 had gone and I was left wondering if the music had gone with it! Listening to the other preamps in the 7.5's wake made them sound woolly and colored in comparison, and altogether less refined. The 7.5 is a remarkable piece, lacking only in the dynamic contrast and authority that the ARC and the little VTL (!) have in spades. I can't find a reason why this would be, especially since the VTL 7.5 had as black a background as I've heard—I mean pitch black and lined with plush luxuriant velvet.

Since the 7.5 was still coming into its own when it had to leave I was left wondering if that swing might have been a week or two away in break-in. It's hard to imagine that the bigger, outboard power supply of the 7.5 couldn't match its little brother's dynamics and I hope to follow up on that aspect of its performance in a future issue.

Conclusion

Simply put, the VTL 7.5 is the finest preamplifier I've heard. It sounds exactly like what it's intended to be: the ultimate combination of the noise-free clarity and focus of solid-state, with just enough tube sound to impart some of the vividness, life and musicality that will keep you listening for hours on end.

As if all that weren't enough, the 7.5 is the most fully-featured preamp I've seen, with extraordinary capability for assimilation into the most complex music/cinema systems. The VTL 7.5 is the pinnacle of technical innovation and performance in preamplifiers. APJ

VTL TL5.5 LINE STAGE PREAMPLIFIER

by Shane Buettner

Vacuum Tube Logic's tag line is "Making Tubes User-Friendly" and their $3,500 attempt at doing just that is the TL5.5 line stage. This tube preamp has six inputs, one of which is a phono input that can be configured with an integral phono stage for an additional $750. As you've read, we were fairly wowed with the big VTL preamp, which appears to have put its stamp on the state of the art. So, how's the little brother?

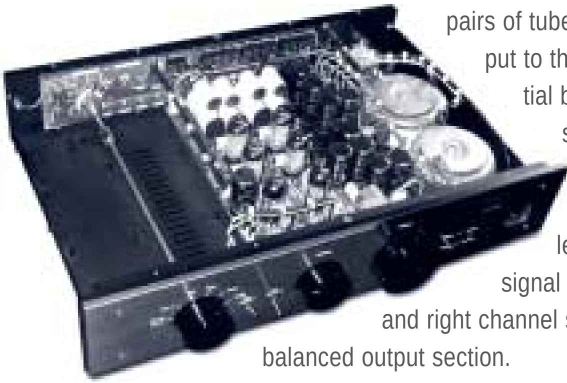
Outside

Good looks aren't what you get for the dough you throw down on a VTL 5.5. Its look is decidedly retro, and is utilitarian even by that standard. On the other hand, the performance-oriented among you will be stoked that what's inside is what you've paid for. There are three large knobs for input source selection, balance and volume, in addition to some toggle switches that switch in the tape and surround processor loops, and another double-duty toggle with a status LED that inverts phase and mutes the preamp.

The 5.5 has five line level inputs, plus a unity gain processor pass-through input, a tape loop in/out, and an input for the optional phono stage with a ground (all single-ended). There are two sets of single-ended outputs and a single pair of balanced outputs. And, of course, the AC cord is removable.

Inside

The 5.5 is a dual-mono design with two surprisingly large toroids in its solid-state power supply. There are six tubes inside the 5.5. The input stage runs directly to two 12AX7 tubes, which are the first differential gain stage. The two other



pairs of tubes are output to the differential buffer stage—one tube for each leg of the signal for the left and right channel so it has a balanced output section.

When phase is inverted, different tubes are selected, and when you use the single-ended outputs just one signal leg per channel is selected. The volume control is a continuously variable, laser-trimmed, motorized pot by Alps. Although much of the circuitry in the 5.5 is balanced, the inputs aren't and the primary reason is that the 5.5 would have to be much more expensive in order to have a properly matched, fully differential volume control.

Another thing you'll notice in the 5.5 is the expanse of real estate along one side of the box. That's for the $750 optional phono stage, which adds six more tubes of its own to the box. Two of the tubes are used for left and right channel gain for moving coil cartridges, and two others are used for l/r channel gain for moving magnet cartridges. The RIAA network is active with a feedback loop varying gain at different frequencies. The final two tubes feed the buffer stage and the phono stage as a whole shares the 5.5's main power supply.

