Lexicon PRO
FULL LINE CATALOG


DIGITAL EFFECTS SYSTEM
lexicon
PCM 80
ΩMEGA
Mic 1
Peak
USB Assign
1-2
3-4
Stereo
Mono
Mic 2
Line 1
Line 2
Line 3
Line 4
Output Level
Monitor Mix
Direct
Play
Instrument
S/PDIF Assign
Lock
+48V
USB
lexicon
Clock
MIDI
A/B
Delay
Reverb

Lexicon PRO HISTORY

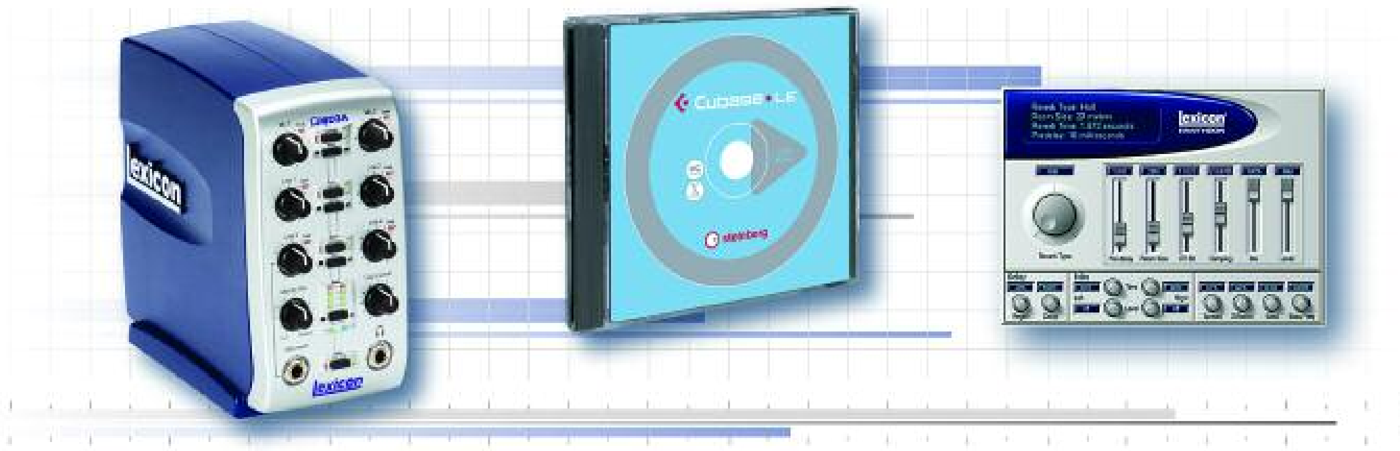
Lexicon® occupies a unique position as a leading innovator in both professional and consumer audio industries. Since the release of the first digital reverb in 1971, Lexicon has stood at the forefront of digital audio with a reputation as a manufacturer of exceptional professional audio and home theater products and an inventor of new technologies. Years of research, development, and experience allow us to continue expanding the boundaries of the listening experience.

Our professional products are prominent in the creation of worldwide music, television and film productions. These products have won numerous awards, including an Emmy® and numerous TEC awards, most recently a TEC Hall of Fame award for the Lexicon Delta T-101, the world's first digital delay. Lexicon processors have been embraced as the standard in professional signal processing since the introduction of the 480L Digital Effects Processor, which has retained tremendous popularity for the past 17 years. It has since been replaced as the standard in professional signal processing by the 960L Multi-channel Digital Effects System, which has itself garnered an impressive following of producers, artists, and engineers.

Growing demand for proprietary Lexicon technologies has led to their appearance in numerous applications – with dramatic results. Our processing is relied upon to enhance the sound of prestigious live halls and venues. Our critically acclaimed LOGIC7® technologies have been successfully incorporated in several world-renowned automobiles, including select models from BMW® and Mercedes®. LOGIC7 technologies have also been licensed to other audio companies such as harman/kardon® and AKG®.

Knowingly or unknowingly – you experience Lexicon products and technologies on a daily basis. Chances are that Lexicon processing was involved in the television program you watch at home, the film you see at the cinema, or the song you listen to on the radio. From the initial tracks to your listening room or automobile, Lexicon is part of the process that brings these recordings to life. Our commitment to the audio professional and content delivery ensures an unbroken chain between the artist and the audience

COMPLETE RECORDING SOLUTION



HISTORY IN THE MAKING

More than 30 years ago, when Lexicon invented digital reverb, most recording studios were multi-million dollar rooms occupied only by famous rock stars and big record labels. And while Lexicon still lives in that rarified air of stardom with the world's most sought-after reverbs, now you can find us a lot closer to home.

THE COMPLETE SYSTEM

Now Lexicon brings you the Omega Desktop Recording Studio, a completely integrated recording system with everything you need to transform your computer into a full-on professional 24-bit digital studio, all in one box. In addition to Steinberg Cubase LE® recording and production software, you'll get an 8-input, 4-Bus, 2-output USB I/O mixer with inserts, instrument input, MIDI I/O and complete metering and monitoring functions.Record up to 4 tracks at once from 8 audio sources, and mix up to 48 audio and 64 MIDI tracks almost anywhere…with the lush, exquisitely rich reverbs that made Lexicon famous in a VST version of Lexicon's Pantheon® reverb plug-in.

A REAL MIXER

Unlike standard computer I/O boxes based on a patch bay concept, the Omega Desktop Studio is designed and built around the same paradigm as large-format recording consoles. An 8-input, 4-Bus, 2-output USB I/O mixer with inserts, instrument input, MIDI I/O and complete metering and monitoring functions gives you the freedom to record up to 4 tracks at once and mix without the need for additional mixing hardware.

Two dbx® Silver Series™ mic preamps with 48V phantom power provide a pristine front end for high-end condenser or more common dynamic microphones, insuring premium sound quality from the first take to the last.

TRS insert points allow you to plug in your favorite dynamic processors before the signal reaches one of the 4 busses, where 24-bit converters transform the analog signal into a digital masterpiece. Four servo-balanced TRS line inputs with up to +22dBu input levels let you hook up everything from keyboards to guitar preamps, including high-output professional gear. Stereo digital inputs are available via SPDIF jacks, and an additional high-impedance instrument input is included on the front panel for easy access to guitars and basses.

Finally, a fully opto-isolated MIDI input to ensures that there is no annoying ground loop hum or MIDI talk-through noise common with many I/O boxes and sound cards; MIDI output has rock-solid synchronization to USB frame rate to support applications requiring critical sync. The Omega I/O mixer also features channel peak indicators for each analog input as well as an assignable bargraph meter for the 4 busses. By monitoring the signal before the A/D converters, clipping and distortion can be averted, unlike software-only level monitoring that can miss "overs", ruining the track.

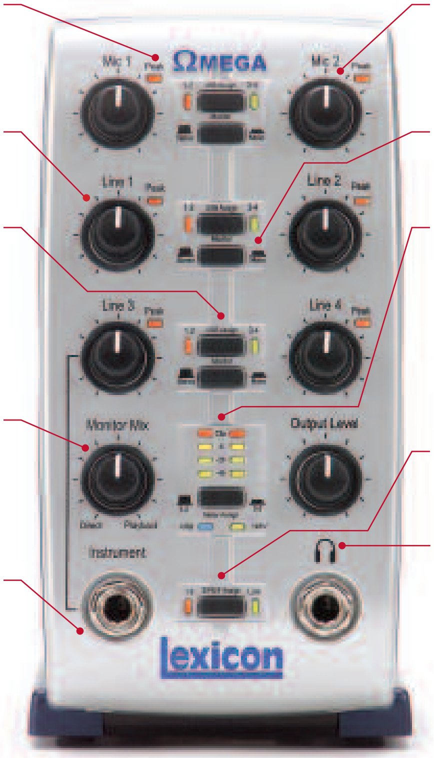
All these features are put together like a big recording console, with input gain controls, four recording busses, and full monitoring features, in one portable I/O Mixer. Just plug it into your Mac® or PC computer with the included USB cable, and you're ready to capture the moment.

