ASPEN

TECHNICAL DATA

Mixer and Input Only Models

- **Ultra low noise input preamps**
- **Full crosspoint matrix with 48 outputs**
- **Unlimited input expansion**
- **TCP/IP Ethernet addressable**
- **Adaptive Gain Proportional Automatic Mixing at the matrix crosspoints**
- **Simultaneous multi-point 3rd party and native control**
- **48 final mixes, ultra-low latency**
- **Automatic Master/Slave detection**
- **Single CAT6 interconnection carries data, audio and control signals**

The variety of mixers available are created by combining “building block” printed circuit board assemblies:

- 8 input, 12 output mixer board
- 16 channel input only board
- 8 channel input only board

Multiple chassis of any 1RU or 2RU model can be stacked to expand the number of inputs and outputs needed for the system design. The ASPEN digital matrix provides a maximum of 48 total outputs, but there is no limit to the number of inputs that can be added to a system by stacking multiple units.

Mixer models include:

- SPN812 8 input, 12 output mixer, 1 RU
- SPN1612 16 input, 12 output mixer, 2 RU
- SPN1624 16 input, 24 output mixer, 2 RU
- SPN2412 24 input, 12 output mixer, 2 RU

Input only models include:

- SPN16i 16 channel , 1 RU
- SPN32i 32 channels, 2 RU

Input only units deliver outputs to the digital bus, so they are always used with a host mixer or conference unit to provide physical audio outputs.

When multiple units are stacked, Master and Slave units are automatically detected and configured. All data and audio from the Slave units in the system is gathered in the Master, so a single connection between a computer and the Master allows access to all units in the stack. The throughput latency of all audio inputs in a stack is automatically synchronized to maintain absolute signal phase at the audio outputs.

All models fully support the 48 outputs provided by the digital matrix, regardless of how many physical outputs are present on the rear panel. Any physical output can deliver the signal from any output in the matrix.

Extensive, simultaneous control can be applied through ethernet, USB, RS-232 and logic I/O ports using the large command library. Combined with the comprehensive macro library, control options are available to meet a very wide range of requirements and user interfaces.

The ASPEN control protocol has several key features:

- It is a request-response message protocol
- An error reporting mechanism is provided
- The message syntax is text based
- Messages may carry data payloads

For more information, browse the Help files in the downloadable software from the web site.

Rio Rancho, NM, USA
www.lectrosonics.com

Signal Flow

Every input can be used with microphone or line level signals with gain adjustable from -10 to +60 dB. Following the analog to digital converter (ADC) processing stages are arranged in logical order. After processing, the signal can be assigned to any one or more crosspoints in the matrix for mixing and forward propagation to the Master unit.

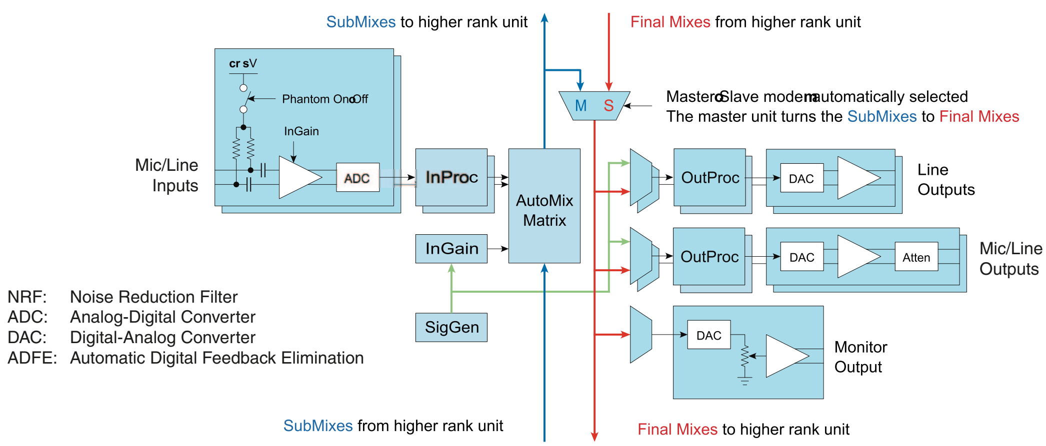
Audio signals and data in each processor are added to the audio signals and data from the processor below it and propagated to the next processor above it (forward propagation) through the 1 Gbps bus. The final total of all signals and data in a stack are gathered in the Master and then propagated back to the Slave units (backward propagation). In this manner, the signal source for each output on every Slave in the stack can be taken from any of the 48 crosspoints in the matrix.

