DMTH4

TECHNICAL DATA

Digital Telephone Hybrid

- **3-in/24-out digital matrix architecture**
- **Fully integrates with DM Series processors**
- **Telephone, codec and auxiliary inputs and outputs**
- **Two Acoustic Echo Cancellers - 126 ms tail time**
- **Line echo canceller - 30 ms tail time**
- **6 filters plus compressor on each input**
- **6 filters plus compressor/limiter on each output**
- **USB and RS-232 interfaces for setup and control**
- **Fully balanced audio signal flow through entire system - no pin 1 problem**
- **Digital I/O ports for "daisy chaining" and to connect other LecNet 2 devices**
- **Proportional gain auto mixing algorithm with AutoSkew™ - US Patent 5,414,776**

The DMTH4 integrates telephone lines, video codecs and external audio sources into the digital bus structure of DM Series processors so these sources operate as though they are another microphone or audio input in the sound system. The unit is much more than just a telephone interface. Instead, it is a complete DM Series digital matrix processor, with a 3-in/24-out digital matrix, automatic mixing and comprehensive signal processing on every input and output. In essence, it simply connects to telephone lines, video codecs and external audio sources instead of mic/line inputs and outputs and integrates seamlessly with DM Series matrix processors.

The primary applications are in sound reinforcement and conferencing systems 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 challenge in teleconferencing using a sound system on one or both ends of a conference is to minimize echo heard at the far end caused by the coupling between loudspeakers and microphones in the local sound system. As sound from the far end enters the local sound system and is delivered by the loudspeakers in the local room, it will enter the local microphones and be sent back to the far-end. At the far-end the listeners will hear an echo of their own speech.

The integration of adaptive gain proportional auto mixing* with an all new proprietary echo canceller provides a remarkable solution that is as easy to install and set up as it is effective. Echo-free teleconferencing and clean local sound reinforcement is provided even in poor acoustical environments.

The DMTH4 shares the large digital matrix bus with other DM Series processors to handle a wide range of sound system requirements from a modest boardroom to large systems with hundreds of inputs. Multiple units can be stacked with multiple DM processors to handle very large systems with multiple phone lines.

Extensive control capability is built into the unit with an intuitive command structure to allow external control with USB or RS-232 connections. Up to 128 macros can be stored in internal memory. Each macro can contain up to 64 commands, with 115 characters in each command. A built-in macro recorder greatly simplifies the creation of and use of macros.

**US Patent 5,414,776*

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Rio Rancho, NM, USA
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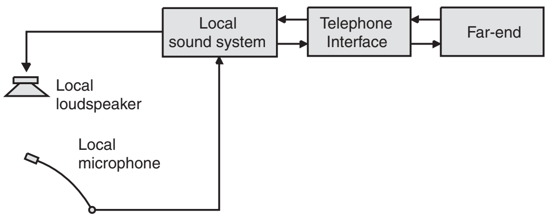
Echo and Echo Cancellation

The fundamental problem with microphone/speaker acoustical coupling is illustrated below. Far end audio is delivered by the loudspeakers in the room and the microphones pick it up and return it to the far end. The delay through this process creates an echo heard on the far end.



There are several methods used to reduce or eliminate the echo heard on the far end of the conversation:

- Optimal design in the sound system to minimize the coupling between loudspeakers and microphones.
- Mix-minus matrix routing.
- Automatic microphone mixing.
- Digital echo cancelling.

Matters become more complex when the sound system is required to provide both teleconferencing and sound reinforcement. A gain proportional automatic mixing process is widely recognized as the optimum solution for sound reinforcement, but it places significant demands on an acoustic echo canceller used for teleconferencing.

The matrix mixer enables complex signal routing and level controls without limitations. The matrix mixing allows **"mix-minus"** zoning of microphones and loudspeakers to decouple them and reduce or eliminate acoustic feedback and echoes. NOM attenuation is applied by the DSP at the crosspoints in the matrix, which essentially provides 24 separate automatic mixers, each with its own NOM mixing bus. Four different mixing modes can be selected at the crosspoint for each input, so each input can participate differently in each output mix.