ΩMEGA
DESKTOP RECORDING STUDIO

EVERYTHING YOU NEED TO TRANSFORM YOUR COMPUTER INTO A PROFESSIONAL 24-BIT RECORDING STUDIO.

COMPLETE RECORDING SOLUTION

THE OMEGA™ 8x4x2 USB I/O Mixer



PEAK INDICATORS: Make sure nasty, distortion-causing peaks don't sneak into your recordings.

LINE LEVELS: Control the input gain of the four professional-quality balanced line inputs.

BUS ASSIGN: Assign the mic and the line inputs in pairs to the 4 USB busses where ultra-transparent, very low noise 24-bit A/D converters turn your analog signal into pristine digital audio.

MONITOR MIX: An easy way to balance between live input and playback mix levels during recording and mixing. Zero-latency monitoring is a knob turn away.

INSTRUMENT IN: The ultra-Hi-Z input, designed for magnetic and piezo pickups, won't load down your instruments and rob them of their high end. Conveniently located on the front panel for easy access.

MIC LEVELS: Control the input gain of the professional-grade dbx® Silver Series™ mic preamps. Up to 50dB of gain!

STEREO/MONO: Selects whether you monitor the inputs in stereo or mono.

METERING: Omega features a 2-channel, 4-segment assignable bargraph meter to monitor exact levels at the A/D converters. Other products that rely on software metering can miss clipping at the converters, allowing distortion to sneak through without warning.

S/PDIF ASSIGN: Assign the S/PDIF input to USB busses 1-2.

TRU-REFERENCE™ HEADPHONE AMP: Omega Studio's Tru-Reference™ discrete design headphone amplifier offers ultra clear monitoring while delivering ample power needed for all types of headphones.

LINE OUTPUTS: Two RF-filtered, TRS servo-balanced outputs are smart enough to know whether they are connected to balanced or unbalanced equipment.

MIDI IN/OUT: Fully opto-isolated MIDI input eliminates ground loop hum and MIDI talk-through noise. MIDI output has rock-solid sync to USB frame rates.

S/PDIF: Direct digital transfers to or from your digital gear avoids unnecessary and sonically degrading A/D and D/A conversions. An additional DAC allows zero-latency monitoring of the S/PDIF source.

USB: Connect to your computer with the included USB cable. No PCI cards to install, no IRQ conflicts to resolve–just Plug and Play.

POWER: Omega Studio's external power supply is superior to using the USB power provided by the computer, which is often noisy and insufficient for professional applications.

LINE INPUTS: Four RF-filtered TRS active-balanced inputs accept either balanced or unbalanced signals. Maximum input levels of up to +22dBu allows interfacing to professional, high-output level equipment.

MIC PREAMPS: Two world-class dbx® Silver Series™ mic preamps present high-end studio condenser mics with an immaculate front end. Their extremely low noise design makes them stellar performers when using more common dynamic mics as well.

-20dB PAD: Reduce input gain by 20dB to accommodate high output microphones and XLR line-level devices.

TRS INSERTS: Easy access to your favorite outboard processors with the TRS insert points.

+48V PHANTOM POWER: Provide clean +48V DC voltage to high-quality condenser mics requiring phantom power.

ΩMEGA™
DESKTOP RECORDING STUDIO

FEATURES

- Peak Indicators prevent distortion
- Zero-latency Monitor Mix Control knob
- Ultra Hi-Z input, designed for magnetic and piezo pickups
- USB Plug & Play connectivity. No PCI cards to install, no IRQ conflicts
- External Power Supply eliminates noisy and insufficient power from computer
- 2-channel, 4-segment assignable bar-graph meter monitor exact levels at the A/D converters
- Tru-Reference™ Headphone Amp for ultra clear monitoring
- dbx® Mic Preamps
- TRS Inserts for easy access to outboard processors
- Phantom Power provides clean +48V DC voltage to high-quality condenser mics requiring phantom power.

STEINBERG® CUBASE LE™

Steinberg® is a world leader in computer-based recording. From the first idea to complete compositions, Cubase™ is the production suite of choice for musicians and producers looking for a creative way to realize their projects. Steinberg's intuitive cross-platform Cubase LE features 48 audio and 64 MIDI tracks with full automation, 2 inserts and 4 aux sends per channel, up to 8 VST instruments, and supports VST System Link and ReWire 2.

Cubase LE™ communicates seamlessly with the Omega USB I/O Mixer through our custom-written ASIO drivers to achieve a completely integrated, easy-to-use recording solution that includes all of the modules that you need to track, edit and mix your masterpiece.

- **48 audio and 64 MIDI tracks**
- **Professional audio editing and processing features**
- **Full automation of volume, pan, mute and effects**
- **MIDI score editor with advanced quantization and logical presets**
- **Supports up to 8 VST instruments**
- **2 insert and 4 effect sends per channel**
- **VST System Link and ReWire 2 compatible**

COMPLETE RECORDING SOLUTION

LEXICON® PANTHEON™

From the company that brought you the world's first digital reverb comes the world's best VST reverb plug-in: Lexicon Pantheon. Featuring many of the same algorithms that can be found on legendary recorded music and movie soundtracks, Pantheon delivers mega-studio quality signal processing to your home, with 6 reverb types and 35 factory presets. Each reverb type has 16 editable parameters to let you create your own variations on the legendary "Lexicon Sound."

- **Gives recordings that legendary "Lexicon Sound"**
- **35 factory presets**
- **6 reverb types**
- **16 editable parameters per reverb type**
- **Mono and stereo operation**
- **Advanced yet easy-to-use interface**
- **Floating point DSP processing**
- **16 and 24-bit compatible**
- **Efficient CPU utilization**

