ASPEN

TECHNOLOGY OVERVIEW

Architecture and Signal Flow

- Optimized Architecture™
- Advanced Echo Cancellation
- Third Octave Noise Reduction
- Full crosspoint matrix with 48 outputs and unlimited input expansion
- TCP/IP Ethernet addressable
- Simultaneous multi-point 3rd party and native control
- Seamless auto-mixing with PGA™ at the matrix crosspoints
- Ultra-low latency
- Automatic Master/Slave detection
- Single CAT6 interconnection carries data, audio and control signals between units

ASPEN

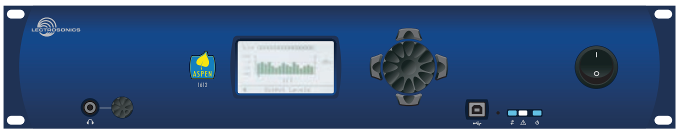
LECTROSONICS®
Rio Rancho, NM, USA
www.lectrosonics.com

Hardware Architecture

The variety of models in this series are created by combining “building block” circuit board assemblies:

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

A single board can be enclosed by itself in a stand-alone 1RU chassis, or combined with another board in a 2RU chassis to create a variety of models. The 2RU models include an LCD with comprehensive access to all system settings and activity.



Mixer and input only units include the following models:

- SPN812 — 8 input, 12 output mixer
- SPN1612 — 16 input, 12 output mixer
- SPN1624 — 16 input, 24 output mixer
- SPN2412 — 24 input, 12 output mixer
- SPN16i — 16 channel input only
- SPN32i — 32 channel input only
- SPNConference — Conference interface
- SPNTrio — 8 input, 12 output mixer with Conference interface

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

The SPNConference model is used with a mixer to provide mic/line audio inputs and outputs.

When multiple units are stacked, the Master unit will automatically be detected and configured and the other units will be configured as Slaves.

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 software access to all units in the stack.

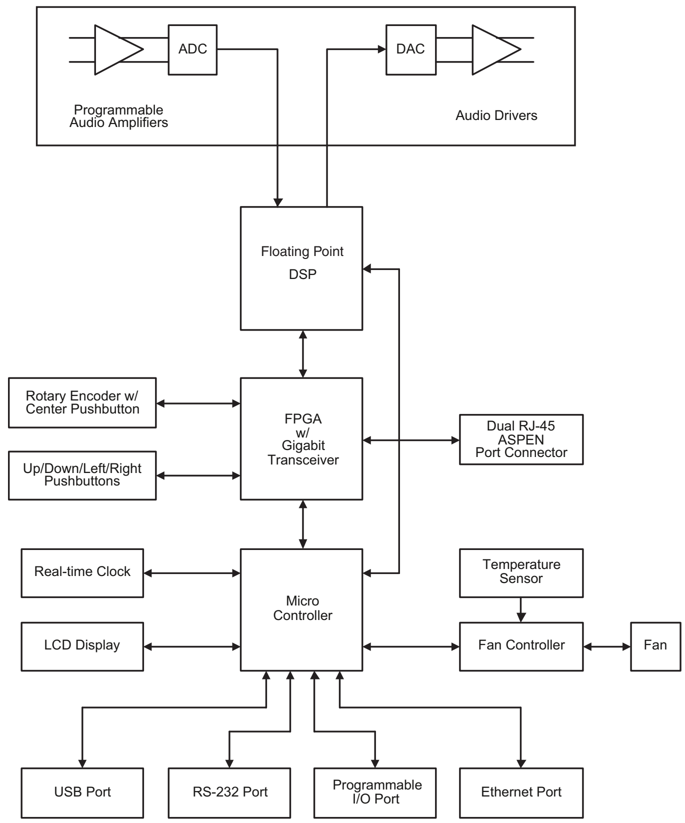
The slight throughput delay of inputs from slaves and the master in the ASPEN bus is automatically synchronized to maintain absolute signal phase at all outputs.

At the core of each ASPEN board is a powerful communications and control structure.

A latest generation SHARC® processor* performs the millions of calculations required to implement signal processing, auto mixing, echo cancellation and noise reduction.

