high fidelity audio products

GRACE
design

m906 5.1 monitor control system

high fidelity 5.1 and stereo monitor control for recording, mastering, film, broadcast and post

What we know: the landscape of pro audio has changed. Computer based production has greatly reduced the need to own a large recording console - minus one part - the master section. This change has generated a whole new genre of products - the monitor controller, of which there are now many offerings, from simple to full featured. Grace Design proudly presents the pinnacle - the m906 high fidelity 5.1 monitor controller.

The original concept for building the m906 is to provide high fidelity level control for engineers working with 5.1 surround playback systems. But from there, we get blissfully carried away and ultimately obsessed with producing the most full featured and highest quality monitor controller available.

At its core, the m906 is a purist level control for monitoring surround (or stereo) audio sources. But a closer look reveals a highly intelligent feature set that makes it a vital nerve center for any recording, mastering, or post facility centered around a digital audio workstation. It is configured as a 2U rack mount mainframe controlled by a sleek, powerful desktop remote control, with all audio kept in the mainframe and all control via the remote control.

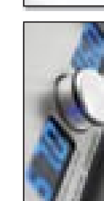
The m906 offers a full compliment of balanced and unbalanced analog 5.1 and digital 5.1 and stereo inputs, the latter featuring our latest generation, reference quality 24bit/192kHz DAC technology. All inputs and outputs on the m906 can be user calibrated for seamless integration into any playback environment with all types of playback equipment.

The m906's signal path has been painstakingly designed to deliver nothing short of the best performing, transparent audio available. All audio and control components are chosen for audiophile performance and critical reliability. Level controls, the heart of any monitor controller, are high performance precision devices which provide a 100dB range in silent .5dB steps.

Included as a standard feature is our critically acclaimed reference headphone amplifier circuitry (as featured in our m902), with one output jack on the mainframe and one additional output on the remote. This circuitry provides pristine audiophile headphone amplification for critical monitoring, mixing and quality control.

While the m906 boasts a remarkable complement of useful features, the real achievement for us is to provide all of this while maintaining the impeccable, transparent audio performance for which all of our products are famous. In its few short years in production, the m906 has found a home in countless of the world's premier audio production, HD production, post and mastering facilities.

The m906 is here to empower audio professionals with the highest performance and most comprehensive feature set that any surround monitor control system has to offer.



features

- multiple 5.1 and stereo analog inputs
- 24bit/192kHz digital 5.1 and stereo inputs- AES3, S/PDIF, ADAT and TOSLINK
- s-Lock™ phase lock loop sample clock regeneration for ultra-low jitter and rock solid digital stability
- precision speaker and headphone level controls with a 100dB range in 0.5dB steps
- every electrical and mechanical design element maximized for absolute purist audiophile performance
- all controls are built into an elegant, convenient desktop remote control unit
- all I/O and audio are routed in a 2U rack mount mainframe powered by an external 1U 1/2 rack power supply
- control up to three speaker systems - two 5.1 or up to three stereo
- m902 style high-current audiophile headphone amplifier built in with one output on remote control unit and one on the mainframe unit
- complete system level calibration (inputs, outputs, inter-channel balance, dim) and individual channel solo/mute
- balanced talkback microphone input and activation switch with additional external foot switch control jack
- optional downmix module supports all standard Dolby mix ratios
- optional AES Loop-thru feature provides four AES digital outputs for passing signals to other devices downstream
- fixed level 5.1 DAC output for direct audio transfers
- five year warranty on parts and labor

specifications

ANALOG IN

THD+N (@ 0dB gain, 1kHz)	
@+20dBu out	<0.001% (0.00075% typical)
@ 0dBu out	<0.003% (0.001% typical)
Frequency Response	
+/-3dB	3Hz - 250kHz
Dynamic Range	
20-22kHz bandwidth	112dB
Output Noise (20-22kHz bandwidth, -20dB gain)	
High gain mode	-86dBu
Low gain mode	-96dBu
Output Noise (A weighted, -20dB gain)	
High gain mode	-91dBu
Low gain mode	-99dBu
Phase Deviation	
20Hz to 20kHz	< 4°
Crosstalk any channel	
1kHz	< 100dB
Balanced Input CMRR	
60Hz	> 60dB
1kHz	>75dB
10kHz	>60dB
Gain Range	
Low gain mode (0.5dB steps)	-105.5dB to +20dB
High gain mode (0.5dB steps)	-95.5dB to +30dB
Channel Tracking Accuracy	
Any channel	+/-0.05dB
Maximum Output Level	
Low gain mode (balanced)	+15dBu
High gain mode (balanced)	+27dBu
Maximum Input Level	
Balanced inputs	+30dBu
Unbalanced inputs	+20dBu
Impedance	
Balanced inputs	30k Ohms
Unbalanced inputs	50k Ohms
Balanced outputs	300 Ohms
Minimum load impedance	300 Ohms

D/A CONVERTER FIXED OUTPUTS

THD+N	
1kHz, -1dBFS, 20-22kHz bandwidth	0.002%
Dynamic Range	
20-22kHz bandwidth	>111dB
Input Lock Range	
AES3 / S/PDIF	32kHz to 192kHz
TOSLINK	32kHz to 96kHz
ADAT LIGHTPIPE™	30kHz 55kHz
s-Lock™ Sample Clock Intrinsic Jitter	<25ps
s-Lock™ Supported Sample Rates	44.1, 48, 88.2, 96, 176.4, 192kHz

POWER SUPPLY / GENERAL

Power consumption	40W
Input Voltage 50-60Hz	100, 120, 220, 230-240VAC
Dimensions	
Main chassis IEC 2U	H3.5" x W19" x D10.5"
Remote control unit	H1.6" x W8" x D5.25"
Power supply unit	H1.7" x W8.5" x D8.5"
Weight	
Main chassis	14.1lbs / 6.4kg
Remote control unit	2.3lbs / 1kg
Power supply unit	6.5lbs / 2.9kg

"I give the m906 the best compliment I can possibly give to a piece of pro audio gear – I simply refuse to mix without it."
– **Kevin McNoldy, Owner, Crystalphonic Recording Studios**

m904 stereo monitor control system

high fidelity stereo monitor control for all types of recording facilities

Stereo is king. So while 5.1 surround becomes ever more established in film and HD production, stereo audio still remains the standard for producing and delivering music. Accordingly, we also offer our monitor controller in a stereo only version - the m904.