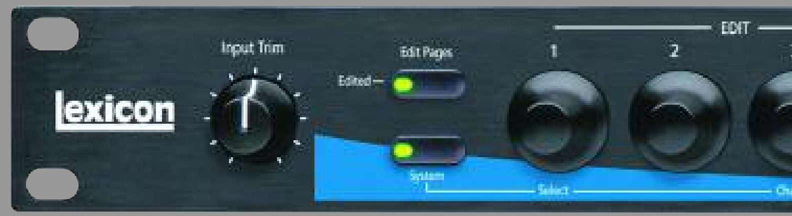
SPECIFICATIONS

Microphone Inputs:	(2) Female XLR Pin 2 Hot
Input Impedance:	600Ω balanced
Phantom Power:	+48 Volt DC
GAIN:	+50dB
EIN:	-120dB A-weighted @ 50dB gain (150Ω source impedance)
Maximum Input Level:	+18dBu
Frequency Response:	+0, -0.2dB 20 Hz - 20kHz, ref. 1kHz
THD+N:	<.005%, 20Hz - 20kHz
Insert Inputs:	(2) 1/4" TRS
Send Level (tip):	+19dBu maximum
Maximum Return Level (ring):	+19dBu maximum
Line Inputs:	(4) 1/4" TRS balanced or unbalanced
Input Impedance:	20kΩ balanced, 10kΩ unbalanced
Maximum Input Level:	+22dBu
Frequency Response:	+0, -0.2 dB 20 Hz - 20kHz, ref. 1kHz
THD+N:	<.009% A/D, 20Hz - 20kHz
Instrument Input:	(1) 1/4" mono jack
Input Impedance:	1 MΩ unbalanced
Maximum Input Level:	+19dBu
Frequency Response:	+0, -0.25 dB 20 Hz - 20kHz, ref. 1kHz
THD+N:	<.0125% A/D
Crosstalk:	<-74dB any input or output to any recording channel, 20Hz-20kHz <-95dB at 1kHz typical
Line Outputs:	(2) 1/4" TRS balanced or unbalanced
Level:	+19dBu maximum
Impedance:	110Ω
Headphone Output:	(1) 1/4" stereo jack 100 mW per channel at 50 Ohms
MIDI Interface:	5 pin DIN connectors for MIDI IN and MIDI OUT
Digital Audio Input:	Coaxial RCA (S/PDIF format)
Digital Audio Output:	Coaxial RCA (S/PDIF format) always transmits the audio data from the USB stream
D/A and A/D Conversion	
Sample Rate:	44.1 kHz or 48 kHz (determined by computer application)
Dynamic Range:	
A/D (24 Bit)	104dB typical, A-weighted, 20Hz - 20kHz
D/A (24 Bit)	109dB typical, A-weighted, 20Hz - 20kHz A/D/A (24 Bit) 103dB typical, A-weighted, 20Hz - 20kHz Analog Path: 118dB typical, A-weighted, 20Hz - 20kHz
USB Type B Socket:	Version 1.1, Version 1.1 hubs are not supported
Power Requirements:	PS0913-B adapter supplied. delivers 9V AC at 1300 milliamps, Class 2 transformer, draws 18W at 120 V
Dimensions:	4.625"W x 7.25"H x 7.75" D (118mm x 184mm x 197mm)
Weight:	2.65 lbs.

Specifications subject to change without notice.

MPX550

DUAL CHANNEL PROCESSOR



FEATURES

- Legendary Lexicon reverb from proprietary LexiChip
- 24-bit internal processing and A/D/A conversion
- Selectable 44.1 or 48kHz sampling rate
- 255 presets and 64 User programs
- Dynamics algorithm with enhanced metering
- S/PDIF I/O
- Balanced analog I/O (XLR and 1/4 inch TRS)
- Simultaneous analog and digital outputs
- Independent processing of each input
- Dual programs that combine two independent effects in four routing configurations plus compression
- Multiple delay, modulation, and pitch effects
- Built-in digital compressor with four adjustable parameters
- Tap Tempo
- Full MIDI control with software selectable MIDI Out/Thru connector
- Large, graphic front panel display
- Four Edit knobs for quick parameter adjustment
- Cue Program Mode
- Global Mix, Tempo, and Compressor Modes
- High impedance inputs for instruments
- Headphone output
- Internal power supply

The MPX550 offers high-end professional features and that famous "Lexicon Sound" at an affordable price. Like its predecessors, the MPX550 is a true stereo processor with dual 24-bit DSP engines and pristine 24-bit A/D and D/A conversion. The legendary LexiChip engine is at the core of its 255 carefully-crafted factory presets, with dual programs that combine two independent effects in four available routing configurations. Tempo for delay or modulated effects can easily be set from the front panel "Tap" button, using audio input, a dual footswitch, or an external MIDI device. A powerful editing tool called Learn Mode allows direct MIDI patching of all the MPX550 parameter controls. Additionally, the MPX550 adds impressive professional features like balanced analog inputs and outputs, a large front panel display, four Edit knobs for instant parameter adjustment, and an internal power supply.

EASY OPERATION

For all the power under the hood, the MPX550 is incredibly simple to use. All 255 presets and 64 User programs are instantly accessible via the Program knob, while a large graphic front panel display indicates program, editing and system status. Pressing the Load button sets the Program knob to select among Single, Dual and User programs. Otherwise, the Program knob activates the Cue Program Mode, which allows previewing of new programs while the currently selected program remains active. The Load button LED lights to indicate a new program is cued for loading: just press Load again during the four seconds the cued program is displayed to bring up the new program, or the display will revert to the currently loaded program.

Dedicated Edit knobs take the guesswork out of editing programs. Each program includes up to 20 parameters organized into Edit Pages, accessed through the Edit Pages button on the front panel, which cycles through the editable parameters available for the selected program. Each of the four parameters within the Edit Page are then accessible via the four corresponding Edit knobs - just grab and go! Each program also features an "Adjust" function - a special parameter selected for each preset that controls the most crucial aspect of that particular sound. Some presets have multiple parameters assigned to the "Adjust" function, allowing multiple related parameters to be controlled with a single knob.

LEGENDARY EFFECTS

In addition to the world-class effects in the MPX550: **Improved Reverb Algorithms** - With a richer, more spacious sound, and a smoother, more natural decay.

Compressor - Provides compression and limiting with ratios varying from 1:1 (nil) to 10:1 (limiting). The Adjust parameter can be used to level match input and output volumes. Reverb can be used in parallel with the compressor; or with the compressor acting as an input limiter before the reverb to tame runaway sibilance.

Dynamics - Intended for use in the studio when dynamic processing of stereo input signals is desired. Peak expansion, compression and tape saturation effects are provided, as well as metering options that provide precise visual indication of peak expansion, gain reduction, and input / output levels.

Live-FOH Mode - Designed for live performances, with controls that are more convenient for live sound engineers. All presets are Dual Mono to accommodate sound reinforcement systems used in most small to mid-sized venues. The first two Edit Pages contain the four most essential parameters for the first and second effects, respectively. Two different knob mappings accommodate multiple working styles.

TAP TEMPO

Tap Tempo greatly simplifies the process of matching delay times and modulation rates of tempo-based effects with the music - no more guessing what "might be" the tempo in milliseconds. When a tempo-based effect is loaded, the Tap button LED flashes at the currently set tempo. Just two presses of the Tap button in time with the music, and the delay or modulation rate is set. Tempo change in the middle of the song? Just two more presses and the effect is instantly set to the new rhythm - it really is that simple. Tempo can also be set using an audio input (a must for live performances), via a dual footswitch, or an external MIDI device that uses Continuous Controller or Program Change messages.

MIDI CONTROL

The MPX550 offers full MIDI control, with a powerful editing tool called Learn Mode that allows direct patching of all panel controls, plus individual parameter controls (up to 20 per program). Standard Continuous Controller, SYSEX commands or Program Change messages can be used to manipulate the parameters, as well as the Bypass and Tap buttons. In the studio, controls can be automated and recorded into a sequencer for complete preset automation.

SPECIFICATIONS

Audio Inputs (2)	Right sums to mono
Input Level:	-30dBu to +8dBu
Audio Outputs (2)	Right sums to mono, left for stereo headphones
Digital I/O	S/PDIF Inputs & Outputs at 44.1k or 48k
Frequency Response:	20Hz - 20kHz +/-1dB
Dynamic Range:	A/D: >105dB typical (20Hz - 20kHz unweighted) A/A: >101dB typical (20Hz - 20kHz unweighted)
THD+ Noise:	< 0.05% (20Hz - 20kHz)
Crosstalk:	> -97dBu
A/D Conversion:	24-bit A/D and D/A, 44.1k / 48k Sample Rate
DSP:	24-bit Processing
Dimensions:	19" W x 1.75" H x 5.5" D - Standard Rack Mount (483mm x 45mm x 140.25mm)
Weight:	3 lbs (1.4 kg)
Power Requirements:	90-250V 50-60HZ AC; 12.5W / 3-Pin IEC

Specifications subject to change without notice.

New edit icons simplify parameter adjustment.

Dynamics level meters show I/O levels, gain reduction, peak expansion.