The 5.5's remote control has volume, mute and phase inversion but not input selection or selection of the tape or surround processor loops. Those must be addressed on the front panel. Further, with a continuously variable pot volume control, fine volume adjustments from the remote are difficult to make. But what the 5.5 lacks in convenience it makes up for in pure performance, as you're about to find out.

Performance—The Life of the Party

The first thing that quite literally jumped out at me with the VTL 5.5 is that it has dynamic swing and contrast up the wazoo. The bass is big and robust without bloat, and the midrange is full and rich with a very airy and lively form of tube life. It has air and transparency up top and is a driving force of musical intensity and, well, fun!

This preamp had me pulling out my records one after another. Not audiophile records to dissect and discern what it was doing and what it wasn't doing, just records of stuff I like, old and new, all genres. All the adjectives I can come up with revolve around snap, crackle and pop, as well as musical bloom and life. The music erupts to life with authority and shocking amounts of texture.

All the 5.5 gives up to the big boys that cost 2 to 3 times more are small amounts of microdetail, a diopter of spatial focus and image holography, and greater ability to play louder without strain.

The 5.5 is a little rough around the edges when it's pushed hard. You'll hear just a bit more musical expression and articulation in an A/B, head-to-head with one of the big boys but that's about it. And the system in which you perform this comparison must be at the very tiptop of its game to reveal these differences.

Unless you've got something right next to the 5.5 that goes to the nth degree, you're going to be too damned busy listening to music to worry about any of that. It's spatially defined, especially front-to-back, and the soundstage is expansive. This thing is just right on the money. And I mean that. This isn't a good, entry-level preamp, it's a damned good preamp, period.

As the review period was coming to a close, VTL sent over a 5.5 outfitted with the optional $750 phono stage. The loading was fixed at 750 ohms, and was used with my Linn LP12 table, and Linn's Akiva cartridge which has a specified output of 0.4mv. The 5.5's phono stage seemed to have very high gain.

A fair amount of "tube rush" was audible, and while it diminished significantly once the needle was dropped onto a record it didn't disappear entirely. The solid-state Linn Linto was dead quiet in comparison, and the much more expensive ($4K) Aesthetix Rhea, which also uses tubes, was also much quieter through the 5.5's line stage than was the 5.5's integral phono stage. The Rhea's tube noise disappeared completely once a record began playing, where the 5.5's phono seemed to let more noise through during quieter passages than I wanted to hear.

Beyond that, the integral phono sounded like the 5.5, which is to say airy, lively, and fun, through the midrange and on top, with snap and musicality to spare. The bottom end is a bit bloated and lacking in articulation.

The 5.5 didn't quite come to life with its own phono stage the way it did with the Linto, which is lighter in the bottom but also more articulate and fast. There is definitely a good degree of that vivid tube sound that I like so much in the 5.5 with other sources, and perhaps enough to recommend the integral phono at an additional $750. I found more magic pairing the 5.5 with the Linto at $1,600, so in that case you do get a little more for the extra bucks, in my opinion.

Conclusion

Except for Ayre's K-1x, there isn't a single preamp I've listened to that beats the VTL 5.5 for apparent dynamic swing and impact. Its only rival is the ARC Ref 2MKII, which costs nearly three times as much.

While some of my initial fascination with the 5.5 was what it was doing relative to its price point, that only lasted during the initial listening period. Even after extended listening to the other excellent preamps reviewed for this issue, I could always sit right down and listen to the 5.5 with a big grin on my face.

I enjoyed it in its own right under all circumstances. The VTL 5.5 is one of the most continuously engaging components I've ever experienced. If you're shopping anywhere near this price point and don't check this thing out, you're just plain nuts. APJ

VTL Richard Hardesty comments on... 5.5 & 7.5 LINE STAGE PREAMPLIFIERS

The VTL 7.5 combines neutrality and transparency with a unique blend of accuracy and emotional satisfaction that beats all rivals. It has all the transparency and freedom from coloration of the finest solid-state designs and resolves every subtle detail in the recording. This resolution is combined with the most natural and satisfying presentation of music I've heard. Nothing in the recording jumps out and becomes more or less prominent to the listener, but substituting any other preamplifier allows the listener to become clearly aware of what's missing without the VTL 7.5 in the system.