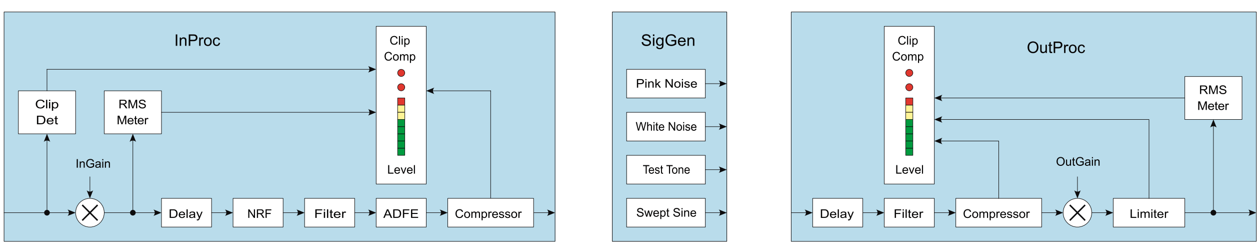
Each output has a signal processing chain, gain control and limiter ahead of the digital to analog converter.

Automatic mixing takes place at the matrix crosspoints with four different characters available:

- Automatic - normal auto mix activity
- Direct - on at all times; for recording
- Override - dominant in the mixing activity
- Background - subordinate in the mixing activity
- Phantom - for multi-zone mix-minus systems

This unique functionality allows each input to behave differently at each output. For example, one channel with a microphone connected can provide a direct signal for recording at some output channels, participate on an equal basis with other output channels in a sound reinforcement system and act as the “chairman” microphone in groups delivered to other output channels.





This diagram depicts the signal flow through a mixer model with physical inputs and outputs. ***Input Only*** models process and route input signals into the matrix, but do not have physical outputs, so they are always used with a mixer or conference model processor.

The ASPEN architecture allows all signal processing to be fully enabled at all times without limitations of DSP resources. With the computer connected to the Master unit in the system, changes in the setup, filters, etc. take effect and are immediately audible.

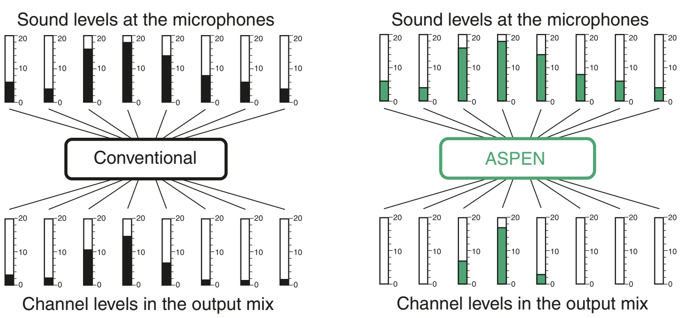
Automatic Master/Slave Detection

Each processor board connects to the other processors through the upper and lower RJ-45 jacks. The presence of a connector determines the ranking of the processor in the stack. If there is no connector in the upper jack, the processor is positioned at the top of the stack and is automatically configured as the Master in the system. If a connector is present in the upper jack, the processor is automatically configured as a Slave in the system.

Adaptive Proportional Gain Automatic MIxing Algorithm*

An automatic mixer that uses a ***Gating*** process turns channels off and on abruptly as the level passes an established threshold. While this can be acceptable in a small sound system, it usually produces “choppy” sound when used with more than a few microphones. A more effective approach is ***Proportional Gain*** mixing where each channel level is compared to the sum of all channels and more gain is given to the louder channels. In this approach, channel levels are adjusted in a seamless manner to eliminate the abruptness of a gate.

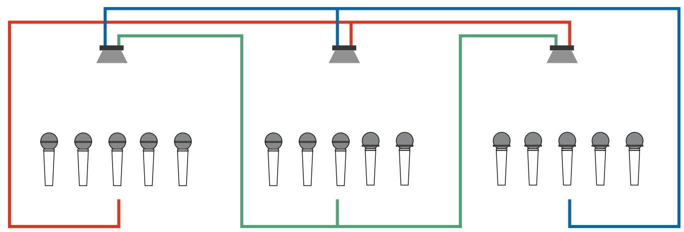
The proprietary algorithm used in ASPEN processors adds a significant improvement to conventional proportional gain auto mixing. The gain allocation in a conventional algorithm is in direct proportion to the activity at the microphones, so gain is reduced on channels with lower levels, but those channels still inject sound and noise into the final output mix.