An FPGA with a gigabit transceiver interacts with the front panel controls and coordinates the data flow in and out of the ASPEN bus.

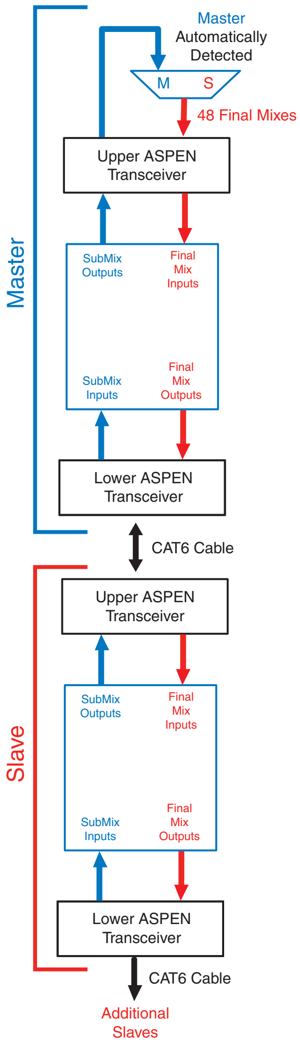
The microcontroller interfaces with the I/O ports, the front panel LCD, the real time clock and oversees the temperature regulation.



*SHARC is a registered trademark of Analog Devices, Inc.

Signal Propagation in the ASPEN Bus

The ASPEN bus provides a 1 Gbps throughput carrying audio and data with a single CAT6 connection between the units in a system.



When multiple units are stacked, Master and Slave units are automatically detected and configured for the correct signal flow.

Audio and data signals propagate through *submixes*. The lowermost slave in the system generates a submix of signals from devices connected to it and passes the submix to the next slave above it.

Each intermediate slave unit adds to the submix from the unit below it, updates the submix and passes it on to the unit above it. The process continues through all slave units in the system with no limitation on the total number of slave units that can be used.

The Master unit gathers the submix from the slave below it, updates it with its own signals and generates the final mix. The final mix is then back propagated to all slave units below it to enable system wide auto mixing and control.

The audio output of all units in the system is taken from the 48 channel final mix.

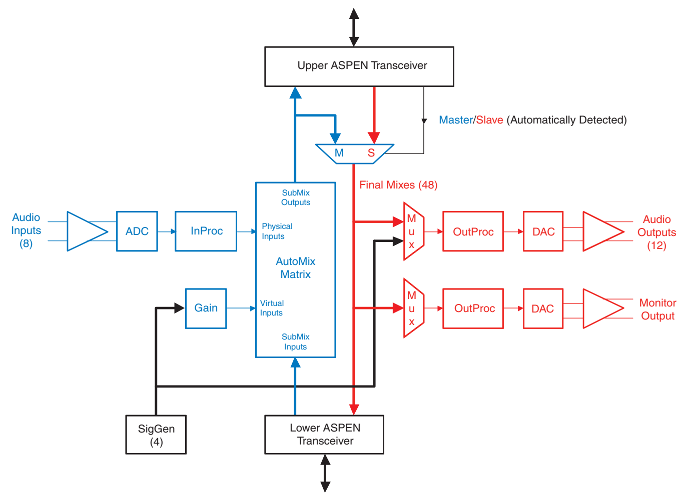
This unique architecture allows a single computer or network connection to the Master to have access to all units in the stack.

Scalability in the ASPEN Matrix

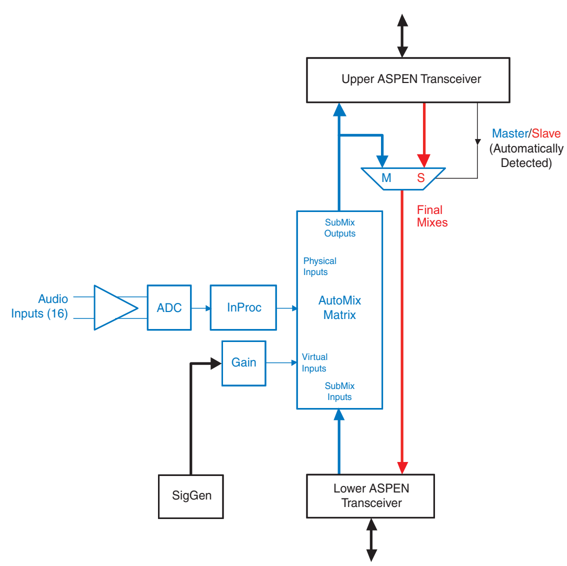
Each 8x12 mixer board provides 8 physical inputs, 12 physical outputs and access to the audio and data from the 48 final mixes in the ASPEN bus. Four *virtual inputs* are provided in any system configuration to feed signals from a built-in signal generator for setup, diagnostics and sound masking.