The m904 combines all the same features and technology found in our flagship m906 surround controller, and is a perfect nerve center for any type of audio production facility - from project studio to world class mastering facility.

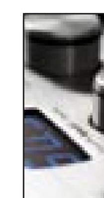
The m904 offers a full complement of balanced and unbalanced stereo analog and digital inputs, the latter featuring our latest generation reference quality 24bit/192kHz DAC technology with our proprietary s-Lock™ PLL for rock solid digital stability. All inputs and outputs are easily user calibrated for seamless integration into any playback environment with all types of playback equipment.

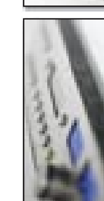
The m904 is configured as a 2U rack chassis, with all system controls built into the front panel, or is available as the m904b, built as a blank 2U mainframe which is controlled by a separate remote control unit (m904RCU sold separately). Additionally, a standard m904 can be used with the m904RCU, so the user can select between local and remote control.

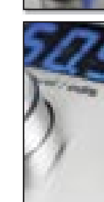
Included as a standard feature is our critically acclaimed reference headphone amplifier circuitry (as featured in our m902), with two output jacks on the mainframe or one jack on the m904b and one on the m904RCU. This circuitry provides absolutely pristine audiophile headphone amplification for critical monitoring, mixing and quality control.

While the m904 provides an unparalleled list of useful features, the real achievement for us was to implement all of this while maintaining our trademark musical, high fidelity audio performance. All components are selected and design decisions made for absolute audio transparency, making this the perfect link between your mix and monitors.

The m904 is here to empower audio professionals with the highest performance, best reliability, and most comprehensive feature set that any stereo monitor control system has to offer.







features

- multiple stereo analog inputs
- 24bit/192kHz digital stereo inputs- AES3, S/PDIF, ADAT and TOSLINK
- s-Lock™ phase lock loop sample clock regeneration for ultra-low jitter and rock solid digital stability
- precision speaker and headphone level controls with a 100dB range in 0.5dB steps
- every electrical and mechanical design element maximized for absolute purist, audiophile performance
- available as the m904b, which has a blank panel mainframe unit and is controlled via m904RCU (sold separately)
- all I/O and audio are routed in the 2U rack mount mainframe powered by an internal AC supply
- multiple speaker set selection - up to 3 stereo sets
- m902 style high-current audiophile headphone amplifier built in with two outputs on front panel
- complete system level calibration (inputs, outputs, inter-channel balance, dim)
- balanced talkback microphone input and activation switch with additional external foot switch control jack
- fixed level stereo DAC output for direct audo transfers
- five year warranty on parts and labor

m904 remote control unit

specifications

ANALOG IN

THD+N (@ 0dB gain, 1kHz)	
@+20dBu out	<0.001% (0.00075% typical)
@ 0dBu out	<0.003% (0.001% typical)
Frequency Response	
+/-3dB	3Hz - 250kHz
Dynamic Range	
20-22kHz bandwidth	112dB
Output Noise (20-22kHz bandwidth, -20dB gain)	
High gain mode	-86dBu
Low gain mode	-96dBu
Output Noise (A weighted, -20dB gain)	
High gain mode	-91dBu
Low gain mode	-99dBu
Phase Deviation	
20Hz to 20kHz	< 4°
Crosstalk any channel	
1kHz	< 100dB
Balanced Input CMRR	
60Hz	> 60dB
1kHz	>65dB
10kHz	>60dB
Gain Range	
Low gain mode (0.5dB steps)	-105.5dB to +20dB
High gain mode (0.5dB steps)	-95.5dB to +30dB
Channel Tracking Accuracy	
Any channel	+/-0.05dB
Maximum Output Level	
Low gain mode (balanced)	+15dBu
High gain mode (balanced)	+27dBu
Maximum Input Level	
Balanced inputs	+30dBu
Unbalanced inputs	+20dBu
Impedances	
Balanced inputs	30k Ohms
Unbalanced inputs	50k Ohms
Balanced outputs	300 Ohms
Minimum load impedance	300 Ohms

D/A CONVERTER FIXED OUTPUTS

THD+N	
1kHz, -1dBFS, 20-22kHz bandwidth	0.002%
Dynamic Range	
20-22kHz bandwidth	>111dB
Input Lock Range	
AES3 / S/PDIF	32kHz to 192kHz
TOSLINK	32kHz to 96kHz
ADAT LIGHTPIPE™	30kHz 55kHz
s-Lock™ Sample Clock Intrinsic Jitter	<25ps
s-Lock™ Supported Sample Rates	44.1, 48, 88.2, 96, 176.4, 192kHz

Power consumption	40W
Input Voltage 50-60Hz	100, 120, 220, 230-240VAC
Dimensions	
Main chassis IEC 2U	H3.5" x W19" x D10.5"
Remote control unit	H1.6" x W8" x D5.25"
Weight	
Main chassis	16.7lbs (7.6kg)
Remote control unit	2.3lbs (1kg)

"I am blown away by its well thought-out design and more than anything else, the full-featured remote. Finally, one huge benefit of the m904 is that it sounds f***ing great!" - **John Baccigaluppi, The Hangar Studios / Tape Op**

m902 reference headphone amplifier

headphone amplifier, DAC and monitor control for professional and audiophile applications

This is the audio microscope. Simply plug in your favorite headphones, cue up your go-to reference disc and get ready to hear, literally, what you've been missing. Presenting the m902 reference headphone amplifier.

At the core of the m902 is our audiophile, high-current transimpedance amplifier circuitry, which has been designed to effortlessly drive even the lowest impedance headphones and provide the high-resolution and low-level ambient detail for which all Grace Design products are famous. This circuitry makes familiar recorded material become at once transparent and alive, and critical details in editing and mastering become obvious and easy to manipulate.