The **automatic mixing** process uses a seamless algorithm that eliminates gating and its ill-effects. Gain is proportioned among 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. Audio from the far-end of a conference participates in the local mixing algorithm just like a microphone in the local sound system.

Two digital acoustic echo cancellers are provided in the DMTH4 to further reduce the return of local signals to the far-end. One operates on the telco connection and the other is dedicated to the video codec connection. In conjunction with the automixing process, echoes are minimized and not heard at the far end.

ERL

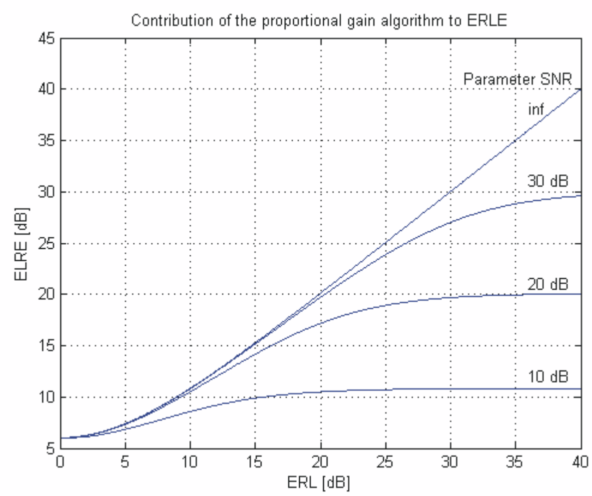
ERL (echo return loss) refers to the natural attenuation of the far-end audio signal as it circulates from the far-end through loudspeakers and microphones in the local sound system and back to the far-end. Good design in the local sound system will reduce the acoustic coupling between loudspeakers and microphones using physical placement and mix-minus matrix routing. Depending upon room size and acoustics, it is often impossible to achieve adequate decoupling to avoid an echo heard by the far-end during a teleconference. Thus, other types of processing are needed to further reduce the return echo.

ERLE

ERLE (echo return loss enhancement) refers to additional circuits and processes used to further increase ERL. Common methods are to use automatic mixing and digital echo cancellation.

Return Loss Enhancement

The gain proportional automatic mixing algorithm* in the DM Series processors not only provides seamless mixing for local sound reinforcement without abrupt gating, but it also contributes significantly to ERLE. The additional contribution is plotted in the following graph.



Digital echo cancellation is another method of reducing the echo delivered to the far-end. The concept, described in very simple terms, is to have the DSP recognize the far-end audio and subtract it from the transmitted audio to remove any echo they might hear at the far-end. Sounds simple, but in a sound system with multiple microphones and loudspeakers, it is not easy to identify the far-end audio in the complex mix of local sound, local noise and the effects of the room on the far-end audio delivered by the local loudspeaker system. When there is no sound or noise in the local room, the DSP can do a decent job of identifying the far-end audio and subtracting it from the transmitted signal, but this is rarely the case in full duplex teleconferencing. People talk, laugh and make noise, and air conditioners and projectors make noise, etc.

In a simple sound system arrangement, the local microphone can be muted when nobody is talking in the local room. A simple gated mixer can provide this function. With no open microphones locally, there is obviously no return echo signal. This requires that a threshold level be set high enough to keep the microphone from being opened by background noise, but low enough to allow it to open when someone speaks. When the local microphone is open, a return echo path is created, which is when a DSP echo canceller is needed. Given the wide variety of human voices and the dynamics of noise in a meeting room, a gated mixer is often not the best choice.

Using a dedicated DSP echo canceller on each input of the local mixer (referred to as "distributed echo cancellation") is an expensive but effective approach to reducing the return echo. The process requires the algorithm to "converge," which is to identify the far-end audio and subtract it from the signal sent to the far-end. This requires at least a brief moment when there is very little local sound or noise, with significant far-end audio present in the room. If nobody moves and there are no gain changes made to local microphones and loudspeakers, it is possible (in theory) to effectively remove return echo, but this is not a very realistic situation.

