TECHNICAL DATA

DM Series Digital Matrix Processors

DM812 DM1612F DM1624F



- Patented proportional gain auto mixing*
- Optimized Architecture[™] for unlimited DSP resources no "Gas Gauge"
- 6 filter stages plus compressor, ADFE and delay on each input
- 9 filter stages plus compressor, limiter and delay on each output
- 128 global macros available with up to 64 commands per macro
- AMX[®] and CRESTRON[®] compatible
- Simultaneous LecNet2 and Third-Party control

The DM Series is a powerful hardware/software family of components providing digital audio processing, mixing and routing. The primary applications are sound reinforcement and conferencing in boardrooms, courtrooms, worship centers, distance learning, hotels and other applications with multiple microphones and loudspeakers. Easily configurable using a computer system with Windows[®] XP, Vista or 7 operating systems, the DM Series design represents a milestone in DSP technology in its basic architecture and processing speed, with outstanding and scalability and flexibility.

The three models shown here provide 8-in/12-out, 16-in/12-out and 16-in/24-out models which may be stacked in a master/slave configuration to expand the number of inputs for any size system. The number of outputs is limited to the number of outputs on the model used as the system master.

Versatile, Easy Control

The LecNet2 communications protocol supports simultaneous use of USB and RS232 ports by the PC software control panel and a third party serial control system such as AMX[®] or Crestron[®]. Installers can use the GUI to monitor the state of the DM device for the purpose of verifying that commands sent from the 3rd party controller (over RS232) are working correctly.

Minimal Latency

Studies have shown that even very slight delays in speech can affect the tempo of a conversation or a conference. The effect is sometimes subtle but delays over 10mS can cause real problems – something we have all experienced when talking on a cell phone. The throughput latency of the Lectrosonics DM Series products is a scant 2 ms, independent of how much processing is being used.

Optimized Architecture™

Some digital matrix processors give you a "gas gauge" or "resource meter" to show you how much processing power remains available. As you add more input modules, delays, filters, etc. you eventually run out of available DSP resources. There is no gas gauge in the Lectrosonics DM Series products, simply because they don't need one. No matter how many filters, delays, or other processing you engage, or how many signals are routed through your DM Series unit, you will never run out of DSP resources.

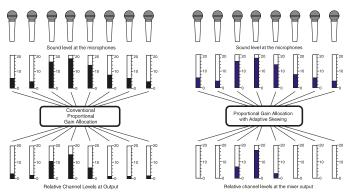


Scalabililty

Some DSP units limit you to a bus of only 8 channels, and some give you up to 16. Either way, most systems limit the amount of scaling available within the matrix. With the Digital Audio Network Interface (DANI[™]) bus, the Lectrosonics DM processors have unlimited input scalability. This way, you can design the right system for the job, no matter how complex and sophisticated, without the concern about being limited by the equipment.

Patented Automatic Mixing

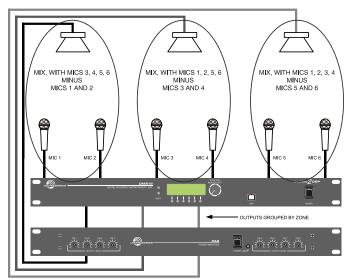
The LecNet 2 automatic mixing algorithm eliminates abrupt switching (gating) and adapts to changes in background noise in the acoustic environment. AutoSkewing[™] lends a subtle but effective priority to the most active channels by further reducing the levels of inactive channels to reduce background noise, and also prevents comb filtering by never allowing two competing channels to operate at the same level.



The auto mixing takes place at the crosspoint in the DSP matrix, which allows each input channel to behave differently at each output. The participation of each input channel can be set to one of four modes: Auto, Direct, Override, Background plus a special Phantom mode. Since the mixing activity is applied at the outputs, each input can participate differently at every output. For example, the microphone on input channel 3 can participate in an auto mixing mode at outputs 4, 5 and 6 for sound reinforcement, and also provide a continuous, fixed level at outputs 14 and 16 to feed a recorder.

The Signal Chain

The basic architecture consists of mic/line inputs and outputs with a DSP-based full 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 output provides a digital delay, multiple DSP filters and a compressor/limiter. 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 16x24 architecture actually functions like 24 separate automatic mixers, each with its own NOM mixing bus.



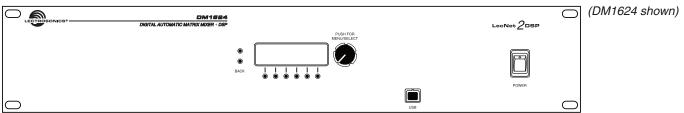
Up to 128 macros can be created with up to 64 commands per macro. These macros can be created off-line and uploaded through either the front panel USB or the rear panel USB and RS-232 serial interfaces.

Control signals are distributed through multiple units via standard shielded 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. Optimized Architecture[™] is a unique design that allows automatic mixing, matrix mixing, plus all filters to be used on all 16 inputs and all 24 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



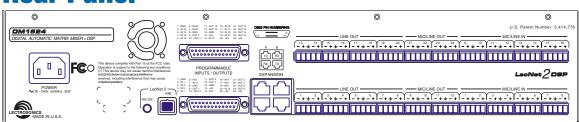
The DM Series chassis is made of machined aluminum, with a rugged electrostatic powder coated finish and laser engraved nomenclature for high visibility.

The front panel provides a graphical LCD interface (except on the DM84) 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.

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 DM, it is often very convenient to allow fine adjustments to be made after the hardware is installed.