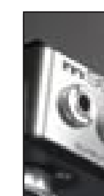
Professionals and audiophiles alike are continually floored by the m902's latest generation, high performance 24bit/192kHz DAC, which includes our unique s-Lock™ dual stage PLL (Phase Lock Loop) for extremely low intrinsic jitter and rock solid digital performance.

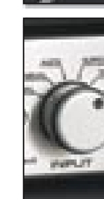
A full compliment of analog and digital inputs are provided (AES3, S/PDIF, TOSLINK and USB), which are all selectable via a front panel rotary switch. We offer an optional infrared remote control, which provides the essential controls for anyone working out of reach of their m902.

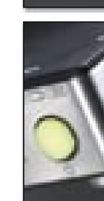
The m902 cleverly adds stereo unbalanced analog line outputs for connection to studio monitors or any speaker system, allowing it to double as an elegantly simple high fidelity monitor controller or even a purist audiophile preamplifier / DAC. And, similar to its big brothers (m906, m904), the m902 provides some essential user calibration settings which helps it to integrate into any playback environment.

The m902 includes a crossfeed circuit, or XFeed, which simulates the natural acoustics of a loudspeaker listening environment and HRTF (Head Related Transfer Functions). This circuitry, designed by Dr. Jan Meier, can significantly improve imaging, while reducing listening fatigue when using headphones.

Celebrated by audiophiles and audio engineers alike, the m902 is our crossover smash hit and is a must have for anyone searching to discover new depths in music and sound.







features

- balanced and unbalanced analog inputs
- 24bit/192kHz digital stereo inputs- AES3, S/PDIF, TOSLINK and USB (USB input supports 16 bit/44.1 and 48k only)
- s-Lock™ phase lock loop sample clock regeneration for ultra-low jitter and rock solid digital stability
- high-current transimpedance headphone amplifier circuitry built to effortlessly drive low impedance headphones
- variable level unbalanced analog outputs for studio monitors
- all electrical and mechanical design elements maximized for absolute purist, audiophile performance
- precision level control with a 95dB range in .5 dB steps
- channel level matching accuracy of 0.05dB
- crossfeed circuitry (XFeed) for improved headphone imaging
- optional infrared remote control available
- internal linear power supply with custom wound toroidal transformer
- no electrolytic capacitors in the signal path
- highest quality metal film resistors used throughout
- sealed gold contact relays used for all signal switching
- five year warranty on parts and labor

optional infrared remote

specifications

ANALOG SPECIFICATIONS

Gain	
Normal Mode	+0dB
Boost Mode	+10dB
Frequency Response	
@ 0dBu out +/- .25dB	22Hz - 120kHz
@ 0dBu out +/- .5dB	12Hz - 260kHz
@ 0dBu out +/-3dB	4Hz - 600kHz
Maximum Output Level	
@1kHz, 50 Ohm load	+21.4dBu (9.11Vrms)
Impedance	
INPUT	106K Ohms balanced
INPUT	53K Ohms unbalanced
OUTPUT	1 Ohm
Dynamic Range	
@ 0dB gain	116dB
@ -10dB gain	119dB
THD+N	
+10dBu out, 50 Ohm load, SMPTE 4:1	<0.01%
Headphone +10dBu out, 50 Ohm load	<0.008%
Line Out +10dBu out	<0.002%
Attenuator Channel Matching	
24 position switch	<0.05dB
Attenuator Range	99.5dB

D/A CONVERTER

Input Sample Rate	32, 44.1, 48, 88.2, 96, 176.4, 192kHz
THD+N	
44.1kHz, 24bit, 1kHz, +20dBu out	<0.002%
Noise Floor	
0dB gain, 22-22kHz	-94dBu
0dB gain, A-weighted	-97dBu
Power Requirements	
115VAC	.16A
230VAC	.08A
Dimentions and Weight	
Dimentions	H1.7" x W8.5" x D8.25"
Weight	5 lbs (2.2kg)

"The m902 combines comprehensive functionality, intuitive ease of use and luscious sound, all in one compact and elegant package."
- **John Marks, Owner/Producer, John Marks Records**

Grace Design has been building high-performance gear for the recording industry for over fifteen years. All of our designs are the result of extensive field testing, critical listening, and thoughtful revision. The following is a description of our unique approach to electronic design, our implementation of specific design concepts, and the reasoning behind their use. While the true tests of any audio product are sonic performance and reliability, we believe that understanding the designer's intentions and beliefs is helpful when assessing the inherent value of any high end audio product.

Our 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. Furthermore, transimpedance amplifiers reveal greater detail and provide a more open, musical sound quality. Also, 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 long capacitive 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. We feel that the advantages of this over a traditional "discrete" layout approach are important to understand. With the amplifiers 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.

In the early days of integrated circuits, it was better to design a discrete circuit for audio applications because it was impossible to create transistors and resistors of the same quality as discrete devices. Today, however, 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.

A note 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. Changing the operation mode to class A eliminated the crossover distortion, but at the expense of dramatically higher power consumption and lower available power for the speaker. 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 microphone preamplifiers or any other low level electronics, 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 mode.

Controls

Each gain control used in all our preamplifiers 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 dynamic distortion. Most conductive plastic potentiometers have temperature coefficients of +/-1000ppm/°C, which can cause considerable signal degradation at high amplifier gains. By comparison, a metal film resistor has a typical temperature coefficient of +/-50ppm/°C, 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 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 model 101) incorporate robust linear power supplies. The IEC 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 complete long term reliability.

Capacitors have perhaps the most significant impact on signal purity. The only capacitors used in our signal paths 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.

Electrolytic capacitors 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 a 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 than electrolytics, 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 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.

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 input stage and the best performing A/D converter are only as good as their 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 an 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 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 problems to a level well below the noise floor of the A/D converter.

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, which is 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.