MPX1

MULTIPLE PROCESSOR FX

FEATURES

- World-class Lexicon reverb: Plate, Gate, Hall, Chamber, and Ambience
- 2 DSP engines including Lexicon's proprietary LexiChip
- 24-bit A/D and 24-bit D/A conversion
- 200 presets; 50 User programs
- Up to six simultaneous effects
- Multiple delay, modulation, and pitch effects
- Effects include Reverb, Delay, Flange, Pitch, Chorus, Rotary, Parametric EQ, Sweep Filter, and Tremolo
- Large graphic display
- S/PDIF input and output
- Balanced analog inputs and outputs (XLR and 1/4 inch TRS)
- Independent processing on each input
- Tap Tempo for instant setting of delay and modulation times (may be set using footswitch)
- Full MIDI control
- Internal power supply

With two separate DSP processors, the MPX 1 Multiple Processor FX stands in a class all its own. The proprietary LexiChip delivers legendary Lexicon® reverb and ambience algorithms, and a second DSP processor delivers additional effect algorithms. The result is uncompromising stereo Reverb and Ambience effects that are available at all times, whether running alone or with Pitch, Chorus, EQ, Modulation, or Delay effects. While many "multi-effect" processors do not meet this basic requirement, the MPX 1 has enough processing power left over to run even more effects.

EASY OPERATION

Front panel display and indicators provide instant feedback about active and available features. The Tempo LED flashes to indicate the current tempo in programs that use tempo-controlled delay times and modulation rates, with a Tap button that allows quick tempo changes. The A/B LED lights to indicate that the A/B button is available to morph between effect or parameter variations. The numeric display makes program and patch numbers highly visible. The alphanumeric display shows parameter names and values, preset numbers, and algorithm routings. Dedicated Mix and Patch buttons provide instant access to mix and level settings for single or multiple effects and to the MPX 1 patch system. The Patch button lights whenever a patched parameter is displayed.

Six effect blocks are always accessible via dedicated, back-lit buttons on the front panel. Each block is independent, with its own bypass, mix, and level controls. Effect buttons light to indicate active effects. Selecting new effects for programs is simple, with an arsenal of effects for each block (56 total) just a button push away. Push-button access to a soft row that contains the most critical parameters for each program simplifies the process of fine-tuning presets to match music or soundtracks. Edit Mode provides an extensive pool of parameters for creating and restructuring presets.

ON-BOARD HELP

A built-in help feature provides guidance through all front panel controls. Whenever a button is pressed and held, a message appears explaining its function without executing its action.

PROGRAM LIBRARY

The MPX 1 offers an extensive program library that has been crafted to provide sounds suitable for performance, production, and sound design applications. These programs exploit the unique characteristics of each effect, and furnish useful examples of effect combinations that can be created with the MPX 1. A built-in database function makes it simple to locate the desired program. This database is user-definable, allowing reorganization as needed.

WORLD-CLASS EFFECTS

Lexicon's renowned reverb algorithms serve as the foundation for the MPX 1. Plate, Gate, Hall, Chamber, and Ambience effects are true stereo with independent processing of the left and right inputs.

The MPX 1 features 56 total Pitch, Chorus, EQ, Modulation, Delay, and Reverb effects. This includes high-precision (32-bit) Parametric EQ, rotary speaker cabinet simulator, high-quality 2-voice pitch shifter, delays and echoes (dual, mono, and stereo), Looper and Ducker, multi-voice stereo Autopanner, Chorus, Flangers, Phase, Wah and more.

Effect ordering and routing within each program is flexible. Effects can be arranged in any order by "dragging and dropping" them on a visual map. Similarly, a routing map allows distribution of the six effect blocks across two parallel stereo paths, which can be split and merged at multiple points. The routing map can be rewired by changing graphic "patch cords."

OPTIONS BUTTON

The MPX 1 provides as much control as possible, while keeping unused options hidden until needed. The Options button lights when additional features are available. For example, delay parameters can be displayed in feet, meters, milliseconds, or echoes-per-beat. Rate parameters can be displayed in Hertz(Hz) or cycles-per-beat. Feedback options allows effect blocks to be inserted in delay or echo feedback loops.

PATCHING SYSTEM

The ability to control dynamic effect parameters is essential to creating great-sounding programs. The MPX 1 patch system provides over 150 controllers that can be assigned to any parameter, including LFO (2), ADR (2), Envelope Follower (2), Arpeggiator, Random Generator, A/B Glide, Tempo, and a Sample and Hold Generator. Five patches are available per program. In addition, 10 global patches are available at all times.

MIDI CONTROL

Full MIDI control is available for individual and master Mix, Bypass, and Level controls; Tap and A/B controls; and all effect parameters. Tempo parameters can be synchronized to incoming MIDI clock, or the MPX 1 can transmit its own MIDI Clock based on Tap Tempo. Internal control sources such as audio levels, LFOs, ADRs, S/H, and pedals can be sent as Continuous Controllers. A built-in arpeggiator "plays" MIDI sound generators and produces synchronized audio effects.

SPECIFICATIONS

Analog Audio Input:	XLR and 1/4" balanced (T/R/S)
Input Level:	-2dBu to +20dBu, balanced; -14dBu to +8dBu, unbalanced
Input Impedance:	100kΩ, balanced; 50kΩ, unbalanced
Analog Audio Output:	XLR and 1/4" balanced (T/R/S)
Output Level:	+18dBu
Output Impedance:	600Ω unbalanced
Digital Audio Input:	Coaxial RCA
Format:	24-bit S/PDIF
Sample Rate:	44.1kHz
Digital Audio Output:	Coaxial RCA
Format:	24-bit S/PDIF
Sample Rate:	44.1kHz
Conversion:	24-bit A/D; 24-bit D/A
Internal Audio DSP:	20-bit; 32-bit
Frequency Response:	20Hz - 20kHz ±1dB
Crosstalk:	-60dB
THD:	0.01%, 20Hz - 20kHz
Dynamic Range:	D/A: >95dB typical, 20Hz - 20kHz, unweighted A/D: >90dB typical, 20Hz - 20kHz, unweighted A/A: >95dB typical, 20Hz - 20kHz, unweighted
MIDI Interface:	7-pin DIN connector for MIDI IN and powered bidirectional remote; 5-pin DIN connectors for MIDI THRU and OUT
Footswitch:	1/4" T/R/S connector for bypass and tap
Footpedal:	1/4" T/R/S connector (10kΩ - 100kΩ impedance)
Power Requirements:	100 - 240 volts AC, 50 - 60Hz, 25 watts (3-pin IEC connector)
Remote Power In:	2.5 mm 9 volts AC (not included)
Dimensions:	19" W x 1.75" H (1U) x 9" D (483 x 45 x 289 mm), rack mount standard
Weight:	6.125 lbs. (2.8 kg)
Operating Temperature:	32° to 104°F (0° to 40°C)
Maximum Humidity:	95% without condensation

Specifications subject to change without notice.

PCM 81

DIGITAL EFFECTS PROCESSOR

FEATURES

- World-class Lexicon reverb
- 2 DSP engines including Lexicon's proprietary LexiChip
- 24-bit A/D, D/A, and internal processing
- 300 presets; 50 User programs
- Studio-standard reverbs and effects including: Reverb and six voices of Delay or Reverb, four voices of Pitch Shift, and Vocal Fix for replacing out-of-tune melody lines
- A single Adjust knob that automatically attaches to all applicable parameters in factory tailored presets
- PCMCIA card slot for program storage
- Optional cards for algorithms, programs and User presets
- Pro and Go Modes: Go mode allows access to as many as 10 handpicked parameters for any particular preset; Pro Mode allows full access to an Edit matrix of as many as 100 parameters
- Dynamic Spatialization: 3D effects processing (compatible with surround sound systems)
- Up to 20 seconds of delay
- Balanced analog inputs and outputs (XLR and 1/4")
- S/PDIF and AES/EBU digital inputs and outputs
- Tap Tempo for instant setting of delay and modulation times
- Full MIDI control

The PCM 81 is equipped with an industry-standard PCMCIA card slot, allowing users to store personal programs and setups on RAM cards.