The patented algorithm used in ASPEN processors applies a subtle priority to the channel that has been the loudest for the longest period of time. This increases the gain on the dominant channel and decreases the gain on subordinate channels to a greater degree than a conventional proportional gain mixer. Only one channel is dominant, which further reduces background noise, suppresses recirculating sound in the room and prevents comb filtering when a single voice arrives at two microphones at close to the same level.

Mix-Minus Sound Reinforcement

Mix-minus routing through the matrix establishes loudspeaker coverage ***Zones*** that function as sub-systems within the room, each using a dedicated Final Mix to provide the signal routing. This design approach excludes nearby microphones from the mix sent to each loudspeaker group to improve feedback stability, noise suppression and the echo return loss in audio conference connections. In this example, the audio from microphones in the red zone are routed to the green and blue zones, but not back to the red zone.

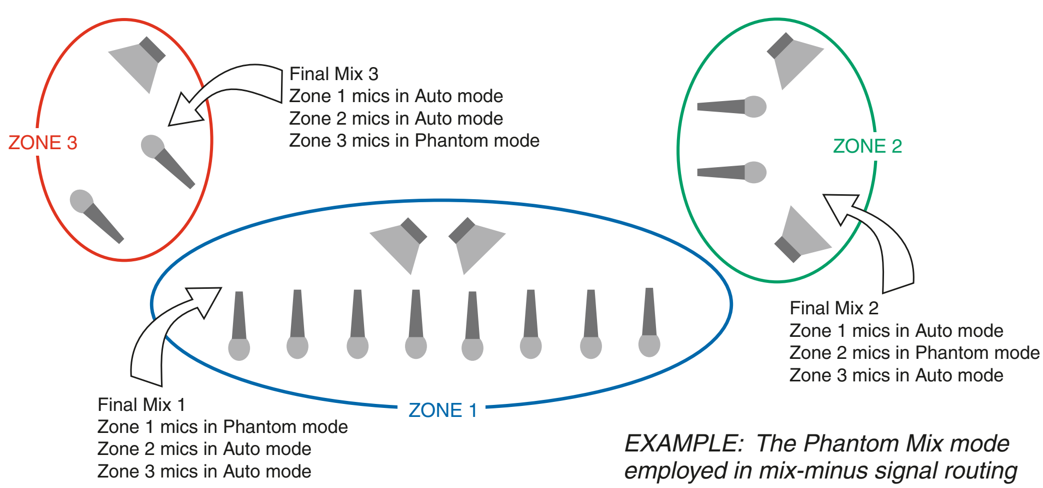


As a finishing touch on a mix-minus design, a unique mixing mode called ***Phantom Mix*** can also be employed.

The Phantom Mix Mode

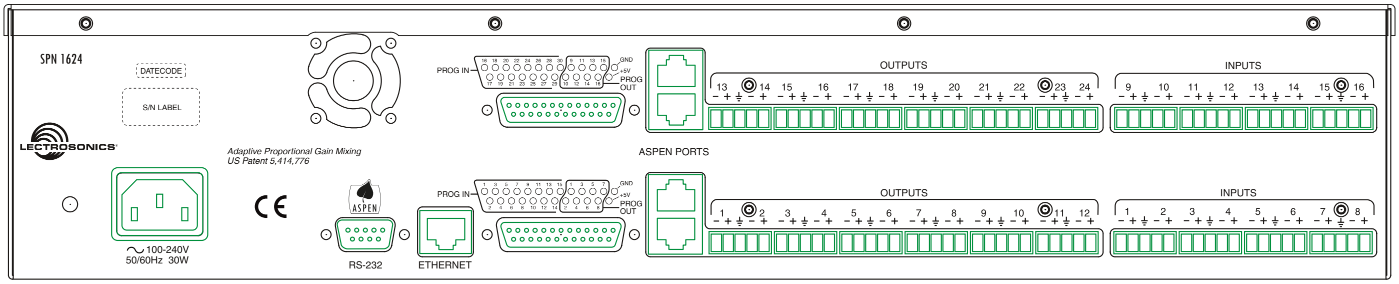
In a meeting space, sound from loudspeakers is not contained strictly within the defined acoustic zones. In addition, all of the microphones are in the same overall acoustic space. Even though a mix-minus routing has been established, sound from a loudspeaker bleeds into adjacent zones, is picked by those microphones, and it is routed back to the loudspeaker where the sound originated. This recirculation reduces ERL and intelligibility, and can even cause feedback.

The ***Phantom Mix*** mode allows multiple zones to participate in an overall room mixing activity, but delivers the audio signals only to the desired acoustic zones defined in the mix-minus setup. Microphone channels that are excluded in the routing are set to the phantom mode to accomplish this finishing touch.