"We have been using Grace mic preamps for a long time and if you told me that the 802 could have eclipsed the incredible sonic performance of our 801, I would have thought you were crazy. When we got the new 802 in and auditioned them we were completely knocked out. Better in every way. The Grace Design 802 sets new standards for performance in microphone preamps.If you are looking for a mic pre that doesn't get in the way of what your mics are capable of delivering, the 802 is it."- **Mike Pappas Chief Engineer KUVO Jazz**

m802 remote controlled microphone preamplifier

remote controlled transformerless, transparent microphone preamplification for all critical recording applications

The m802 isn't just in a class of its own, it's an entirely different species. Arguably the world's most evolved mic preamplifier, the m802 delivers functionality and performance which simply doesn't exist in any other mic amp design.

The core is a high fidelity 8 channel microphone preamplifier designed to be extremely transparent and musical. Then the fun begins. The m802 is fully remote controllable. With our full featured stand alone hardware RCU, up to 64 channels (a system of 8 m802's) can be addressed and controlled seamlessly from up to 1000' away. All preamplifier functions, including gain, phase, 48V phantom are instantly accessible and adjustable. These settings can then all be stored into preset locations and instantly recalled.

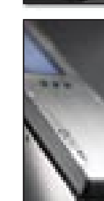
Or, the m802 can also be controlled directly from Digidesign® ProTools|HD® systems, compatible control surfaces and many other MIDI devices. Working closely with Digidesign®, we have adapted their control spec to directly address the m802 as an integrated part of a ProTools|HD® system. Additionally, the m802 can be controlled by a wide variety of MIDI mappable devices, making it easily integrated into a wide variety of remote recording situations.

Next, consider the optional 8 channel 24bit/192kHz A/D card option. Developed as the perfect complement to the m802 mic preamplifier design, this converter card provides pristine, musical A/D conversion to rival even the finest stand alone units available. Included is our proprietary unique s-Lock™ dual stage PLL for extremely low intrinsic jitter and rock solid digital performance.

DPA® 4000 series 130V microphones are widely regarded as some of the finest reference-quality microphones available. Yet, with their unique 130V power requirement, preamplifier options for these mics are quite limited. So we have made the m802 an excellent complement for DPA® microphones by adding a 130V power option, available on a per channel pair basis. The m802 now includes ribbon mic mode, which shifts the preamplifier's entire gain range up 10dB, while deactivating 48V phantom power, optimizing input impedance and bypassing the decoupling capacitors.

While the m802 works beautifully as a stand alone mic preamplifier, the optional Remote Control Unit (RCU) provides fantastic ergonomics and enhanced system control. This updated RCU now offers dedicated individual channel select switches and a new highly visible blue LED numeric display for channel gain settings. Both the RCU and the m802 now come with a high contrast, neutral white backlight LCD.

Whatever your application, we invite you to discover the remarkable sonic performance, functionality and reliability of the new m802. We're confident that it will help you make the finest recordings of your career.



features

- fast, musical transimpedance amplifier architecture
- fully balanced, transformeless design
- flexible digital metering with peak hold and reset
- controllable from Digidesign® ProTools|HD® systems and compatible control surfaces (and many other MIDI devices)
- 130V phantom power available, per channel pair, for DPA microphones
- dual high-current outputs (XLR+D-Sub) drive long cable runs and loads down to 50 Ohms
- fast, efficient Philips I^2C serial protocol to control preamplifiers from up to 1000 feet away
- gain range of -7dB to +63dB in 1.5 dB steps
- new selectable ribbon mic mode
- channel group mode (perfect for stereo pairs, subgroups)
- RCU able to address up to 8 units (64 channels) from up to 1000 feet away
- front panel lockout mode
- large white backlit LCD display
- 15 User preset memory locations, for saving preamplifier settings (gain, phantom power, phase)
- optional reference quality 24bit/192kHz A/D converter card
- A/D outputs include 2x8ch AES3 on DB25, 1x8ch AES3-id on 4 BNC connectors (supports double wire mode)
- optional ADAT Lightpipe output with SMUX for 96kHz
- s-Lock™ dual stage PLL for extremely low jitter without sample rate conversion
- wordclock in and out
- five year warranty on parts and labor

optional A/D converter module

specifications

Frequency Response	
@ 40dB gain ± 0.2dB (50Ω source)	15Hz-300kHz
@ 40dB gain ± 3dB (50Ω source)	4.5Hz-1.0MHz
THD+N	
@ 19.6dB gain +20dBu out, 1kHz	<.0007%
@ 40.5dB gain +20dBu out, 1kHz	<.0010%
@ 60.0dB gain +20dBu out, 1kHz	<.0050%
Intermodulation Distortion	
@ 40dB gain +20dBu out	
SMPTE/DIN 1:1 (50Hz, 7kHz)	<.0015%
SMPTE/DIN 4:1 (50Hz, 7kHz)	<.0040%
Noise - Referred to Input	
@ 60dB gain, 50Ω source	-129dB
@ 60dB gain, 150Ω source	-127dB
@ 60dB gain, 600Ω source	-123dB
Phase Deviation	
100-20KHz @ 40.5dB gain	<3°
Crosstalk	
Any Channel @ 40.5dB gain 1kHz	-110dB
CMRR	
@ 60dB gain, 3.5Vcm, 1kHz	>70dB
@ 60dB gain, 3.5Vcm, 10kHz	>70dB
Maximum Output Level	
Balanced	+27dBu
Unbalanced	+21dBu
Impedance	
Input	4.35kΩ
Output	200Ω
Minimum load impedance	50Ω
Weight and Dimensions	
m802 preamplifier	12lbs (5.4kg), 2U rack mount x D10”
Remote control unit	2.3lbs (1kg), H1.6” x W8” x D5.25”
power supply	4lbs (1.8kg), H1.7” x W 8.5” x D8.5”
Power Consumption	
100/120/230/240V~ 50-60Hz	60 Watts

A/D CONVERTER MODULE (without preamplifier)