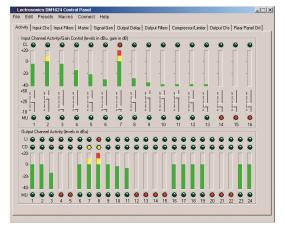
The rear panel of DM Series Matrix Mixers provide all connections for microphones, line inputs, line outputs, external computer control, and EXPANSION ports for "daisy chaining" multiple units or connecting other LecNet2[™] devices. Logic I/O ports are also provided for remote control and signaling with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

LecNet2 Software

LecNet2[™] Software is included with all DM Series Processors and available for download from the website 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 DM runs without a host computer. Industry standard depluggable connectors are used for audio inputs and outputs, DB25 connectors provide taps for logic control. USB and serial jacks are provided for the computer interface. A quad RJ45 connector block provides expansion ports for "daisy chaining" multiple units, or interfacing with another LecNet2[™] device.

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® XP, Vista and 7 operating systems using a familiar tabbed layout. Two sample screens are shown below as an example of the appearance.

Sample Screens



ivitu Input Chs Input Filters	Matrix Signal Gen Dutput De	day Output Filters Compressor	Limiter Output Chs Rear Panel (
	1.5 1.1		
In 1 ✓ Phan	In 2	Fhan Muted	In 4
			Phan S Muted
Invert O Clipping	Invert Olipping	Invert Olipping	Invert Olipping
+27 +	+40 ÷	0 +	0 ÷
āain.(dB) ⊶40 dBu +20	Gain (dB) -40 dBu +20	Gain (dB) -40 dBu +20	Gain (dB) -40 dBu +20
In 5	- In 6	- In 7	In 8
Phan 🕥 Muted	F Phan \Theta Muted	🗖 Phan 💊 Muted	🗏 Phan 💊 Muted
Invert \ominus Clipping	🗆 Invert 💊 Clipping	🗆 Invert 🥥 Clipping	🗐 Invert 💊 Clipping
+27 +			+27 -
āain (dB) -40 dBu +20	Gain (dB) -40 dBu +20	Gain (dB) -40 dBu +20	Gain (dB) -40 dBu +20
In 9	- In 10	- In 11	- In 12
Phan \Theta Muted	Finan OMuted	Finan \varTheta Muted	F Phan 💊 Muted
Invert O Clipping	T Invert O Clipping	T Invert O Clipping	Invert O Clipping
+44 -	+41 -	+44 -	+26 -
Gain (dB) 40 dBu +20	Gain (dB) -40 dBu +20	Gain (dB) -40 dBu +20	Gain (dB) -40 dBu +20
In 13	- In 14	- In 15	- In 16
Phan 🙆 Muted	E Phan 🔴 Muted	Finan 🔴 Muted	E Phan 🔴 Muted
Invert O Clipping	T Invert	T Invert	Invert O Clipping
+10 -	F10 -	F10 -	+10
ain (dB) -40 dBu +20	Gain (dB) -40 dBu +20	Gain (dB) -40 dBu +20	Gain (dB) -40 dBu +20

Stacking Multiple Units

	Audio Final Mix and Data Back Propagation
Master	
Slave	
Slave	
Audio Submix and Data Forward Propagation	

Control signals and audio are distributed through multiple units via shielded CAT5 cables.

Specifications and Features

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Audio inputs		Proprietary network		
Gain: Input impedance:	-10 dB to +60 dB, programmable in 1 dB steps 2.5 k Ohm	Physical level:	LVDS (Low Voltage DIfferential Signal) high speed	
Phantom voltage:	15 V, programmable	Connector:	Four RJ-45	
Connector:	5-pin Phoenix	Cable quality:	Shielded CAT-5	
Audio outputs:	Floating balanced, either side can be grounded 0 dBu all outputs, -40 dBu selectable on outputs 9 theorem 12	Transmission speed:	50 Mbits/s	
Nominal level:		Programmable control input	ts	
Output impedance:	through 12	Number of inputs:	22	
Line Only:	 50 Ohms differential line level only outputs 	Analog voltage range:	0-5V	
Mic/Line	 600 Ohm differential programmable outputs at line level 130 Ohm differential programmable outputs at microphone level 	Logic input:	TTL, LVTTL, CMOS, LVCMOS	
		Programmable control outputs		
		Number of logic outputs:	16	
Input Dynamic Range:	96 dB at -50 dBu input level 102 dB at all other levels (unweighted 20 - 20 kHz)	Logic control:	active low	
		Max sink current:	100 mA	
Output Dynamic Range:	105 dB (unweighted 20 - 20 kHz)	Max supply voltage:	40 V	
Audio Performance:		Supply voltage for control I/O:	5 V	
IMD + noise:	0.1% max. 0.02% nominal input level 0.1% (worst case) 0.02% nominal input level	Max current:	750 mA	
THD + noise:		Power requirements:	100-240 VAC, 47-63 Hz	
		Power consumption:	30 Watts	
EIN:	-126 dBu	Fan Cooling:	Continuous: DM1612F and DM1624F only	
Connectors:		Dimensions:	•	
Audio I/O: Expansion:	5-pin Phoenix RJ45	DM812	(1.75 in. high) (7.5 in. deep) (19 in. wide)	
Logic I/O:	DB25	DM1612F	(3.5 in. high) (7.5 in. deep) (19 in. wide)	
Serial:	Standard USB and mini TRS	DM1624F	(3.5 in. high) (7.5 in. deep) (19 in. wide)	
		Weight:		
Specifications subject to change without notice		DM812	(3lb 12oz) (1.7012 kg)	
		DM1612F	(5lb 4oz) (2.3814 kg)	



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DM1624F

(5lb 6oz) (2.4381 kg