For over 30 years, Lexicon® has been considered the pioneer in digital audio effects. With more experience than any other manufacturer, it's no surprise that the PCM 81 Digital Effects Processor represents the most advanced systems in its class. Lexicon's unique dual DSP platform enables the PCM 81 to combine reverb with powerful, flexible effects.

EASY OPERATION

The PCM 81 is as simple to operate as it is pleasurable to listen to. Just load a preset and a useful parameter will become instantly available on the Adjust knob. The next level was designed for professionals who want to further customize programs, but lack the time to wade through the myriad of available controls. In this mode, as many as 10 of the most logical parameters in a given effect are easily accessible for customization. For the sound designer, another mode allows access to the full Editing matrix available in both units, as well as a user-assignable Soft Row in which to store favorite parameters. It also provides access to the extensive modulation capabilities of the PCM 81.

DYNAMIC PATCHING

The PCM 81 raises Dynamic Patching to a new level, providing unprecedented control over the effects module. Dynamic Patching gives these processors a truly unique set of capabilities, from modulating sounds, to producing unusual and ethereal spaces, to altering the attack and decay characteristics of the sounds. The Dynamic Patching matrix maps data from 143 possible control sources to any effect parameter. These sources include 126 different MIDI controllers, as well as external sources such as footswitches and footpedals. Internal controllers include Tempo (both internal Tap and external MIDI clock), LFOs (Sine, Cosine, Square, Triangle, Pulse, and Sawtooth), Time Switches, Latch, AR Generator, and Left and Right Envelope Followers. Up to 10 patches can be created per effect. In the Dynamic Patching matrix, eight pivot points can be established to create complex and interesting modulation paths.

TEMPO CONTROL

The PCM 81 offers Tap Tempo control of delay lines as well as several rhythmic variations on the tap. Tempo can be "dialed-in" in beats-per-minute, MIDI clock can be generated from tap, or received via MIDI from an external sequencer or drum machine. Tempo can also control LFO speeds and time switches, allowing modulations to be synchronized with music. Independent rhythmic values can be set for each parameter within the same program. The PCM 81 also offers more than 20 seconds of stereo delay.

EFFECTS

The PCM 81 has everything that made the PCM 80 the top choice among studio effects processors. In addition, it offers more effects, more algorithms, and full AES/EBU input and output. Two digital signal processors; Lexicon's proprietary LexiChip for reverb, and a second DSP engine to handle other effects, create versatile effect combinations without compromising sonic clarity. The PCM 81 offers more than all other processors in its class with; 24-bit internal processing, a true stereo signal path, balanced analog inputs

and outputs, AES/EBU and S/PDIF digital inputs and outputs, the power to combine analog and digital outputs, extensive modulation capabilities, and 300 presets. The PCM 81 is equipped with an industry-standard PCMCIA card slot, allowing users to store personal programs and setups on cards. For the PCM 81, adding Lexicon's Dual FX algorithm and specially designed preset cards for the PCM 80 (compatible with the PCM 81) increases its number of algorithms to more than 40 and presets to nearly 800.

SOUNDS

The PCM 81 boasts an enormous selection of sounds. Each combines uncompromised stereo reverb with several voices of additional effects. A full complement of Pitch Shifters provides unique special FX as well as doubling, quadruple tracking, Chorus, and Pitch Correction within a range of three octaves up or down. With 300 presets, the PCM 81 allows instant access to Pitch, Reverb, Ambience, Modulators, up to 20-second stereo Delays, and Dynamic Spatialization effects for 2-channel or surround applications. Its presets were designed to accommodate a wide range of applications, from effects designed for musical uses and recordings to effects designed specifically for pitch correction, sound effects, and video post-production.

ALGORITHMS

The PCM 81 utilizes 4-voice, 6-voice, and Pitch algorithms to create effects. The 4-voice algorithms; Chamber, Concert Hall, Infinite, Inverse, and Plate. Each combine a specific reverb type with a 4-voice stereo "effect toolbox" called the Reverb Shell, which provides post-processing for the reverb. For example, it is possible to produce a Ghost Flange by assigning a Modulated Delay to an Inverse Reverb (to Detune it). The 6-voice algorithms; Chorus + Reverb, Glide, Hall, Multiband + Reverb, Res 1 > Plate, and Res 2 > Plate, combine a specific reverb type with a specialized 6-voice stereo effect. In these algorithms, it is possible to combine the shimmer of a multi-voice chorus with a lush reverb tail (as in Wet Chorus). Seven algorithms include Pitch Correct for correction of monophonic sources, and Stereo Chamber for full stereo pitch-shifting with Chamber reverb. A powerful submixer is built into the Dual Chamber, Dual Inverse, and Dual Plate algorithms for flexible ordering and routing of two independent voices of pitch-shifting with reverb. A 4-voice Quad > Hall algorithm provides four independent pitch-shift voices with full stereo reverb, and a VSO-Chamber algorithm provides stereo time and pitch correction with Chamber reverb and variable speed pitch control (in percent). Two independent spatial processors accommodate the placement of effects virtually anywhere between or beyond the loudspeakers. Effects can also be located dynamically, creating different spaces that change with the music.

SPECIFICATIONS

Analog Audio Input:	XLR and 1/4" balanced (T/R/S)
Input Level:	-2dBu to +20dBu, balanced; -22dBu to 0dBu, unbalanced
Input Impedance:	100kΩ, balanced; 50kΩ, unbalanced
Analog Audio Output:	XLR and 1/4" T/R/S balanced
Output Level:	+18dBm, balanced; +4dBm, unbalanced
Output Impedance:	125Ω, balanced
Digital Audio Input:	XLR, AES/EBU; Coaxial RCA, S/PDIF
Format:	AES/EBU, balanced; S/PDIF, unbalanced
Sample Rate:	44.1kHz and 48kHz
Digital Audio Output:	XLR, AES/EBU; Coaxial RCA, S/PDIF
Format:	AES/EBU, balanced; S/PDIF, unbalanced
Sample Rate:	44.1kHz and 48kHz
Conversion:	24-bit A/D; 24-bit D/A
Internal Audio DSP:	24-bit
Frequency Response:	10Hz - 20kHz ±.5dB
Crosstalk:	-55dB 10Hz to 20kHz
HD:	<0.006%, 10Hz - 20kHz
Dynamic Range:	D/A: >98dB typical, 10Hz - 20kHz A/D: >105dB typical, 10Hz- 20kHz A/A: >96dB typical, 10Hz - 20kHz
MIDI Interface:	5-pin DIN connectors for MIDI IN, THRU and OUT
Footswitch:	1/4" T/R/S connector for two independent momentary footswitches (system detects normal-open, or normal-closed on power up)
Foot Controller:	1/4" T/R/S connector for foot pedal (100Ω - 10kΩ impedance)
Memory Card Connector:	PCMCIA type l cards, 68 pins
Standards:	PCMCIA 2.0 and JEIDA 4.0
Card Format:	Up to 1MB of SRAM (attribute memory not required)
Power Requirements:	100 - 240 volts AC; 50 - 60Hz; 35 watts Automatic switching (3-pin IEC connector)
Dimensions:	19" W x 1.75" H (1U) x 12" D (483 x 45 x 305 mm), rack mount standard
Weight:	6.4 lbs. (2.9 kg)
Operating Temperature:	32° to 104°F (0° to 40°C)
Maximum Humidity:	95% without condensation

Specifications subject to change without notice.

