DM1612

TECHNICAL DATA

Digital Matrix Processor

Feature Highlights

- **16 mic/line inputs, 12 line outputs (4 outputs are mic/line switchable)**
- **16 x 12 digital matrix mixing; gain is adjustable from -69 to +20 dB**
- **Proportional gain auto mixing algorithm with AutoSkew™ - US Patent 5,414,776**
- **Auto mixing operates at the output of the matrix - each input channel can participate differently in each output mix**
- **Each output has its own NOM mixing bus**
- **6 filter stages plus compressor, ADFE and delay on each input**
- **9 filter stages plus compressor, limiter and delay on each output**
- **128 macros available for storing up to 64 commands each**
- **USB and RS-232 interfaces available for setup and control**
- **Digital I/O ports for "daisy chaining" and to connect other LecNet 2 devices**
- **AMX® and *CRESTRON*™ compatible***

The DM1612 is a powerful digital audio processing, mixing and routing system. The primary applications are sound reinforcement and conferencing in boardrooms, courtrooms, worship centers, distance learning systems, hotels and other applications with multiple microphones and loudspeakers. The design represents a milestone in DSP technology in its basic architecture and in its processing speed and efficiency.

The basic architecture consists of 16 mic/line inputs and 12 outputs with a full DSP-based crosspoint matrix that allows every input to be routed to any or all outputs. Following the A-D conversion at each input, the signal passes through multiple DSP filters, ADFE (automatic digital feedback eliminator), a compressor and a digital delay. In the matrix, gain is adjustable from -69 to +20 dB in 1 dB steps at each crosspoint. Each of the 12 outputs provides a digital delay, multiple DSP filters and a compressor/limiter.

The DSP functions include a patented automatic mixing algorithm* that applies NOM attenuation to the input channels assigned to each output bus. The participation of each input channel can be set to one of four modes: Auto, Direct, Override and Background. Since the mixing activity is applied at the outputs, each input can participate differently at every output.

The auto mixing process uses a seamless algorithm that eliminates gating and its ill-effects. Gain is proportioned amongst all inputs assigned to a particular output channel in a seamless and continuous manner based upon microphone activity. The algorithm operates in a natural, transparent manner and incorporates an adaptive AutoSkew™ process to eliminate artifacts such as comb filtering and abrupt gating that occur with conventional automatic mixing schemes. There is never a late mic, nor missed syllable in conversation and speech, and background noise and audio recirculation are minimized.

The matrix mixer enables complex signal routing and level controls without limitations. The matrix mixing allows "mix-minus" routing to reduce acoustic feedback and eliminate echoes caused by speaker/mic coupling in teleconferencing applications. The 16x12 architecture actually functions like 12 separate automatic mixers, each with its own NOM mixing bus.

Up to 128 macros can be created with up to 64 commands per macro. These macros can be created off-line and uploaded through either the front panel USB or the rear panel USB and RS-232 serial interfaces.

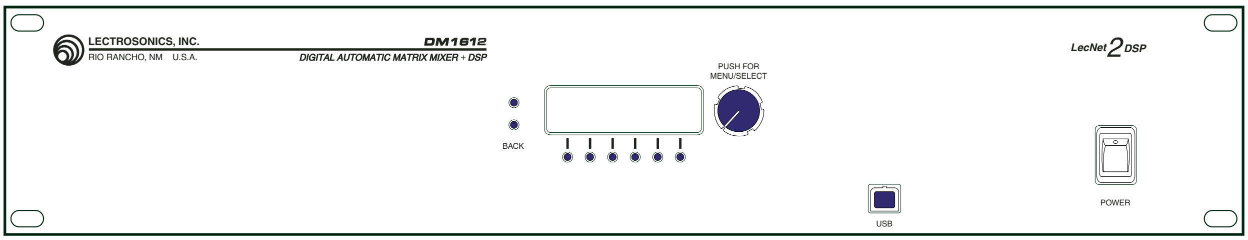
*See references on back page

Control signals are distributed through multiple units via standard CAT5 cables with RJ45 connectors. Audio is distributed through the EXPANSION ports on the rear panel.

Eight different types of filters are available: Low and High Pass (Bessel and Butterworth types), Band Pass, High and Low Shelving and Parametric EQ, plus an OFF position. Six filters are provided at each input and nine filters at each output. In addition, each input also includes 6 notch filters that can operate as fixed or ADFE (Automatic Digital Feedback Elimination) types.

There is no "gas gauge" or "resource meter" necessary when configuring a system setup. The power and elegance of the overall design allow the automatic mixing, matrix mixing, plus all filters to be used on all 16 inputs and all 12 outputs at the same time with an overall latency of less than 2 ms. The latency in a single unit does not change regardless of the number of functions, processes or features selected. Even with 8 units stacked for 128 inputs, the overall latency is still less than 3 ms.