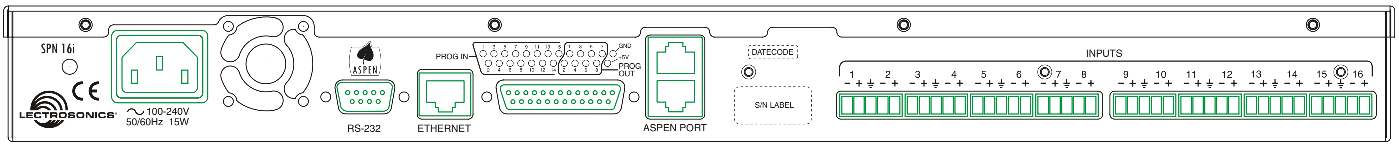


EXAMPLE: The Phantom Mix mode employed in mix-minus signal routing

**US Patents 5,414,776 and 5,402,500*



The model SPN1624 houses two 8x12 mixer boards in a 2RU housing.



The model SPN16i houses a single input only board

Specifications

Audio inputs

All inputs are digitally programmable-gain microphone to line level differential inputs. Either side can be grounded or left floating. The cable shield shall be connected to ground.

Max. input level:	20 dBu
Gain:	0 dB to 56 dB, programmable in 8 dB steps (the analog gain is automatically selected by selecting the input gain)
Input impedance:	8 kΩ differential mode, 2 kΩ common mode
Phantom voltage:	48 V
Dynamic range:	102 dB
EIN:	-127 dBu (20Hz – 20kHz, unweighted)
THD + noise:	0.01%

Audio outputs

All outputs are floating transformerless differential outputs. Either side can be grounded or left floating. The cable shield shall be connected to ground.

Nominal level:	0 dBu, channels 1-8 0 dBu, -20 dBu, -40 dBu, channels 9-12
Headroom:	20 dB
Output impedance:	< 50 Ω, all outputs, at all attenuator settings
Dynamic range:	105 dB
THD + noise;	0.01%

Latency

Single-board:	64 audio samples = 1.333 ms
System:	64 + 6 * (total number or boards – 1) audio samples = 1.333 + 0.125 * (total number or boards – 1) ms

Monitor output (1/4" headphone jack)

Signal:	any of the 48 final mixes
Output power:	50 mW (<50 ohm impedance recommended)

Filters

All filters, including the noise reduction filter (NRF), have zero processing delay.

Noise reduction filters:	Adjustable 6 to 35 dB on every input
Tone control stages:	4 per input channel
Parametric EQ stages:	8 per output channel
ADFE:	8 per input channel Configurable as Static or Dynamic

Filter types

Low Pass:	Butterworth (6, 12, 18, 24 dB/octave) Bessel (6, 12, 18, 24 dB/octave) Linkwitz-Riley (12, 24 dB/octave) Additional parameters: frequency [Hz]
High Pass:	Butterworth (6, 12, 18, 24 dB/octave) Bessel (6, 12, 18, 24 dB/octave) Linkwitz-Riley (12, 24 dB/octave) Additional parameters: frequency [Hz]
Low Shelving	Butterworth (6, 12, 18, 24 dB/octave) Bessel (6, 12, 18, 24 dB/octave) Additional parameters: frequency [Hz] boost/cut [dB]
High Shelving	Butterworth (6, 12, 18, 24 dB/octave) Bessel (6, 12, 18, 24 dB/octave) Additional parameters: frequency [Hz] boost/cut [dB]
Peaking EQ (parametric)	Parameters: frequency [Hz] bandwidth [octave] boost/cut [dB]

Internal Signal Generator:

Swept sine:	Modes:	single sweep, continuous sweep
	Waveforms:	sawtooth (up or down), triangle
	Sweep rate:	linear, logarithmic
	Parameters:	start freq, stop freq, level [dBu], sweep time [sec]
White noise:	Parameter:	level [dBu]
Pink noise:	Parameter:	level [dBu]
Tone (sine wave):	Parameters:	level [dBu], frequency

Power Requirements: 100-240 VAC, 50/60 Hz

Power Consumption:

1RU models:	15 Watts
2RU models:	30 Watts

Dimensions:

1RU models:	1.75 x 19.00 x 7.70 inches
2RU models:	3.50 x 19.00 x 7.70 inches

Weight:

1RU models:	3.64 lbs., 1651 grams
2RU models:	5.73 lbs., 2600 grams

LECTROSONICS®
581 Laser Road NE • Rio Rancho, NM 87124 USA • www.lectrosonics.com
(505) 892-4501 • (800) 821-1121 • fax (505) 892-6243 • sales@lectrosonics.com

11 June 2013