As multiple units are stacked, the size of the matrix grows accordingly. For example, a 16 input 2RU mixer model actually has 20 inputs when the virtual inputs are included. The matrix then consists of 960 fully functional crosspoints (20x48). As more inputs are added, the size of the matrix continues to increase without limitation.

Even with hundreds of inputs, every input can feed any one or all of the 48 outputs, with full signal processing available on every input and output.



ASPEN 8X12 Mixer Signal Flow



ASPEN Input Only Signal Flow

Optimized Architecture™

The ideal structure for signal flow and functional blocks through a system wide matrix is a direct path from inputs to crosspoints to outputs, with no extra paths or taps necessary to add signal processing. It must offer a full capability of routing every audio input to any one or all audio outputs without limitation. Every audio input should have its own dedicated signal processing blocks present at all times. Every audio output on any unit in the system should have full access to any crosspoint in the matrix, and have its own dedicated signal processing present at all times. This ideal structure is fully realized in the Optimized Architecture™ of ASPEN.

All available signal processing is enabled on every input and output with no resource meter or "gas gauge." Signal processing blocks are configured in the optimal sequence needed to ensure the highest signal to noise ratio and lowest distortion. This architecture eliminates the need to manually construct a drawing and connect one processing block to the next one in the chain. Simply enable a crosspoint and the connections are made.

Setup is straightforward and simple in spite of the immense amount of processing available. Settings are applied in real time as the system is operating without the need to compile and download files to the hardware. Once the setup is complete, it is saved to a preset in the hardware and to a disk file for backup.

Input Processing

In addition to the delay, filters and compressor, there are two special purpose processing blocks:

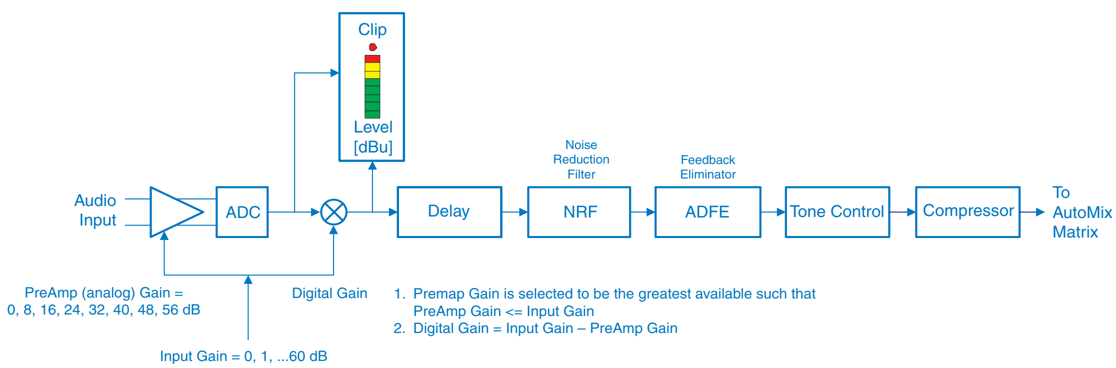
- NRF (noise reduction filter)
- ADFE (auto digital feedback eliminator)

NRF employs a proprietary noise reduction algorithm on every input channel using a 1/3 octave analysis and downward expansion. The amount of noise reduction applied to the signal at each input is adjustable from 6 dB to 35 dB as needed for the signal conditions and to satisfy individual preferences. The process is very effective, with almost no audible artifacts at 18 dB or more. Higher values are available for extremely poor conditions where noise is extremely high and intelligibility is preferred at the expense of subtle artifacts in the audio.