Dynamic range		
20Hz-20kHz		>115dB
“A” weighting		>117dB
THD+N		
1kHz, -1dBFS, 20Hz-22kHz		< 0.00026% (-112dB)
Frequency response	+/-0.2dB	-3dB
44.1kHz Fs	5Hz-21kHz	22kHz
48kHz Fs	5Hz-23kHz	24kHz
88.2kHz Fs	5Hz-41kHz	44kHz
96kHz Fs	5Hz-45kHz	48kHz
176.4kHz Fs	5Hz-54kHz	72.5kHz
192kHz Fs	5Hz-59kHz	79kHz
Full scale input level		+16dB or +24dB (+/-2dB trim)
CMRR		
60Hz		>65dB
1kHz		>80dB
10kHz		>60dB
IMD SMPTE 4:1 60Hz, 7kHz, -3dBFS		<100dB (0.0008%)
Interchannel crosstalk		<120dB
Group delay		
44.1-48kHz		13/Fs
Sample rates, internal crystal (kHz)		**44.1, 48, 88.2, 96, 176.4, 192**
External Clock Lock Range		
Wide lock mode		40.1kHz-207kHz (+/-8% at each sample rate)
s-Lock™ mode		+/-250ppm (+350, -400ppm typical)
Intrinsic Jitter, 200Hz-20kHz BW		
Wide lock mode		< 60ps RMS
s-Lock™ mode		< 40ps RMS
Jitter Rejection Corner Frequency		
Wide lock mode, -3dB, 12dB/octave		800Hz
s-Lock™ mode, -3dB, 12dB/octave		10Hz

“Virtually every recording I have presided over during the past dozen years has been made using Michael Grace’s preamps - and this includes the m802, practically from the day the unit was launched. I cannot imagine getting more musical nuance from any other product.” -**Tim Martyn, Phoenix Audio**

m801 eight channel microphone preamplifier

transformerless, transparent microphone preamplification for all critical recording applications

Introduced in 1995, the model 801 microphone preamplifier was slightly ahead of its time. But as digital audio workstations soon flooded the industry, an accompanying high performance 8 channel mic preamplifier suddenly made perfect sense. So the model 801 quickly became an industry standard, which remains inimitable and unsurpassed.

Eleven years later, with a nearly flawless service record and an endless supply of critical acclaim, the model 801 gets a much deserved update, including an updated audio path, a new chassis forged of stainless steel and a slightly new name: the m801.

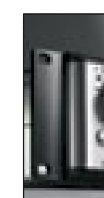
First and foremost, the new m801 delivers unmatched audio performance – with massive headroom and ultra-wide bandwidth which contribute to a markedly open, musical character. The m801 is designed to effortlessly resolve even the lowest level ambient information, resulting in a sonic picture of astonishing clarity and detail, which serves to capture the essential character of the music being recorded.

The m801's signal path has been hot-rodded to be fully balanced from start to finish, resulting in wider dynamic range, and new higher current output drivers enable even longer cable runs without signal loss.

Now included on each channel is a dedicated ribbon microphone switch, which shifts the preamplifier's entire gain range up 10dB, while deactivating 48V phantom power, optimizing input impedance and bypassing the decoupling capacitors. And, we've added an additional set of 8 channel balanced outputs for sending signals to a secondary recorder, workstation or console.

Each channel offers 48V phantom, phase reverse and a 20dB pad. Gain controls are our finest 24 position gold contact rotary switches, wired with 1% metal film resistors for maximum audio performance and perfect resetability. All components, active and passive, are selected for pure audio performance and long term reliability. No corners are cut and no compromises are made.

While its predecessor now takes its place in pro audio history, the new m801 faithfully furthers the tradition of breathtaking audio performance and perfect reliability in a beautiful, functional 8 channel package.



features

- fully balanced, transformerless design
- all new, higher-current outputs drive even longer cable runs
- ribbon mic mode switch on each channel
- fast, musical transimpedance amplifier architecture
- precision 24 position gold contact rotary switch gain controls
- 48V phantom power, 20 dB attenuator and phase reverse
- no electrolytic capacitors in the signal path
- sealed gold contact relays for all signal switching
- all electrical and mechanical design elements maximized for absolute purist, audiophile performance
- large two color, bi-phase LED peak indicator
- regulated linear power supply with custom wound toroidal transformer
- highest quality metal film resistors used throughout
- five year warranty on parts and labor

specifications

Frequency Response	
@ 40dB gain ± 3dB 50Ω source	4.5Hz-350KHz
@ 40dB gain ± 0.2dB 50Ω source	18Hz-65KHz
THD+N	
@ 20dB gain +20dBu out, 1kHz	<.0008%
@ 40dB gain +20dBu out, 1kHz	<.0009%
@ 60dB gain +20dBu out, 1kHz	<.0070%
Intermodulation Distortion	
@40dB gain +20dBu out	
SMPTE/DIN 1:1 (50Hz, 7kHz)	<.0020%
SMPTE/DIN 4:1 (50Hz, 7kHz)	<.0030%
Noise - Referred To Input	
@60dB gain 50Ω source	-130dB
@60dB gain 150Ω source	-127dB
@60dB gain 600Ω source	-123dB
Phase Deviation	
100-20KHz @40dB gain	<3°
Crosstalk	
Any Channel @40db Gain 1khz	-140db
Any Channel @40db Gain 10khz	-130db
CMRR	
@60db Gain, 3.5vcm, 1khz	>70db
@60db Gain, 3.5vcm, 10khz	>70db
Phantom Power	
Voltage +48v	+0.9/-0.0
6.8kΩ resistor match tolerance	+/- 0.1%
Maximum Output Level	
1khz, 100kΩ Load	+28dbu
Impedance	
Input	8100Ω
Input, Ribbon Mode	20kΩ
Output	190Ω
Dimensions	
Weight	15lbs (6.8kg)
Height	2u
Width	19"
Depth	10"
Power Supply Specifications	
Power Consumption	
100-240vac 50/60hz	60 Watts Max
Power Supply Dimensions	
Weight	6.25lbs (2.8kg)
Height	1.7"
Width	8.5"
Depth	8.5"

m201 two channel microphone preamplifier

transformerless, transparent microphone and instrument preamplification for all critical recording applications

During its 12 years in production, the model 201 mic preamplifier has found its way into countless recording facilities around the world, while gathering a considerable amount of critical acclaim along the way. Now this classic two channel mic preamplifier has been completely redesigned and transformed into the m201.

