GRACE design

Our goal is to design, engineer and build timeless audio products which enhance the user's ability to capture, relate and understand the musical experience. There are many paths which lead to this goal, and its our strong conviction that all must be investigated, tested and evaluated. The following is a discussion of what we do to achive this goal and why.

ANALOG DESIGN

Amplifier Topology - Grace Design microphone preamplifiers are fully balanced, with transformerless inputs and outputs. The input amplifiers are based on monolithic, low noise instrumentation amplifiers. Gain core sections are transimpedance, or "current feedback" amplifiers, which have many important advantages over traditional "voltage feedback" op-amps.

First, transimpedance amplifiers have a nearly constant bandwidth over a wide gain range and are not prone to large-signal slew rate limiting, which results in improved dynamic performance. Compared to voltage feedback op-amps (which are found in most preamplifier designs), transimpedance amplifiers have better dynamic performance and are able to track complex harmonics without adding any metallic or "solid-state" character to the sound. Also, transimpedance amplifiers reveal greater detail and provide a more open, musical sound quality. Furthermore, transimpedance amplifiers maintain the same sound quality all the way up to the highest gain levels, which makes them dramatically superior for use with low output ribbon microphones.

Output Driver Stages - The balanced output stage of our preamplifiers is capable of driving extremely long cable runs and load impedances down to 50 Ohms. Transient performance is superb and is not compromised when operating in unbalanced modes.

Integrated Vs. Discrete - Our gain core integrates the balanced input amplifiers on a single die and we feel that the advantages of this over a traditional "discrete" layout approach are important to understand. With the integrated amplifier components in very close proximity to each other, the paths of critical signals are kept very short. As well, energy storage from stray circuit board capacitance and inductance is eliminated. Integration of the input amplifiers also provides superior thermal tracking of input transistors and feedback resistors. It's true that in the early days of integrated circuits, it was better to design a discrete circuit because it was not yet possible to create transistors and resistors of the same quality as discrete devices. But today the quality of integrated transistors and resistors allows integrated circuits to achieve a much higher level of sonic purity than even the best discrete designs.

Misconceptions about "Class A" operation - In the early days of solid state power amplifiers, class B push-pull output stages were prone to crossover distortion, which was particularly bad sounding. By changing the operation mode to class A, designers were able to eliminate this crossover distortion, but at the expense of dramatically higher power consumption and lower available power for the speaker. But subsequent advancements in output stage topology resulted in class AB mode, which dramatically reduced crossover distortion without the power requirements of class A operation.

In regards to low level electronics, such as microphone preamplifiers, class A or AB is essentially irrelevant in determining over all sound quality, as these are not power amplifiers. Modern low-level high speed integrated amplifiers are free from the problems of primitive solid state power amplifiers because they do not employ large geometry power transistors, and are not required to drive low impedance loudspeakers. Other design elements such as capacitor type, resistor quality, power supply grounding and circuit board layout combine to have a much greater net effect on sound quality - much more so than whether or not a design operates in class A or class AB mode.

Controls - Each gain control used in all our preamplifiers (with the exception of the m802 remote controlled preamplifier) consists of a precision switch wired with 1% metal film resistors. The advantages of a fixed resistor switch over a potentiometer for gain setting are substantial: as music signal flows through a potentiometer, the potentiometer's temperature modulates, causing its resistance to modulate. As the value of resistance changes, so does the amplifier gain, resulting in unwanted dynamic distortion. Most conductive plastic potentiometers have temperature coefficients of +/-1000ppm/ºC, which can cause considerable signal degradation at high amplifier gains. By comparison, a metal film resistor has a typical temperature coefficient of +/-50ppm/ºC, which represents more than an order of magnitude improvement in thermal stability.

Relays - To keep signal paths as short as possible, relays are used to perform phase reverse, input selection and input attenuator functions. Sealed, gold contact relays offer no signal degradation and are completely reliable (rated for over 50 million operations without significant change in contact resistance).

Power Supply - The power supply is the heart of any audio device. All of our products (excluding the Lunatec V3 and m101) incorporate robust linear power supplies. The IEC AC power entry module includes a power line RFI filter and provides switching for 100/120/220/230-240VAC. High efficiency, low radiated noise toroidal transformers are used for power conversion. All of the audio DC power supplies have two stages of regulation- one in the power supply and one on each audio pcb, which ensures that there is no interaction between channels. Phantom power is provided with a two-stage low-noise +48V regulator.

Grounding - Correct power supply and signal grounding is a prerequisite to high fidelity audio performance. Each of our audio circuit boards incorporates a low inductance copper ground plane which provides a very stable signal and power supply reference. For added noise immunity, the 5V logic/relay and LED power supplies have their own ground returns.

Passive Components - Selection of passive components is as critical as any other aspect of audio design. The quality of resistors, capacitors, inductors, connectors, wire, switches and even solder have significant effects on sonic quality. All of the passive components in our designs are selected for optimum sonic performance and long term reliability.

Capacitors have the most significant impact on signal purity. The only signal path capacitors used are for phantom power de-coupling, which are ultra-linear metalized film capacitors. The rest of the audio circuit is direct-coupled with DC servo control. There are no electrolytic capacitors in the signal path, which have poor temperature stability, high dielectric absorption, wide tolerances and a relatively short life span. Dielectric absorption causes a capacitor to retain a certain quantity of its charge when discharged, which then leaks out over time, causing a smearing of high frequencies, which can result in the all too common harsh, "solid state" sound quality.

By contrast, film capacitors have dramatically less dielectric absorption than even the best electrolytic capacitors, which results in much improved transient performance, enhanced spatial accuracy and low frequency realism. As well, film capacitors are available in much tighter tolerances which, in the case of balanced input de-coupling, is important for acceptable low frequency common mode rejection. The use of film capacitors instead of electrolytic types results in a much more natural, less "metallic" sonic character.