The theory behind distributed echo cancelling is that once the DSP has converged, it can continue to subtract far-end audio even when the local microphone is open and far-end audio is present at the same time. If there are any changes in gain, noise or acoustics in the local space and equipment, the DSP must re-converge, which requires another brief moment with little or no local noise or sound, and significant far-end audio present.

A gated automatic mixer does not change the gain when the microphone is open, it just turns the channel off and on abruptly. This helps with distributed echo cancelling since the microphone is completely muted when not in use, but it is very "choppy" sounding in the local sound reinforcement system.

A gain proportional automatic mixer applies the most gain to the most active microphone with smooth, continuous changes. This makes it extremely effective for local sound reinforcement, but the continuous gain changes make it difficult for the echo canceller to remain converged and effectively reduce the echoes at the far end.

The DMTH4 in conjuction with a DM Series processor offers a unique approach to the problems with simultaneous teleconferencing and sound reinforcement. The patented adaptive gain proportional mixing algorithm works in conjunction with a centralized echo canceller to address a variety of issues. The automatic mixer provides seamless allocation of gain to local microphones through a mix-minus matrix to reduce background noise and decouple loudspeaker and microphones, while a very fast converging DSP echo canceller operates on the composite transmitted signal being sent to the far end. This combination of processes is possible only with the latest DSP technology.

The auto mixing algorithm adapts to changes in background noise continuously, and unlike a gated mixer there are no threshold levels to adjust. A sum of all channels is the reference signal, each channel level is compared to this reference and the individual channel gain is adjusted to apply NOM attenuation. Gain is adjusted continuously to eliminate audible artifacts that gating and abrupt level changes can cause. As the common mode noise in the room changes, all channels are affected equally. The end result is seamless, adaptive auto mixing that requires no calibration or threshold adjustments.

Each individual output of the matrix operates as a separate NOM bus, so a particular input can be assigned to multiple outputs with mix parameters adjusted differently for each output. In other words, gain and mix mode are configured independently for each matrix crosspoint, resulting in great flexibility. Four mix modes are supported: Auto, Direct, Override and Background.

The echo canceller converges continuously when the level of the far side signal exceeds a minimum level, and the ratio of the far side signal to local room sound exceeds a minimum ratio. This dynamic control prevents divergence during periods of silence from the far side room or in "doubletalk" situations. The convergence takes place very quickly to keep up with the changes made by the automatic mixing algorithm and other changes that occur in the room. Setup is greatly simplified and any adjustments, such as level changes made with a remote control system, are accommodated automatically.

The convergence speed is adjustable in the control panel GUI to fine tune it to a particular situation. Faster convergence times can track changes in the room almost instantaneously, but the depth of echo cancellation will be reduced. Slower convergence times take a bit longer to fully converge, but produce greater echo cancellation. The ERLE value achieved by the echo canceller is displayed on the GUI and the effects of altering the convergence rate will be immediately visible and audible.

An important final note on the DMTH4 is the fact that the echo canceller will never "diverge" (lose convergence). This unique algorithm will also converge on a continuous sine wave, which is especially important when DTMF tones are present in the room. Since the echo canceller will never diverge, there is no need for a "panic button" (as is used in other designs) to generate a noise burst to help the echo canceller re-converge.