The new m201 delivers unmatched audio performance, with massive headroom, ultra-wide bandwidth and a very open, musical character. The signal path has been hot-rodded to be fully balanced from start to finish, resulting in a wider dynamic range, while new high-current output drivers enable even longer cable runs without signal loss. The m201 translates a sonic picture of astonishing clarity and detail, which serves to capture the essential character of the music being recorded.

Our ribbon mic mode is now included, which shifts the gain range up 10dB while deactivating 48V phantom power, bypassing the decoupling capacitors and optimizing the input impedance. Each channel has an 'input mode' rotary switch which selects between a standard 48V phantom input, ribbon mic mode, a front panel DI input or optional DPA high voltage inputs (130V or 190V).

Included is our newly designed M+S (mid side) decoder circuitry with a built-in width control. The front panel width control is wired with a precision 12 position rotary switch which provides a range from 100% mid (mono) to 30% mid / 70% side. The ultra-precision summing and difference amplifiers feed a set of dedicated outputs, which allows simultaneous recording of discrete mid+side signals as well as the stereo matrix.

The new front panel mounted HI-Z inputs are designed to accommodate a wide variety of high impedance input sources, making the m201 an excellent choice as a DI box which will flawlessly preserve the sound of any plugged-in instrument. Imagine the same resolution and detail as our microphone preamplifiers available for DI recording situations.

Additionally, we will be offering the m201 with a state of the art 24-bit/192kHz A/D converter option (retrofitable), rounding the m201 off as the ultimate transparent input solution for any recording facility.

Whatever the application, the remarkable sonic performance and functionality of the new m201 will help you make the finest recordings of your career.

features

- redesigned, fully balanced, transformer-less design
- dual parallel XLR outputs for each channel
- ribbon mic mode
- front panel DI / instrument inputs
- DPA microphones 130V (190V) phantom power option
- built in M+S decoder with dedicated outputs and front panel width control
- optional reference quality 24bit/192kHz A/D converter section available in late 2007
- precision 24 position gold contact rotary switch gain controls
- new, higher current outputs drive even longer cable runs
- no electrolytic capacitors in the signal path
- sealed gold contact relays for all signal switching
- large two color, bi-phase LED peak indicator
- regulated linear power supply with custom wound toroidal transformer
- highest quality components used throughout
- five year warranty on parts and labor

specifications

Frequency Response	
@ 40dB gain ± 0.2dB (50Ω source)	18Hz-65kHz
@ 40dB gain ± 3dB (50Ω source)	4.5Hz-350kHz
THD+N	
@ 20dB gain +25dBu out, 1kHz	<.0008%
@ 40dB gain +25dBu out, 1kHz	<.0009%
@ 60dB gain +25dBu out, 1kHz	<.0070%
Intermodulation Distortion	
@ 40dB gain +25dBu out	
SMPTE/DIN 1:1 (50Hz, 7kHz)	<.0020%
SMPTE/DIN 4:1 (50Hz, 7kHz)	<.0030%
Noise - Referred to Input	
@ 60dB gain 50Ω source	-130dB
@ 60dB gain 150Ω source	-127dB
@ 60dB gain 600Ω source	-123dB
Phase Deviation	
100Hz-20kHz @40dB gain	<3°
Crosstalk	
Any Channel @ 40dB gain 1kHz	-125dB
Any Channel @ 40dB gain 10kHz	-115dB
CMRR	
@ 60dB gain, 3.5Vcm, 1kHz	>70dB
@ 60dB gain, 3.5Vcm, 10kHz	>70dB
Maximum Output Level	
1kHz, 100KΩ load	+28dBu
Impedance	
Input	8100Ω
Input, ribbon mode	20kΩ
Output	190Ω
Peak Led Meter	
Green threshold	-14dBu
Red threshold	+16dBu
Weight and Dimensions	
dimensions	2U, 19" rack mount x 10" deep
weight	16.5lbs (7.5kg)
Power Consumption	
100/120/230/240V~ 50-60Hz	20 Watts Max

"I take the 201 with me whenever I record out of town. It's been my pre-amp of choice for acoustic piano for several years, but I have to leave the 801 at Skywalker because so many other engineers request it." **- Leslie Ann Jones, Director of Music Recording and Scoring, Skywalker Sound**

model 101 single channel microphone preamplifier

high performance, affordable microphone and instrument preamplification for all recording and live sound applications

The model 101 is the golden child of our mic preamplifier line - a high performance, yet surprisingly affordable single channel serving of our critically acclaimed microphone preamplifier circuitry. It uses the same fast, musical transimpedance amplifier architecture as its bigger siblings, but is packaged and priced to be within the reach of any recording engineer, musician, or studio.

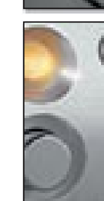
Maintaining Grace Design pedigree throughout, the model 101 employs our fully balanced, transformerless, transimpedance circuit topology and has no electrolytic capacitors in the signal path. With incredible bandwidth, headroom and resolution, the model 101 is remarkably natural, musical and detailed, allowing engineers working at any level or in any studio environment to record consistently amazing tracks.

Standard features include a precision 11 position rotary gain switch with a 10dB output trim control, 48V phantom power and a 75Hz high pass filter. Connectors include a standard XLR mic input, balanced XLR and 1/4" TRS outputs on the rear panel, and a high impedance 1/4" instrument input on the front panel. Also, the model 101 can be ordered in a ribbon microphone version, with an additional 10dB of gain - making it a perfect partner for all ribbon microphone models.

The included instrument DI input is designed to accommodate a wide variety of high impedance input sources, making the model 101 an excellent choice as a DI box which will flawlessly preserve the sound of any plugged-in instrument. Imagine the same resolution and detail as our microphone preamplifiers available for DI recording situations. Lackluster sounding instrument pickups suddenly become alive with clarity and detail, bringing out essential nuances which help musicians become more engaged with their instrument.

Furthermore, the model 101 is thoroughly rugged and road worthy, so discerning live performers can use it to achieve studio quality sound with their live performance setups. It is rack mountable in a standard 1U rack tray, and two units happily nest side by side in one rack space.