All audio related resistors are precision metal film types for low noise and superior temperature stability, and all critical signal path resistors are fabricated with copper end caps rather than steel, which helps to avoid any inductive reactance. Ferrite beads, common mode chokes, and highly stable polypropylene capacitors form an input filter to protect the input amplifiers from RF interference. Ferrite beads also protect the output amplifiers from any interference on the output wires.

DIGITAL DESIGN

Analog to Digital Converters - Our new A/D converters represent the state of the art in high performance conversion for the most demanding music recording applications. An A/D converter can be divided into three basic functional blocks; the analog input stage, the A/D converter, and the clock generation (or regeneration). With any high performance A/D converter design, all three of these areas require complete attention and must be perfectly realized to achieve true high fidelity audio performance.

Analog Input Stage - Our converter's analog input circuitry is designed to be simple and pure. This fully differential circuitry contains no electrolytic capacitors and utilizes precision 0.5% tolerance thin film resistors, which guarantees no signal coloration from capacitor dielectric absorption and resistor thermal modulation. Careful attention to grounding and power supply layout guarantees no sonic aberrations caused from unwanted interference from digital circuitry. The analog input amplifiers are powered from their own dedicated voltage regulators and each analog power supply rail is double regulated to provide extreme isolation from power line related noise. Driving the input of modern delta-sigma converters requires a low driving impedance and high current from the A/D input driver, and our experience from designing high performance microphone amplifiers with precision line drivers translates very well to this task.

A/D Converter - The A/D converter itself is the Cirrus CS5381. This converter employs a fifth order delta-sigma modulator and a low pass filter that exhibits very wide dynamic range and low latency. With a more gentle slope digital filter than other ADC chips, the CS5381 is

the most musical of the high end converter ICs and is capable of 120dB dynamic range with distortion and noise artifacts at less than -110dB. Separate power supplies for the analog and digital portions of the A/D converter ensure that the precision delta sigma modulator will not be affected by other surrounding circuitry. As well, the differential inputs provide additional input noise rejection.

Clocks - A perfectly executed analog stage and the best performing A/D converter is only as good as the sampling clock. Even the smallest amounts of clock jitter will readily degrade the performance of the converter by introducing non-harmonically related noise and distortion.

When operating on the internal clock, the A/D converter sample clock is generated by a quartz oscillator which has very low jitter. However, many applications require that the A/D converter be synchronized to an external clock source, usually in the form of a word clock, which is where the greatest challenge arises.

There is a common misconception that using the word clock input on any piece of digital audio equipment will result in better sonic performance. In reality, the vast majority of word clock PLL circuits will have more intrinsic jitter than the word clock signal itself and usually more intrinsic jitter than the PLL within the AES/SPDIF receiver chip. As well, these PLLs usually have no incoming jitter attenuation in the audio band.

Our A/D converters employ s-Lock™, a two stage PLL system to provide practically jitter-free recovered clocks with extremely high incoming jitter attenuation. The first stage is a wide lock range ultra low noise analog PLL, which can lock to incoming word clocks and Digidesign® Loop Sync signals with a +/-8% lock range. Rather than using standard CMOS based PLL chips for the first stage PLL, we use an extremely quiet, discrete VCO (voltage controlled oscillator). When using a quality external clock source, this PLL alone will provide a nearly crystal quality clock. The incoming jitter rejection of this PLL begins at 500Hz and increases at 12dB/octave with rising frequency. The wide lock range allows for the converter to operate in vari-speed applications without a sacrifice in audio performance, provided that the external clock source is relatively low jitter.

When the incoming sample rate is within +/-200 ppm of the selected sample rate, the secondary (our proprietary s-Lock™) crystal based PLL will lock to the first stage PLL, providing the quietest possible recovered sample clock to the A/D converters. The s-Lock™ PLL is capable of extreme rejection of incoming jitter, which begins at less than 0.1Hz and is better than 60dB at 1kHz.

While it is very easy (and economical) to provide "jitter free" sample clocks by routing the audio data through an ASRC (asynchronous sample rate converter), we believe that the convenience of these circuits does not outweigh the potential sonic degradation caused by additional signal processing. The performance of the s-Lock™ PLL renders even the worst case interface jitter artifacts to a level well below the noise floor of the A/D converter.

Digital to Analog Converters Like the A/D converter, a D/A system is made up of three basic blocks; analog output circuitry, the D/A converter, and the clock system. The analog stage and clock design has very similar requirements to the A/D converter design. However, the D/A analog interface has some special requirements. All of our products that contain reference quality D/A converters use Burr Brown DACs with current outputs. This important feature allows us to construct an ultra-critical current to voltage converter using transimpedance amplifiers. The use of transimpedance amplifiers assures that reconstruction of the analog signal is done without the non-linear slew rate limiting of regular op amps. While the benefits of this type of topology are not readily apparent in measured performance it makes a significant improvement in the clarity and resolving power of the DAC.

Our D/A converter products all employ a two stage PLL system with the same s-Lock crystal based oscillator as in the A/D converters.

SUMMARY

Creating audio circuitry which performs to a high level of measured performance is the starting point of our designs. Measurable performance parameters such as distortion, noise, phase and bandwidth are important, but music does not behave like a steady state sine wave test tone - it is an infinitely complex waveform. Our designs are always optimized for music and all design decisions are verified by ear, the most sensitive test instrument available.

We regard electronic design as a holistic process which should result in products of uncompromised sonic performance, complete reliability, and beauty. If all electrical and mechanical details are considered at every stage of development, the result will stand the test of time.