PCM 91

DIGITAL REVERBERATOR

FEATURES

- World-class Lexicon reverb
- Dual proprietary LexiChip DSP engines
- 24-bit A/D, D/A, and internal processing
- 450 presets; 100 User programs
- Lexicon's "Greatest Hits" reverb algorithms, including Chamber, Concert Hall, Random Ambience, Random Hall, Rich Plate, and Split Chamber
- Keyword search for quick, intelligent sorting of presets by alphabetical order or application
- Front panel Adjust knob automatically attaches to one or more parameters in each factory tailored preset
- PCMCIA card slot for program storage
- Pro and Go Modes: Go mode allows access to as many as 10 handpicked parameters for any particular preset; Pro Mode allows full access to an Edit matrix of as many as 100 parameters
- Balanced analog inputs and outputs (XLR and 1/4")
- S/PDIF and AES/EBU digital inputs and outputs
- Tap Tempo for instant setting of delay and modulation times
- Full MIDI control

The PCM 91 is equipped with an industry-standard PCMCIA card slot, allowing users to store personal programs and setups on RAM cards.

From the company that invented digital reverb more than 30 years ago, comes the PCM 91 Digital Reverberator, the most advanced systems in its class. Lexicon's® unique dual DSP platform enables the PCM 91 to offer the highest quality reverb.

EASY OPERATION

The PCM 91 is simple to operate and a pleasure to listen to. Just load a preset and a useful parameter will become instantly available on the Adjust knob. The next level was designed for professionals who want to further customize programs, but lack the time to wade through the myriad of available controls. In this mode, as many as 10 of the most logical parameters in a given effect are easily accessible for customization. For the sound designer, another mode allows access to the full Editing matrix available in both units, as well as a user-assignable Soft Row in which to store favorite parameters. It also provides access to the dynamic reverb aspects of the PCM 91.

DYNAMIC PATCHING

The PCM 91 raises Dynamic Patching to a new level, providing unprecedented control over the effects. Dynamic Patching gives this processor a truly unique set of capabilities, from modulating sounds, to producing unusual and ethereal spaces, to altering the attack and decay characteristics of the sounds. The Dynamic Patching matrix maps data from 143 possible control sources to any effect parameter. These sources include 126 different MIDI controllers, as well as external sources such as footswitches and footpedals. Internal controllers include Tempo (both internal Tap and external MIDI clock), LFOs (Sine, Cosine, Square, Triangle, Pulse, and Sawtooth), Time Switches, Latch, AR Generator, and Left and Right Envelope Followers. Up to 10 patches can be created per effect. In the Dynamic Patching matrix, eight pivot points can be established to create complex and interesting modulation paths.

TEMPO CONTROL

The PCM 91 offers Tap Tempo control of delay lines as well as several rhythmic variations on the tap. Tempo can be "dialed-in" in beats-per-minute, MIDI clock can be generated from tap, or received via MIDI from an external sequencer or drum machine. Tempo can also control LFO speeds and time switches, allowing modulations to be synchronized with music. Independent rhythmic values can be set for each parameter within the same program.

PCM 91 PRESETS

The PCM 91 includes 450 presets which provide sounds for real-world applications. The most useful parameters of each sound are located in a user-definable Soft Row, allowing users to make quick and simple adjustments. Navigation is further simplified with labeled banks and rows. A unique KeyWord Search function in the PCM 91 enables users to locate a group of programs designed for a given application. There are 50 keywords in total, including four user-definable groups of effects.

PCM 91 REVERB

Lexicon's research into the physics of acoustics is embodied in the Random Hall algorithm. Echograms of real halls have dispelled the myth of pre-delay and early reflections. In actual spaces, there is no empty interval between the arrival of direct sound and the maximum reverb density fille by early reflections. Instead, ambience builds gradually, with diffuse and complex reflections that do not color the timbre of sound like fixed-delay taps. Random Hall's unique Shape, Size, and Spread parameters control the build-up and decay of the ambient envelope. Size determines how large the space will be. Shape controls the contour of the ambient build. Spread controls the duration of Shape, setting the build-up and sustain. Precision filters provide spectral control of reverberation time, and unique Spin and Wander parameters add random movement ensuring smooth reverberant decay. In the PCM 91, Lexicon's classic Concert Hall algorithm has been enhanced with Spatial EQ and a Compressor to increase its versatility. The Rich Plate algorithm provides simulated plate reverberation, as well as new variations on this classic effect. The Ambience algorithm provides effects tailored specifically to the post-production environment, permitting accurate matching of previously recorded ambience. This allows new elements to blend seamlessly, and sound effects, dialog, or music to be placed realistically at different positions in the "space." Each of the PCM 91 algorithms include selected tools for ambience, post-processing, and compression/expansion, as well as modulation and patching parameters common to each. Ten Dual Reverb algorithms are built-in to the PCM 91. These algorithms contain two independent reverb blocks to create superb Dual and Cascade-configured stereo reverbs, each with all the control features of the single effects.

PCM 91 CUSTOM CONTROLLERS

Control in the PCM 91 has been increased with the addition of four Custom Controllers placed on the Soft Row. These controllers consist of one or more parameters patched together, each with individual scaling values. Custom controllers effectively add four more Adjust knobs to each program.

SPECIFICATIONS

Analog Audio Input:	XLR and 1/4" balanced (T/R/S)
Input Level:	-2dBu to +20dBu, balanced; -22dBu to 0dBu, unbalanced
Input Impedance:	100kΩ, balanced; 50kΩ, unbalanced
Analog Audio Output:	XLR and 1/4" T/R/S balanced
Output Level:	+18dBm, balanced; +4dBm, unbalanced
Output Impedance:	125Ω, balanced
Digital Audio Input:	XLR, AES/EBU; Coaxial RCA, S/PDIF
Format:	AES/EBU, balanced; S/PDIF, unbalanced
Sample Rate:	44.1kHz and 48kHz
Digital Audio Output:	XLR, AES/EBU; Coaxial RCA, S/PDIF
Format:	AES/EBU, balanced; S/PDIF, unbalanced
Sample Rate:	44.1kHz and 48kHz
Conversion:	24-bit A/D; 24-bit D/A
Internal Audio DSP:	20-bit
Frequency Response:	10Hz - 20kHz ±.5dB
Crosstalk:	-55dB 10Hz to 20kHz
HD:	<0.006%, 10Hz - 20kHz
Dynamic Range:	D/A: >98dB typical, 10Hz - 20kHz A/D: >105dB typical, 10Hz- 20kHz A/A: >96dB typical, 10Hz - 20kHz
MIDI Interface:	5-pin DIN connectors for MIDI IN, THRU and OUT
Footswitch:	1/4" T/R/S connector for two independent momentary footswitches (system detects normal-open, or normal-closed on power up)
Foot Controller:	1/4" T/R/S connector for foot pedal (100Ω - 10kΩ impedance)
Memory Card Connector:	PCMCIA type I cards, 68 pins
Standards:	PCMCIA 2.0 and JEIDA 4.0
Card Format:	Up to 1MB of SRAM (attribute memory not required)
Power Requirements:	100 - 240 volts AC; 50 - 60Hz; 35 watts Automatic switching (3-pin IEC connector)
Dimensions:	19" W x 1.75" H (1U) x 12" D (483 x 45 x 305 mm), rack mount standard
Weight:	6.4 lbs. (2.9 kg)
Operating Temperature:	32° to 104°F (0° to 40°C)
Maximum Humidity:	95% without condensation

Specifications subject to change without notice.

960L

DIGITAL EFFECTS SYSTEM

The Lexicon Legacy: Welcome To the Future

THE SOUND OF TODAY – and Tomorrow. Multi-channel sound is home ground for Lexicon®. For almost as long as it has been a leader in professional digital audio technology, Lexicon has been at the forefront of consumer audio – its line of digital controllers and processors have become the envy of the consumer electronics industry.

It's no wonder – Lexicon Professional audio processors have led the way in multi-channel digital effects, with technologies like Logic 7 and 3DPM.