Front Panel



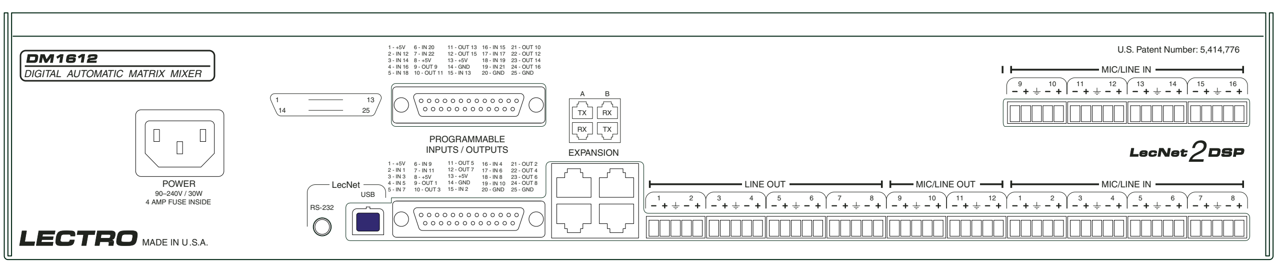
The DM1612 chassis is a 2RU configuration made of machined aluminum, with a rugged electrostatic powder coated finish and laser engraved nomenclature for high visibility.

The front panel provides a graphical LCD interface that allows access and control of settings and adjustments. A USB port is also provided on the front panel for more complete, computerized setup adjustments and control without having to access the rear panel.

The LCD display and a rotary encoder control knob enable access for fine adjustments without the use of a computer. Switches next to the LCD display simplify the user interface.

While the overall configuration is normally created with the software and downloaded into the DM1612, it is often very convenient to allow fine adjustments to be made after the hardware is installed.

Rear Panel



The rear panel of the DM1612 provides all connections for microphones, line inputs, line outputs, external computer control, and EXPANSION ports for "daisy chaining" multiple units or connecting other LecNet2™ devices. Logic I/O ports are also provided for remote control and signaling with external pots, switches and LEDs.

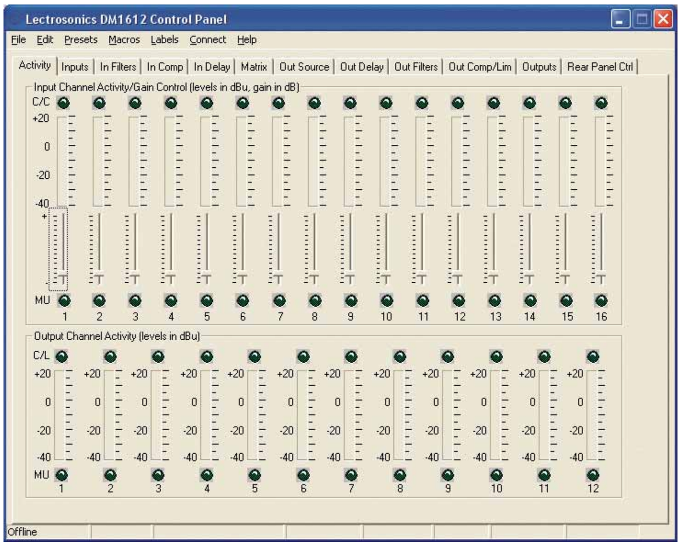
Industry standard depluggable connectors are used for audio inputs and outputs, DB25 connectors provide taps for logic control and USB and serial jacks for computer interface. A quad RJ45 connector block provides expansion ports for "daisy chaining" multiple units, or interfacing with another LecNet2™ device.

LecNet2™ Software

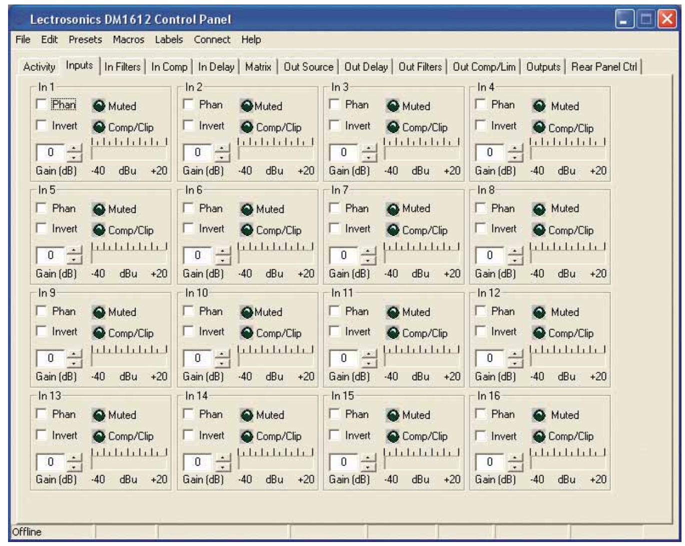
Software is included with the DM1612 and available for download from the Web site at: *www.lectrosonics.com*. The software is used primarily for setup, with the configuration saved on file and into the unit's memory for actual operation. Once configured, the DM1612 runs without a host computer.