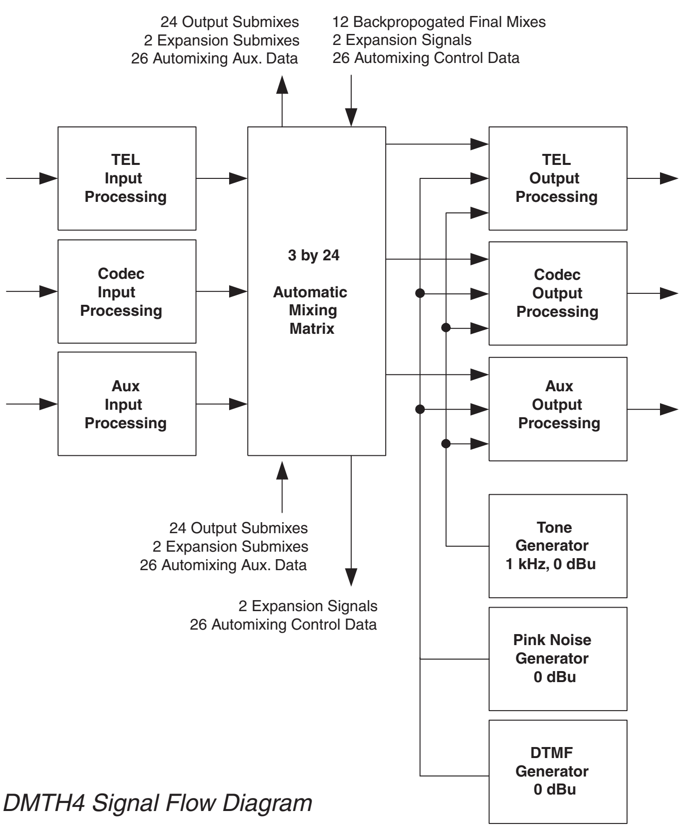
General Overview

The DMTH4 integrates telephone lines, video codecs and external audio sources into the digital bus structure of DM Series processors, allowing these audio signal sources to operate as though they are another microphone or audio input in the local sound system. This is a complete DM Series digital matrix processor, with a 3-in/24-out digital matrix, automatic mixing and comprehensive signal processing on every input and output.

The latest generation DSP microchips and microprocessors are the core of the engineering of the DM Series and the DMTH4 is no different. The focus and purpose is to meet the requirements of modern applications and also the demands for convenience and automation.

The DMTH4 is configured through the DMTH4 Control Panel which is part of LecNet2™, a user-friendly, yet powerful control program. The Control Panel offers quick configuration and full command of the system through either a USB or RS-232 compatible interface. Once configured, it operates independently.

All models in the DM Series offer the same signal processing functions, and vary only by the number of audio inputs and outputs available. The DM's basic structure consists of three stages: Input, Matrix and Output. (See *DMTH4 Signal Flow Block Diagram* and *DMTH4 Functional Block Diagram*.)



DMTH4 Signal Flow Diagram

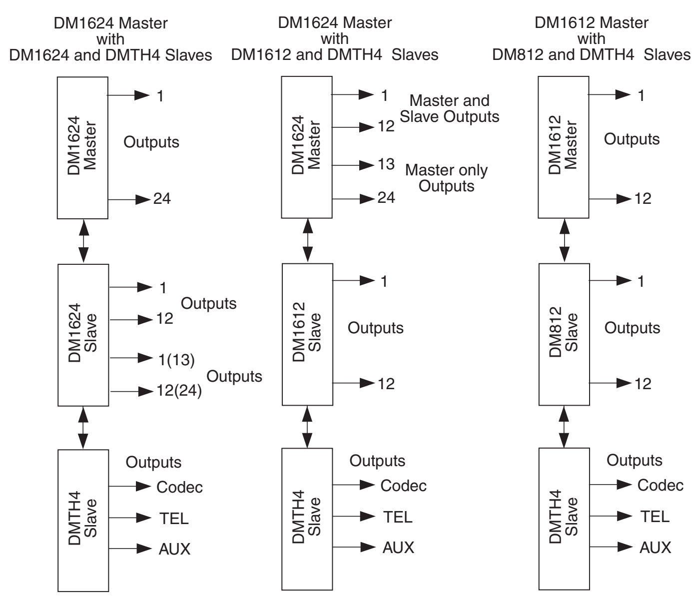
Each input channel includes a high quality 24-bit A-D converter. Extensive digital signal processing is provided on each input channel. Each input channel is processed and filtered as needed and the signal is delivered into the matrix.

The digital matrix mixer distributes each input signal to any selected combination of mix busses, with level control at each crosspoint. The matrix processes the signals and communicates them to other devices in the system. Each output receives signals from the mixing matrix, the pink noise generator or the tone generator as needed for setup, diagnostics or operation. Each of the 3 outputs includes extensive signal processing to optimize the mixed signal for the intended purpose, such as sound reinforcement, recording or teleconferencing.