THIS IS THE FUTURE

The heart of the 960L is an upgradable DSP powerhouse with eight analog and digital inputs and outputs as standard. For users who work in all-digital environments, a "digital-only" I/O version (960L/D) is also available. The analog audio interface is fully-balanced with 24-bit, 96 kHz conversion, and there's AES/EBU digital I/O plus MIDI and word-clock in/out/thru. Plenty of room has been left to add interfacing options – for both additional I/O and control. A multiple D/A converter approach ensures that the noise floor is in line with the enviable performance of the DSP section. And it is here that the true capabilities of the 960L are hidden. A single DSP card carries a combination of the latest incarnation of Lexicon's proprietary LSI Lexichips™ and other industry standard DSP devices. These allow the 960L to be configured as a five-in/five-out plus two-in/five-out, 44.1/48 kHz surround reverb system or – four completely independent stereo machines or – a single 88.2/96 kHz system. When an optional, second DSP Reverb card is added to the system, the "DSP horsepower" doubles; this second card provides the potential of eight stereo of four surround reverbs at 48 kHz, or four stereo and two surrounds at 96 kHz. Since each reverb card can be configured separately, various combinations of reverb configurations are available, such as four stereos and 2 surrounds at 48 kHz or two stereos and one surround at 96kHz. With the second DSP card added, a total of sixteen independent inputs are available. Because the I/O is "mappable," the engineer has the ability to mix and match inputs and outputs in numerous configurations. Any DSP card input (as well as DSP output) may be routed to any physical output. For example, eight channels may be dedicated to one reverb DSP card while the other eight channels dedicated to the second. In fact, the 960L could be set-up as a 16 in X 16 out digital-only processor. Perhaps the most important and powerful capability of two DSP cards is the ability to cascade from one to the other. All inputs can be extensively panned and mixed – eliminating the absolute need to upgrade or invest in a new multi-channel console with dedicated surround panning. A CD-ROM drive allows for software upgrades to be added as easily as slipping in a disc.

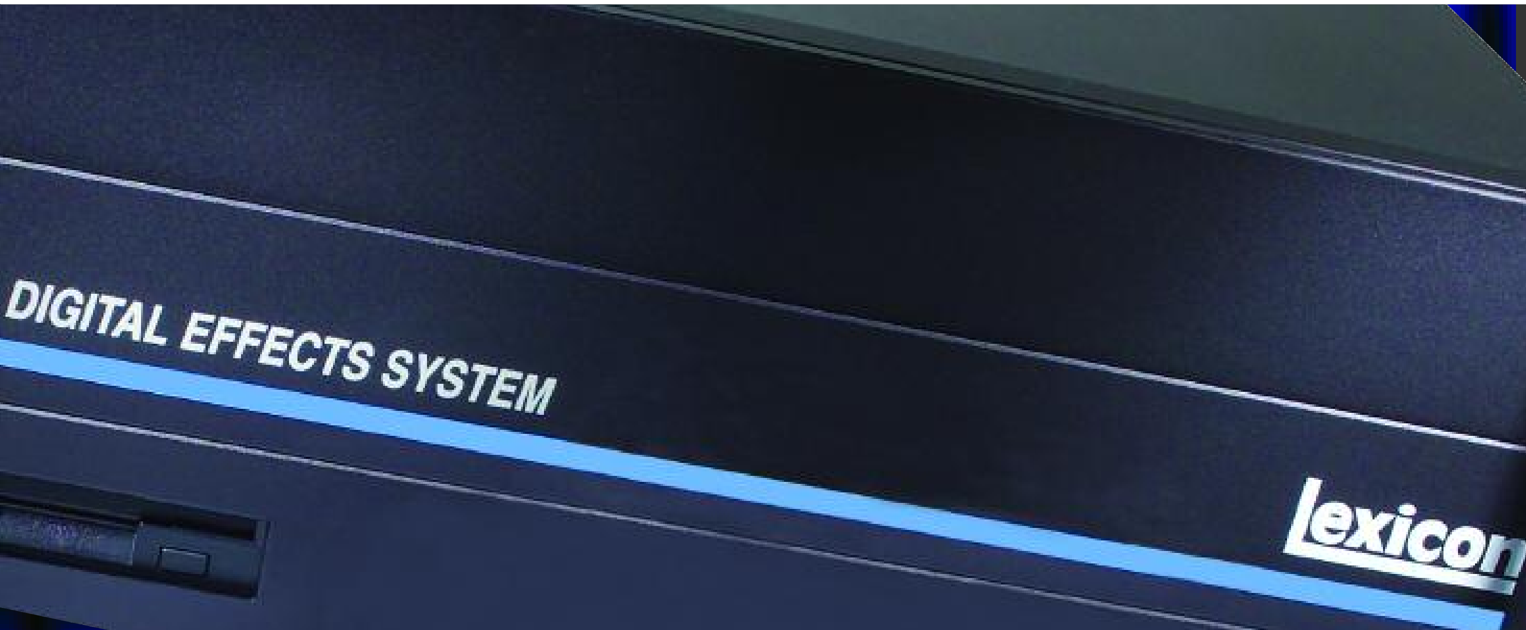
THREE-DIMENSIONAL SOUND: Lexicon 3DPM™

The concept behind the smooth new surround and stereo algorithms of the 960L lies at the very heart of the machine: in Lexicon's unique Three-Dimensional Perceptual Modeling – or 3DPM™. This new approach to reverberation sets the 960L distinctly apart from conventional physical modeling techniques. The smoothest sounding surround-sound reverb must not only work effectively in a sophisticated 5.1 audio environment, but also in stereo and even mono, without artifacts.

The secret: rather than simply model physical spaces, you model instead what the ear and brain hear, and expect to hear, about them – because in many cases, the modeling of real spaces compromises the quality of the listening experience. Reverberation energy within the first 300 milliseconds contains crucial auditory information. Proper utilization of psychoacoustic mechanisms during this time period are vital to creating spaciousness and depth without compromising the intelligibility and clarity of the sound source. The 960L's carefully-crafted 3DPM algorithms take these important factors into account and consistently out-perform traditional physical modeling techniques. For the first time the 960L's virtual surround spaces can sound even better than the real thing.

Trying to fit stereo recordings into a surround world? The 960L features Upmix™, a specially-tailored version of Lexicon's world-famous Logic7 technology. Upmix converts stereo source material into a convincing 5-channel result, with solidly-anchored panning, a stable center and discrete surrounds. Unlike competing products, Upmix gives a surround result that folds down nicely into stereo and mono-compatible results. Upmix can used on individual stereo tracks or on a complete stereo mix.

Thanks to a virtually complete de-correlation of reverb elements, the 960L's unique algorithms ensure those natural-sounding surround reverbs work equally well in stereo and mono – where, let's face it, much of your recordings will still be heard for years to come.



CONTROL OUTSIDE THE BOX

What's outside the mainframe is equally as impressive as what's inside – because the 960L is controlled by the LARC2 – a completely new remote with a vast array of new features.

At the center of the LARC2 are two elements: eight touch-sensitive motorized faders and a superb high-resolution color LCD display. Between these are eight soft keys associated with the display, a numeric keypad, cursor control, illuminated function keys, and a compact joystick. Above the bright screen are three LEDs for each channel which give you a quick and clear level indication.

For complex music or film mixes, the 960L provides a comprehensive automation package. Every single control of the 960-panning, reverb, program load, mute-can be automated and touched-up until the mix is perfect. Automation data is immune to changes in frame rate. Frame rate can be changed at any time, with no loss of accuracy in the automation. Automation data may be stored on the internal hard drive or transferred to diskette for storage with other source materials. It's like having a giant automated console inside a tiny LARC2 controller.