The VTL 7.5 was the best preamplifier I've auditioned and it made the others sound slightly colored by comparison.

The VTL 5.5 was perhaps the most musically satisfying preamplifier in this group. It couldn't be singled out as the most detailed and revealing. It didn't have the most up-to-date cosmetics or the newest remote control features. It lacked balanced inputs and it used a motorized pot for attenuation. But the VTL 5.5 reached out and grabbed me by the soul and implored me to simply sit back and enjoy the music.

*This is perhaps one of the most enjoyable preamplifiers ever, regardless of cost. It gets the music right and faithfully follows the rhythm and dynamics of the performance. No musical nuance ever sounds wrong and the thrill lasts through many hours of extended listening. I urge every **Journal** reader to seek out and listen to the VTL 5.5. When you do, you'll know what high-end audio is all about. Finding out that you can buy one for just $3,500 will surely add to your excitement.* APJ

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What did we learn TODAY?

by Shane Buettner

It may seem at some points here that I'm balancing quite deftly in comparing some of the products, and hesitant to draw iron-clad conclusions. And that's because I am!

I'm happy to report that most of the preamps we've looked at here, particularly the ones that cost the most, are excellent in virtually all ways. While I think the VTL 7.5 stands apart in ultimate resolution of inner detail and depth/dimensionality, I can also say just as firmly that anyone could make an excellent sounding, high resolution hi-fi with no apologies to be made whatsoever using any of the following preamps: Ayre K-1x, Aesthetix Calypso, Audio Research Ref 2MKII, or the BAT VK-51SE. All of these products are outstanding performers with the features and ergonomic usability you'll need not only to have a kick-ass hi-fi but to build a seamless home theater system around that hi-fi if you desire.

For the most part, the differences between these products are simply different strokes for different folks. These things are just not as out there in the woods, not as obviously "right" vs.

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"wrong," the way a lot of the loudspeakers are, for instance. The one standout, in my opinion, is the Aesthetix Calypso, which costs less than half as much as some of the others on that list and, frankly, Jim White deserves credit for building something that's so competitive with the ultimate in performance at a fraction of the price.

At just $4K Calypso doesn't sacrifice any ergonomic features, and actually includes one or two that the higher priced competitors don't have and, for all that, Aesthetix deserves a tip of the cap.

Now, ferreting this out further, after having listened to these preamps, I'm of the opinion that the high-end of preamplifier performance starts at $3K with the Ayre K-5x, followed hotly by the VTL 5.5 at $3,500.

The Ayre K-5x is quiet, neutral, and sounds nearly as liquid as solid-state can (among solid-state preamps we've heard, only the K-1x betters it in that regard) and is fully featured and more ergonomically refined than the VTL 5.5, but the 5.5 has some tube magic and an engaging quality that's just infectious and fun as hell to listen to. Either way, these are two excellent choices in preamplifiers that can be made at $3K and $3,500.

The Rogue really isn't far behind either, and its position here says more about its expensive competitors than the Magnum Ninety-Nine's performance. The fact is, the Rogue hits a very attractive price point and makes music that's very hard to fault until you hear a more expensive competitor.

And today's tube gear sets the bar high, with far fewer sacrifices made in exchange for the tube magic than was the case in years past. The Rogue is solid, but so too are its higher priced competitors, which means you pretty much get what you pay for in preamps from $2,500-$4,000.

In addition, we looked at three stand-alone phono stages that excelled at their respective price points: the Ray Samuels Emmeline XR-2, the Linn Linto, and the Aesthetix Rhea.

The $4,500 Rhea is the most flexible and user-friendly, and I liked its overall sound the best. But the Linto holds its own and then some, and the Emmeline is none too shabby either. All in all, there are lots of choices to be made in high quality preamplification components and hopefully this survey can be a guide for you to find the gear that best suits your tastes and system. APJ

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