Equally at home amplifying a vintage condenser mic or a plugged-in acoustic guitar, the model 101 delivers the kind of high fidelity performance previously available only to big budget recording studios and Grammy winning musicians and engineers. This is hands down the ultimate single channel microphone preamplifier available for under $1,000.



features

- fully balanced, transformerless design
- fast, musical transimpedance amplifier architecture
- precision 11 position rotary switch gain
- 10dB output trim control for fine level adjustment controls
- optimized DI instrument input, XLR mic input
- built in high pass filter switch (75Hz,12dB per octave)
- 48V phantom power
- available as a high gain version for ribbon microphones
- Grace Design audiophile performance and rugged chassis construction, packaged to be available to any recordist
- two units can be mounted side by side in a standard 1U rack tray
- no electrolytic capacitors in the signal path
- sealed gold contact relays for all signal switching
- two color, bi-phase LED peak indicator
- highest quality metal film resistors used throughout
- five year warranty on parts and labor

specifications

Gain Range (5dB steps)	
Mic input	10-60dB
Hi-Z input	-10-40dB
Mic input high gain version	20-70dB
Hi-Z input high gain version	0-50dB
Output trim attenuator	0 to -10dB
THD+N	
@ 20dB Gain +20dBu out	<0.00085%
@ 40dB Gain +20dBu out	<0.0010%
@ 60dB Gain +20dBu out	<0.0050%
Intermodulation Distortion	
@ 40dB Gain +20dBu out	
SMPTE/DIN 4:1 7kHz/50Hz	<0.0020
Noise - Referred to Input	
50Ω source	<-130dB
150Ω source	<-128dB
600Ω source	<-124dB
CMRR	
100Hz	>68dB
1kHz	>75dB
10kHz	>65dB
Phase Deviation (HPF off)	
50Hz-25kHz	<6°
Frequency Response	
Mic input @ 40dB Gain -3dB	4.5Hz-390kHz
Mic input @ 40dB Gain -0.5dB	10.5Hz-140kHz
Hi-Z input @ 20dB Gain -3dB	2.5Hz-195kHz
Hi-Z input @ 20dB Gain -0.5dB	6Hz-74kHz
Impedance	
Mic input	3kΩ
Hi-Z input (unbalanced)	1MΩ
Hi-Z input (balanced)	2MΩ
Output	300Ω
Peak LED Meter	
Green threshold	-14dBu
Red threshold	+16dBu
Weight and Dimensions	
Weight	5lbs (2.3kg)
Dimensions	W8.5” x D6.25” x H1.7”

“This thing sounds beautiful!” - **Bill Frisell, Guitarist, 2005 Grammy Award Winner**

lunatec V3 portable microphone preamplifier

portab

Luckily, music isn't confined to the recording studio. So for those dedicated to capturing music wherever and whenever it happens, we continue to offer the lunatec V3 portable DC powered 2 channel mic preamplifier with A/D converter.

The Lunatec V3 delivers the same transparent, high fidelity audio performance as the rest of our mic preamplifier line, while being conveniently DC powered and portable enough to fit in a shoulder bag and accompany you and a recorder anywhere you need to go. And when you get home, plug the lunatec V3 in and enjoy its immaculate audio performance in your studio.

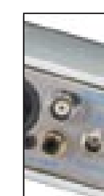
Our ultra-low distortion, 24/192kHz A/D converter section is included as standard, with flexible outputs including S/PDIF, TOSLINK and AES3. Both single and dual-wire 176.4 and 192kHz AES/EBU modes are provided via a pair of XLR connectors, which enable connection to today's high sample rate dual-wire digital recording systems, as well as including the provision for single-wire systems that will likely become the standard in the future. All outputs, analog and digital, are active at all times for even further capture flexibility.

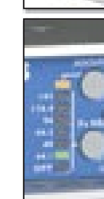
The V3 uses precision 11 position rotary gain switches, with 10dB output attenuators for fine level adjustment. Each channel has 48V phantom power and 2 position high pass filters (presetable at 50-100Hz or 75-150Hz with either 6 or 12 dB per octave slope). Plus, for engineers working with M+S mic technique, the lunatec V3 includes a M+S decoder matrix as standard.

The lunatec V3 is also equipped with our proprietary analog noise shaping dither circuit, called ANSR™, which can be applied for high quality word length reduction when sending signals to a 16-bit digital recorder.

With measured and sonic performance that rivals even the finest microphone studio preamplifiers, the lunatec V3 is designed to effortlessly translate the most subtle, complex low-level ambient information or highly dynamic material without noise or coloration. From ambient field recording, to taping large venue concerts, what your microphones hear is exactly what gets captured.

Sound designers, concert tapers, remote engineers, classical engineers or anyone with a portable recording system will simply be thrilled by the depth, clarity and realism the lunatec V3 delivers. Combined with all the essential features needed for true portability, the Lunatec V3 remains the best performing, portable, DC powered microphone preamplifier available.





features

- fully balanced, transformerless design
- fast, musical transimpedance amplifier architecture
- precision 11 position gold contact rotary switch gain controls
- 10dB output trim controls for fine level adjustment
- A/D converter with 44.1, 48, 88.2, 96, 176.4 and 192kHz sampling rates
- ANSR™ dither circuit for recording to 16-bit recorders
- 2 position high pass filters presetable at 50-100Hz or 75-150Hz with either 6 or 12 dB per octave slope
- 8-segment dot mode LED level meters
- 48V phantom power
- M+S decoding matrix
- converter section disable for battery power conservation
- Grace Design audiophile performance and rugged chassis construction, packaged to be portable and bombproof
- no electrolytic capacitors in the signal path
- highest quality metal film resistors used throughout
- five year warranty on parts and labor