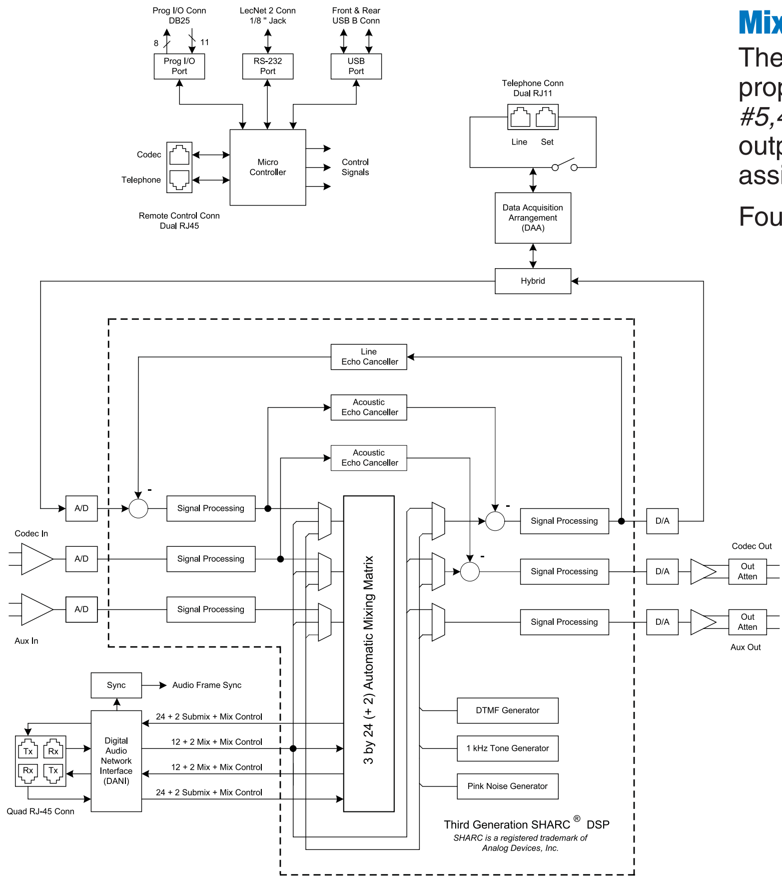
The DMTH4 is integrated into a system of DM Series automixers using the Digital Audio Network Interface (DANI). DANI connects the digital audio outputs of the units through standard RJ45 connectors.

When DM Series automixers are stacked, mixing data and the digital audio are passed between the slave units and the master unit through the DANI. Multiple units can be stacked in a master/slave configuration to expand the number of inputs to hundreds of channels. The DMTH4 is designed to be the end slave unit of a DM Series stack.

The audio and data from all units in the stack is gathered in the matrix in the master unit, which is where the final mix signals are generated. The first 12 final mix signals from the master are back propagated through the DANI to each slave. (See *DM Series Back Propagation Diagram*.)



DM Series Back Propogation Diagram



DMTH4 Functional Block Diagram

Mixing Mode

The automatic mixing algorithm applies a patented gain proportional algorithm (*US Patents #5,414,776* and *#5,402,500*) allowing each input assigned to a particular output to behave differently relative to the other inputs assigned to the output.

Four different mixing modes are available:

Auto - In automatic mode the input applied to the crosspoint is mixed into the output channel using the the Adaptive Proportional Gain automixing algorithm in the normal manner. This is the most common setting.

Direct - In Direct mode the automixing algorithm is bypassed.

Override - Override mode is selected when it is required that the input applied to the crosspoint **always** dominates the output channel when it becomes active.

Background - Background mode is selected when it is required that the input applied to the crosspoint dominates the output channel **only** when all other inputs are inactive.

Digital Matrix

The digital matrix provides signal routing and communication with other devices in the system, and applies automatic mixing and level control. (See *Digital Matrix Functional Block Diagram*.)