The software is user-friendly, with a variety of screens provided for each section of the signal flow and system design. The software runs under Windows®2000 and XP operating systems using a familiar tabbed layout. A few sample screens are shown below as an example of the appearance on-screen.

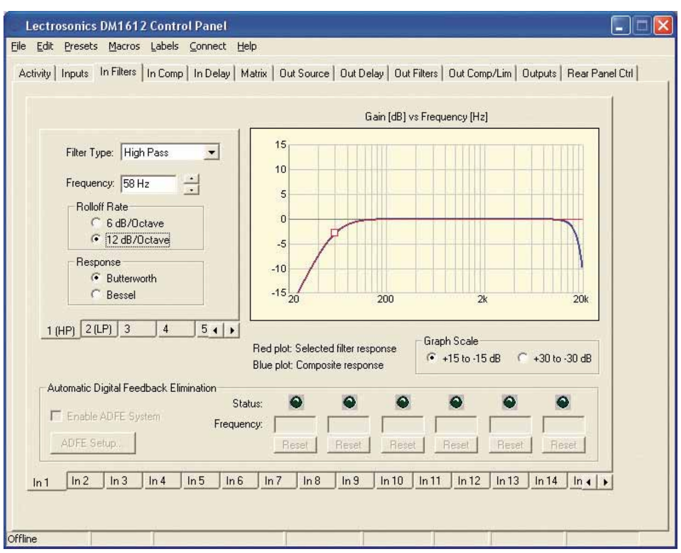
Sample Screens



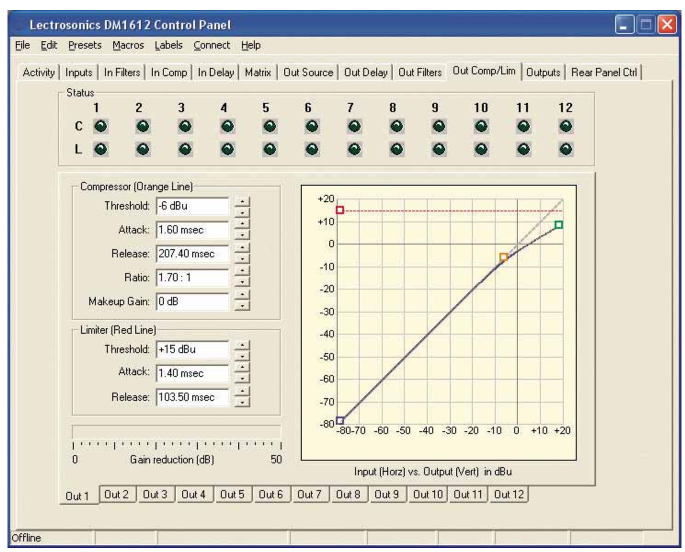
Input and Output signal levels are displayed in a real-time display to monitor the audio activity.



Input settings and gain adjustments are displayed on a single screen showing all channels.



The Input Filters tab provides a display for each input channel, with tabs on the lower section to pop up the individual channels.



The Compressor/Limiter tab provides a display that enables values to be adjusted with numeric values entered directly, scroll bars to make value adjustments, and dynamic "click and drag" control points on a graphical display.

Specifications and Features

Audio inputs	
Gain:	-10 dB to +60 dB, programmable in 1 dB steps
Input impedance:	2.5 k Ohms
Phantom voltage:	15V, programmable
Connector:	5-pin Phoenix
Audio outputs:	Floating balanced, either side can be grounded
Nominal level:	0 dBu all outputs, -40 dBu selectable on outputs 9 through 12
Output impedance:	
Line Only:	• 50 Ohms differential line level only outputs
Mic/Line	• 600 Ohm differential programmable outputs at line level • 130 Ohm differential programmable outputs at microphone level
Input Dynamic Range:	96 dB at -50 dBu input level 102 dB at all other levels (unweighted 20 - 20 kHz)
Output Dynamic Range:	105 dB (unweighted 20 - 20 kHz)
Audio Performance:	
IMD + noise:	0.1% max. 0.02% nominal input level
THD + noise:	0.1% (worst case) 0.02% nominal input level
EIN:	-126 dBu

Connectors:	
Audio I/O:	5-pin Phoenix
Expansion:	RJ45
Logic I/O:	DB25
Serial:	Standard USB and mini TRS
Proprietary network	
Physical level:	LVDS (Low Voltage DIfferential Signal) high speed
Connector:	four RJ-45
Cable quality:	CAT-5
Transmission speed:	50 Mbits/s
Programmable control inputs	
Number of inputs:	22
Analog voltage range:	0-5V
Logic input:	TTL, LVTTL, CMOS, LVCMOS
Programmable control outputs	
Number of logic outputs:	16
Logic control:	active low
Max sink current:	100 mA
Max supply voltage:	40 V
Supply voltage for control I/O:	5 V
Max current:	750 mA
Power requirements:	100-240 VAC, 47-63 Hz
Power consumption:	30 Watts

**References from page 1*
US Patent Numbers: 5,414,776 and 5,402,500
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