To get an impression of how the LARC2 places every aspect of the 960L at your fingertips, take a look at the main Program display: a row of eight parameters, one for each fader, shows you what main features of the program can be controlled. Touch an associated fader and the parameter is highlighted. To navigate the program matrix, use a cursor key or "+/-" key to move up or down a row. When any program is loaded, the faders snap instantly into position, just like moving faders on an automated console.

The high-resolution LCD display is easy to read, colorful and large enough to display machine status plus all the parameters for a preset, making it simple to cursor to the specific parameter you need to change. While the shorthand labels above the faders will be familiar to existing 480L LARC users, the parameter is fully described when selected and there is a full description at the top of the display.

The 960L's set of controls enable you to access all the features of setup quickly and easily. For rapid work, and if you've never used the unit before, an additional set of controls will appear on a "V-Page"; this allows access to all of the key aspects of a program. You can assign parameters to a "V-Page" for instant access – independently for each preset – with their names displayed over faders. The Edit Page gives the user the ability to wring out the finest detail of Lexicon's multi-faceted algorithmic possibilities. The eight faders also offer a "Fine" or vernier mode where they are set to the central position and you can use the whole length of a fader to make value additions or subtractions to a parameter.

While the joystick is easy to use as a means of setting inputs to virtual source positions, it is fully assignable and can be used as a surround panner. Move a source around in a virtual room and you will immediately appreciate both the power of Lexicon's 3DPM system.

The 960L is a powerful and sophisticated digital effects system, and we recognize the need to give you as much feedback as possible. As a result, we have implemented the most comprehensive context-sensitive help system you'll find on any digital effects processor to date. Help is easily accessed and visible thanks to the LARC2's large screen. You can also label your own bank and register creations, describe what they do and how to use them. In addition to more than 200 factory-programmed presets, you can store up to 1,000 User-Registers of your own (in Banks of 10). If you feel the need to write an extensive description, simply plug in a standard PS/2 computer keyboard. The 960L Mainframe will support up to two independent LARC2's for added flexibility and control.

HEARING IS BELIEVING

There are new and challenging requirements for multi-channel content creation. Whether you make records, create feature film soundtracks, or mix high-quality live sound, today's producer and engineer needs, expects and demands the best and most useful technology. When it comes to the 960L's multi-channel effects processing, hearing is believing.

Set the unit up in your studio, select one of the 960L's superb surround reverb algorithms and turn off the lights. You'll hear Lexicon's proprietary 3DPM modeling create a new acoustic environment. Move about inside the speaker array – or even beyond – and hear for yourself how large and enveloping the sweet spot is, how stable the imaging, and how realistic the sound. You'll immediately hear how the 960L redefines the digital effects standard for a new generation of surround-sound professionals and listeners alike – just as the ubiquitous Lexicon 480L is the standard by which reverbs have been judged for nearly two decades.

The Lexicon 960L. This Is The Future. Hear it today.

960L

DIGITAL EFFECTS SYSTEM

SPECIFICATIONS

Analog Input	
Connectors:	Eight, Female XLR
Impedance:	50kWΩ, balanced
Level (for 0 dbFS):	+24dBu
Frequency Response	@ 48kHz: 20Hz-20kHz, ±1dB
Frequency Response	@ 96kHz: 20Hz-40kHz, ±1dB
A/D Conversion:	24 bits, 128x oversampled
A/D Dynamic Range:	>110dB (20Hz-20kHz)
THD:	< .002%
CMRR:	>50dB
Crosstalk @ 1kHz:	< -100dB
Analog Output Connectors:	Eight, Male XLR
Impedance:	50WΩ, balanced
Level (for 0 dbFS):	+24dBu
Frequency Response:	@ 48kHz: 20Hz-20kHz, ±1dB
Frequency Response:	@ 48kHz: 20Hz-40kHz, ±1dB
D/A Conversion:	24 bits
8x oversampled	@ 44.1/48kHz
4x oversampled	@ 88.2/96kHz
D/A Dynamic Range:	>110dB (20Hz-20kHz)
THD:	< .002%
Crosstalk @ 1kHz:	< -100dB
A/A Performance	
Frequency Response	@ 48kHz: 20Hz-20kHz, ±1dB
Frequency Response	@ 96kHz: 20Hz-40kHz, ±1dB
Dynamic Range:	>107dB (20Hz-20kHz)
THD:	< .002%
Digital Audio I/O Connectors:	4 Male XLR Outputs; 4 Female XLR Inputs
Format:	AES/EBU
Word Size:	24-bits
Sample Rates	Internal: 44.1/48/88.2/96kHz
Accuracy:	Within ±10ppm Meets AES 11, Grade 2
External:	44.1/48/88.2/96kHz
Lock Range:	±1.5%
Synchronization	
TTL Word Clock Input:	75WΩ, BNC self-terminating loopthru
TTL Word Clock Output:	BNC
Clock Jitter:	Intrinsic Jitter and Jitter Gain: Exceeds AES3, Amendment 1
Control Interfaces	
Remote Control:	LARC2 ports (2)
MIDI:	In/Out/Thru 5-Pin DIN
Reverb Types	Ambience: 48K Stereo and Surround Chamber: 48K Stereo and Surround Plate: 48K Stereo and Surround
Reverse:	48K Stereo and Surround
Random Hall:	48/96K Stereo and Surround
Ambient Chamber:	48K Surround

Reverb Card Configurations	44.1/48K Performance: Stereo Machines (4) 2 In x 5 Out Machines (2) 5 In + 2 In x 5 Out Machines 88.2/96K Performance: Stereo Machines (2) 2 In x 5 Out Machines (1) 5 In x 5 Out Machines (1)
Internal Hard Disk Storage	Factory Programs: >200
	User Registers: 1,000
	Removable 3.5" Floppy Disk Storage
User Programs:	100
Power Requirements:	100-120/220-240 VAC, 50-60Hz, 300W max,
Connector:	3-pin IEC
Physical Specifications	19.0"W x 17.4"D x 7.0"H (483mm x 442mm x 178mm) (4 rack units)
Weight:	35 lbs.
Regulatory Approvals	FCC: Class A; CE: EN55103-1, EN55103-2; UL: UL1419; C-UL: C22.2; TUV: EN60065
Environment	Operating: 10 to 40 C Storage: -30 to 70 C Humidity: 95% max, non-condensing
LARC2 Specifications	
Display	Type: Passive Matrix LCD Resolution: 640 x 240 pixels Colors: 256 Backlight: CCFL (Fluorescent) Brightness: Software controlled Contrast: rear panel knob
LED Meter Bridge	Configuration: 8 channels x 3 levels
	Levels: -60dB, -6dB, -0.5dB (overload)
Control Surface	Faders: Eight, 60mm throw, motorized, touch sensitive
Joystick:	Two-axis
Dedicated Function Keys:	29 (12 backlit)
Soft Function Keys:	8
Connectors	960L: 9-pin D-sub
Auxiliary PS/2 Keyboard:	6-pin Mini-DIN
External Power:	Concentric, 2.5mm
Power Requirements:	12 VDC, 2 Amps (max)
Physical Specifications	12.7"L x 8.25"W x 5.0"H (323mm x 210mm x 127mm)
Weight:	4 lbs.
Regulatory Approvals	FCC: Class A; CE: EN55103-1, EN55103-2; TUV: EN60065
Environment	Operating: 5 to 40 C Storage: -30 to 70 C Humidity: 95% max, non-condensing
Operating Distance	Powered from 960L Mainframe: Up to 100 feet
With External Power:	Up to 1,000 feet

Specifications subject to change without notice.


960L DIGITAL EFFECTS SYSTEM
lexicon
lexicon
Machine 1
B01 P1: Large Hall
PROGRAM
REGISTER
BANK
STORE
EDIT
CONTROL
MACHINE
ENTER
JOYSTICK
FINE ADJ

A Harman International Company

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