Automixer Cell

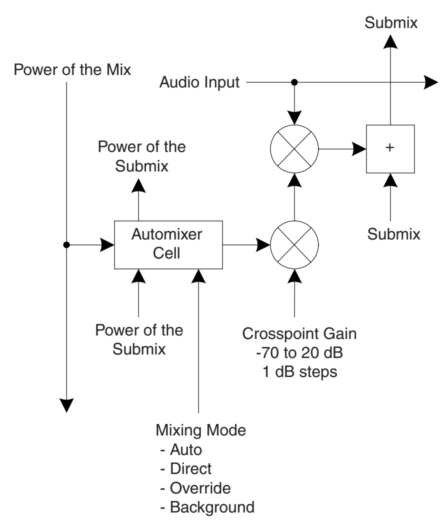
The Automixer Cell is the core of the matrix. It is where level control for the automatic mixing algorithm, mixing mode and crosspoint gain is applied to data gathered from other channels and devices. The cell receives data from the master unit in a multiple unit stacked configuration and from the slave units farther down in the chain.

Power of the Mix

The Power of the Mix is the reference used to determine the gain to be applied to each individual output channel. In a multi-unit stacked configuration, this data is sent to the slaves from the master unit.

Crosspoint Gain

Crosspoint Gain is the gain selected with the control panel that determines the level at the output.



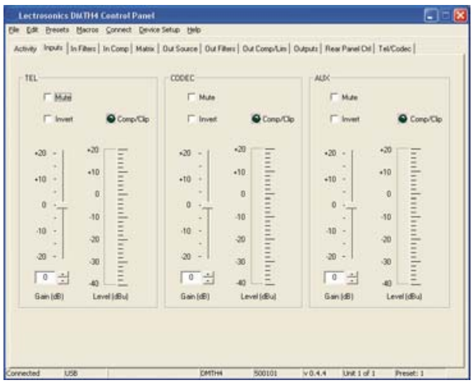
One of 72 Matrix Crosspoints

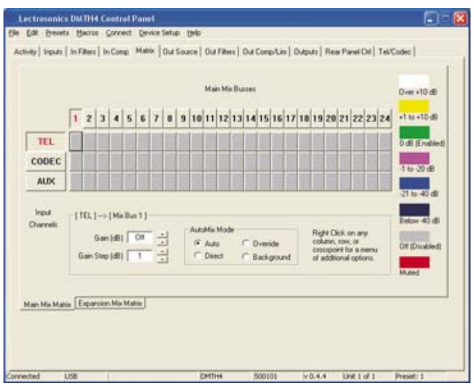
Digital Matrix Functional Block Diagram
(Typical Matrix Crosspoint)

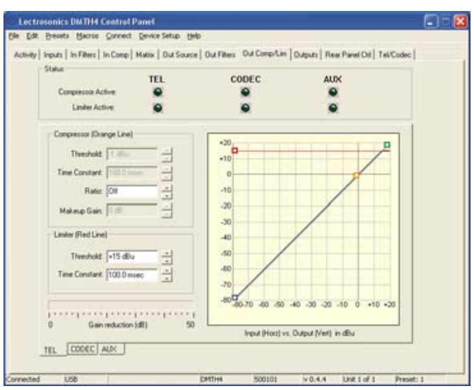
LecNet2 Software

Software is included with the DMTH4 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 DMTH4 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.







**Windows is a registered trademark of Microsoft Corp.*

Input Processing

Each input channel provides individual stages for gain, filtering and compression.

Input Gain

The input applies software controllable gain with a level indicator and clipping indicator.

Filters

Up to six filters can be implemented at each input to idealize the signal equalization.