specifications

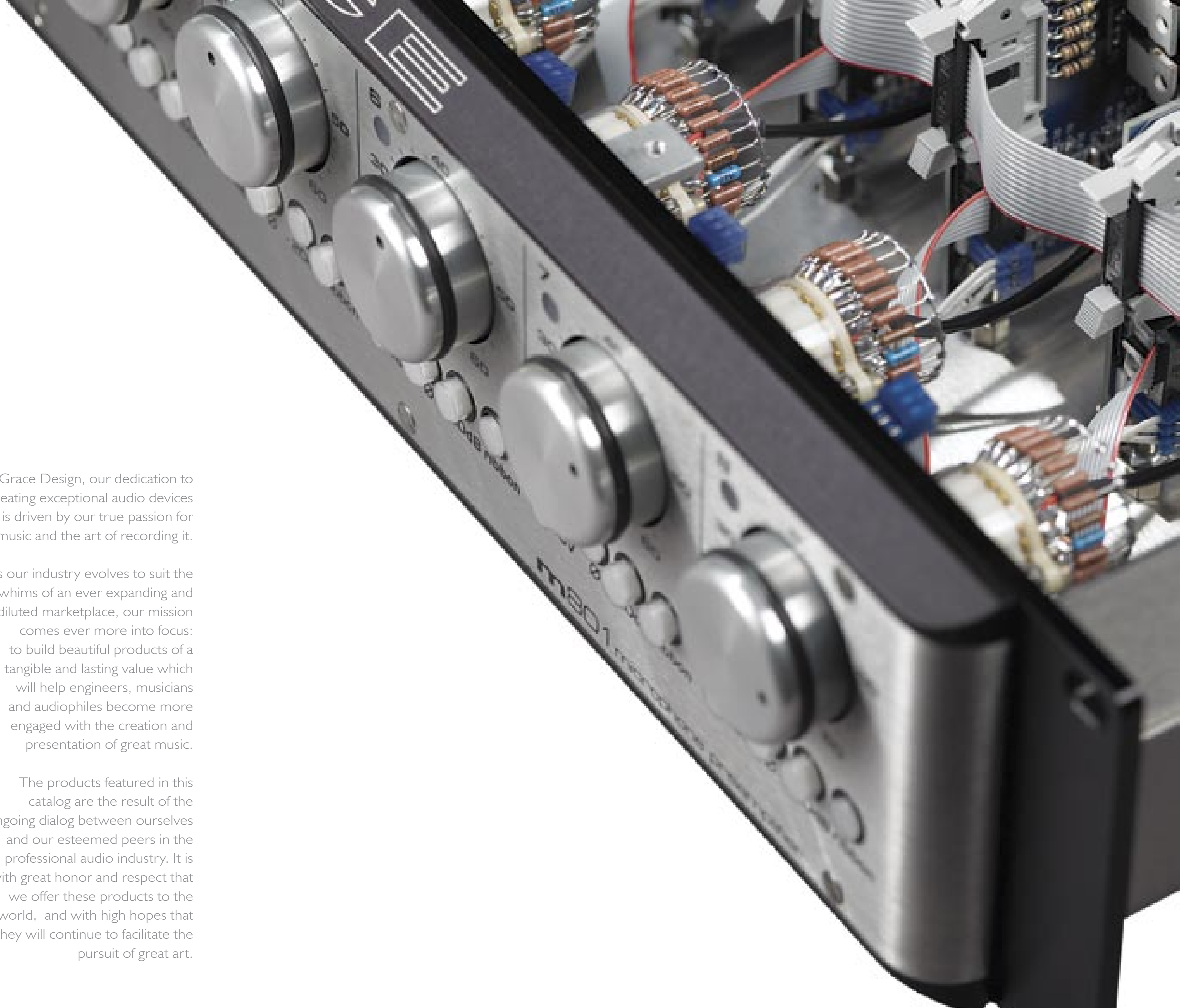
PREAMPLIFIER

Frequency Response	
@ 60dB gain -3dB	6Hz-250kHz
THD+N	
@ 20dB gain +20dBu out	0.0011%
@ 40dB gain +20dBu out	0.0011%
@ 60dB gain +20dBu out	0.0046%
Intermodulation Distortion	
@ 40dB gain +20dBu out	
SMPTE/DIN 4:1 50Hz,7k	<.0045%
Noise - Referred To Input	
@ 60dB gain 50 Ohm source	<-130dB
Phase Deviation	
50Hz-20kHz	<8°
Crosstalk	
Either Channel	-109dB
CMRR	
@ 60dB gain, 3.5Vcm, 1kHz	70dB
@ 60dB gain, 3.5Vcm, 10kHz	80dB
Output CMRR	60dB
Maximum Output Level	
Balanced	+25dBu
Unbalanced	+19dBu
Impedance	
Input	3k Ohms
Output	300 Ohms
Minimum Load Impedance	600 Ohms

A/D CONVERTER

Dynamic Range	
44.1kHz-192kHz, 22Hz-20kHz bandwidth	107dB
44.1kHz-192kHz, A-weighted	110dB
THD+N	
44.1kHz-192kHz, -3dBFS, 22Hz-20kHz bandwidth	-96dB
Frequency Response	
Fs 44.1kHz +0.1/-0.4dB	20Hz-21kHz
Fs 48kHZ +0.1/-0.4dB	20Hz-22kHz
Fs 88.2kHz +0.1/-0.4dB	20Hz-42kHz
Fs 96kHz +0.1/-0.4dB	20Hz-45kHz
Fs 176.4kHz +0.1/-0.4dB	20Hz-75kHz
Fs 192kHz +0.1/-0.4dB	20Hz-82kHz
Power Consumption	
6-12VDC	600-1000mA
	(depending on A/D settings and microphone type)
Weight and Dimensions	
2.6 lbs (1.2kg)	W8.25” x D5.5” x H1.7”

“When I’m in the field, I never have to think twice about my V3. It frees me to focus on the sound.” - **Rudy Trubitt**



At Grace Design, our dedication to creating exceptional audio devices is driven by our true passion for music and the art of recording it.

As our industry evolves to suit the whims of an ever expanding and diluted marketplace, our mission comes ever more into focus: to build beautiful products of a tangible and lasting value which will help engineers, musicians and audiophiles become more engaged with the creation and presentation of great music.

The products featured in this catalog are the result of the ongoing dialog between ourselves and our esteemed peers in the professional audio industry. It is with great honor and respect that we offer these products to the world, and with high hopes that they will continue to facilitate the pursuit of great art.

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