The filter types include:
- Low pass
- High pass
- Band pass
- Parametric EQ
- Low shelving
- High shelving

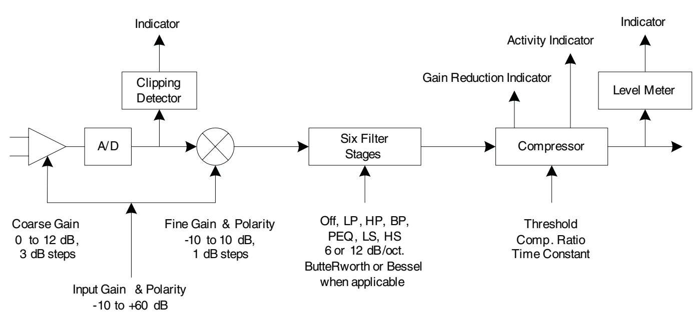
Filter slopes can be selected with 6 or 12 dB per octave Butterworth or Bessel parameters. Multiple filters can be assigned to create steeper slopes in 6 dB steps.

Input Compressor

The compressor implementation is a unique “soft knee” type based on an RMS level detector controlled by a single time constant parameter. This is a new design which responds to varying rates of change in the signal level by dynamically adjusting the attack and release times for best performance. Adjustment is simplified by entering a single value (half of the desired release time). The attack time is then applied by the DSP to vary with the signal.

The default value is 100 ms, which sets the release time at about 200 ms. The attack time is signal controlled and varies from about 2 ms to about 100 ms as is needed to handle the signal dynamics. See the reference manual for a closer look at this unique and very effective compressor.

Compressor adjustment parameters include:
- Threshold
- Time Constant
- Compression ratio
- Makeup gain



One of 3 Input Signal Processing Blocks

Typical Input Signal Processing Blocks

Output Processing

Output Source Select

The TEL, CODEC and AUX outputs can each be set to receive a signal from the pink noise generator, the tone generator, the expansion outputs or from the master unit outputs. The **pink noise** source can be used for sound masking during operation, and for equalization during setup. The **tone generator** is used for level adjustments and signal routing diagnostics. The **DTMF generator** is used to create the DTMF tones for initiating calls.

In normal operation the **digital matrix** delivers the audio signals to the outputs, which consist of the final mixes backpropagated from the master unit in the system via the Digital Audio Network Interface (DANI), with 12 mixes from the main matrix and 2 mixes from the expansion matrix.

CODEC and AUX Output Channels

These outputs include an attenuator to reduce the output level from line to mic level. The passive attenuator does not change the signal to noise ratio of the signal, but simply applies user selectable 20 dB or 40 dB of attenuation to reduce the signal level.

Output Signal Processing Stages

Each output channel provides six filters plus a compressor and limiter to idealize the channel for its function in the sound system. (See *Typical Output Signal Processing Block*)

Output Gain and Level Indicator

The output level can be adjusted from - 70 dBu to +20 dBu in 1 dB steps to perfectly match the requirements of the device being fed by the channel. A bar graph is provided by the on screen GUI to accurately indicate the output level as it operates and is adjusted.

Filters

Up to six filters can be implemented at each output to idealize the signal equalization.

The filter types include:
- Low pass
- High pass
- Band pass
- Parametric EQ
- Low shelving
- High shelving

Filter slopes can be selected with 6 or 12 dB per octave Butterworth or Bessel parameters. Multiple filters can be assigned to creater steeper slopes in 6 dB steps.

Output Compressor and Limiter

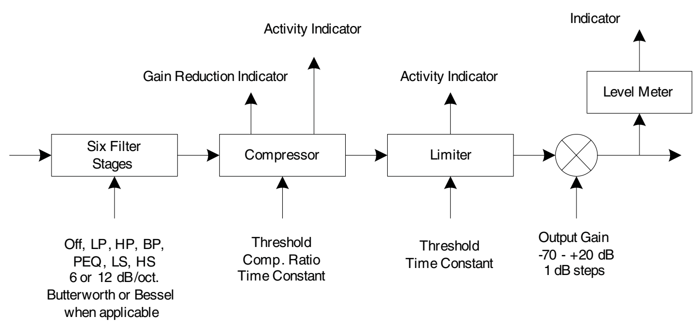
A versatile compressor and limiter are provided at each output to control the average level and dynamics of the audio signal, and restrict the maximum output level to optimize the channel for its purpose. Compression is often needed when the channel is feeding a recorder, and limiting is often used to protect a loudspeaker system and reduce distortion and amplifier overload.

The compressor implementation is a unique “soft knee” type based on an RMS level detector controlled by a single time constant parameter. This is a new design which responds to varying rates of change in the signal level by dynamically adjusting the attack and release times for best performance. Adjustment is simplified by entering a single value (half of the desired release time). The attack time is then applied by the DSP to vary with the signal.

The default value is 100 ms, which sets the release time at about 200 ms. The attack time is signal controlled and varies from about 2 ms to about 100 ms as is needed to handle the signal dynamics. See the reference manual for a closer look at this unique and very effective compressor.

Compressor adjustment parameters include:
- Threshold
- Time Constant
- Compression ratio
- Makeup gain

Limiter adjustment parameters include:
- Threshold
- Time Constant



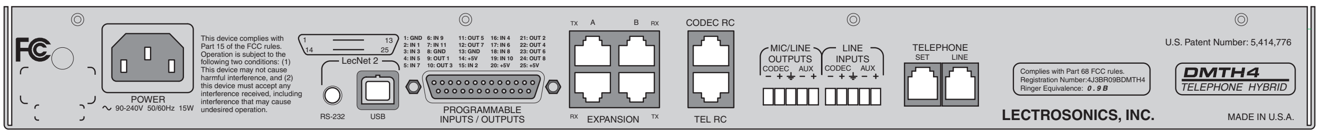
Typical Output Signal Processing Blocks

Front Panel

The DMTH4 is housed in a single space 19” rack mount assembly. The front panel provides a Mode switch to allow booting the unit as a Master when it is configured as a Slave and powered up by itself. The Status LED indicates steadily in normal operation and blinks in the presence of several different errors.

A USB port on the front panel allows easy access for setup or troubleshooting from the front side of the rack. The power switch is a rocker type with positive action.

Rear Panel



A universal 100-240 VAC universal power supply is included on the rear panel with a standard 3-pin receptacle. The USB and RS-232 jacks are used to connect to a computer for setup, or to control systems for operation. Logic input and output connections are made via a DB-25 jack. RJ-45 jacks interface with other DM Series components.

Codec and telephone wall plate and desktop remote control accessories connect through a dual RJ45 connector. Codec and auxiliary inputs and outputs are made via depluggable connectors. The telephone line and handset are connected through standard RJ-11 jacks.

Specifications

Echo Canceller (3 Total):	2 Acoustic - 126 mS tail time 1 Line - 30 mS tail time Will never diverge, regardless of signal type (i.e. sine wave)
Telephone Line Return Loss:	45 dB
Audio inputs (Codec, AUX):	
Gain:	-20 dB to +20 dB, programmable in 1 dB steps
Input impedance:	10 k Ohm
Connector:	5-pin Phoenix
Audio outputs (Codec, AUX):	Floating balanced, either side can be grounded
Nominal level:	0 dBu all outputs, -20 and -40 dBu selectable
Output impedance:	
0 dB Attenuation:	• 450 Ohms differential
-20 dB Attenuation:	• 50 Ohm differential
-40 dB Attenuation:	• 5 Ohm differential
Input Dynamic Range (Codec, AUX):	102 dB (unweighted 20 - 20 kHz)
Output Dynamic Range (Codec, AUX):	105 dB (unweighted 20 - 20 kHz)
Audio Performance (Codec, AUX):	
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” type
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:	Shielded CAT-5
Transmission speed:	50 Mbits/s
Programmable control inputs	
Number of inputs:	11
Analog voltage range:	0-5V
Logic input:	TTL, LVTTL, CMOS, LVCMOS
Programmable control outputs	
Number of logic outputs:	8
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:	15 Watts (no ventilation requirements - no fan)
Dimensions:	
Faceplate:	Standard 19 inch 1RU
Housing:	17.500” W x 1.710” H x 7.500” D
Weight:	1595 grams; 3.516 lbs. without AC cord

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