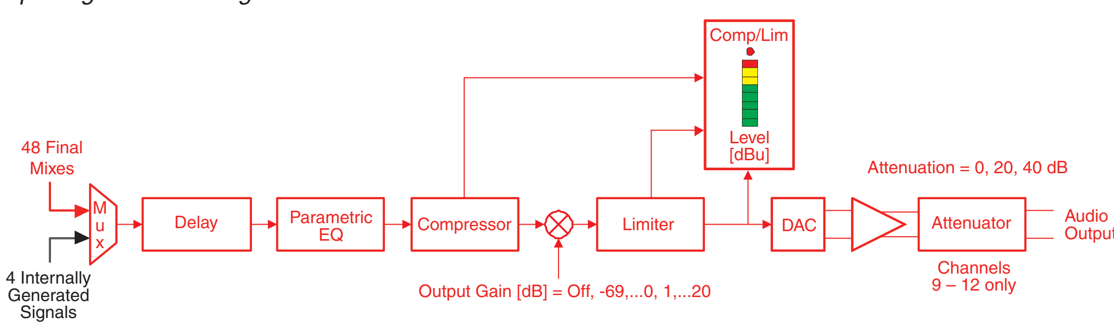
ADFE (auto digital feedback eliminator) is a notch filtering process with static or dynamic behavior as defined in the setup. Fixed notch filters can be configured as needed for appropriate applications, and dynamic notch filters can be defined to deal with changing conditions.

Output Processing

Each output channel can take its signal from the matrix or from an internal signal generator. The generator can deliver a variety of signal types for setup, diagnostics and sound masking. The processing blocks on every output are arranged in the optimal sequence used to feed power amplifiers and recorders.



Input Signal Processing Blocks



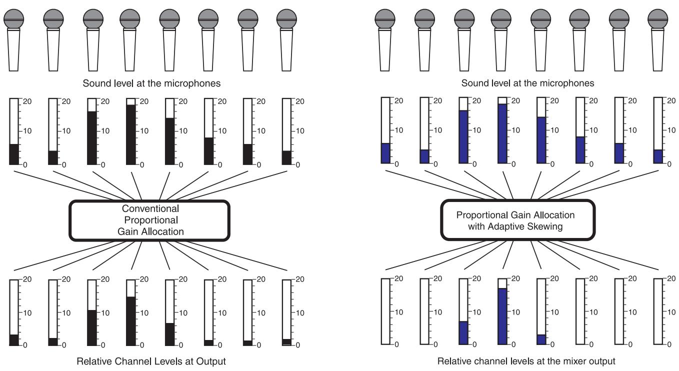
Output Signal Processing Blocks

Seamless Automatic Mixing

Lectrosonics pioneered adaptive proportional gain automatic mixing algorithms. The proprietary algorithm* employed in ASPEN is a seamless process that eliminates abrupt switching (gating), controls acoustic feedback and suppresses background noise.

All active input channels are summed, and then the level of each channel is compared to the total sum. Channels are then attenuated so that the resulting sum is equal to one channel at full level (NOM=1) with the loudest channel still the loudest in the mix. The algorithm operates in the same manner that a human operator would in mixing a conference manually on a console. Unused mics are attenuated and those in use are emphasized.

This auto mixing algorithm, working in conjunction with the AEC in the *ASPEN Conference* processor, provides impressive echo cancellation and noise reduction.



The algorithm includes a unique *automatic skewing* process that applies a subtle priority to the channel that has been the loudest for the longest period of time. The skewing further reduces inactive channels and prevents comb filtering by never allowing two channels to be mixed at the same level.

The auto mixing takes place at the matrix crosspoints, which allows each input signal to exhibit a different behavior at every output in the system. For example, input channel 4 could be configured for *Auto* behavior (normal auto mixing) at output 6 for local sound reinforcement, *Direct* behavior (no attenuation) at output 10 for recording, and so on. There are five different behaviors available:

- Direct - no attenuation
- Auto - normal gain proportional auto mixing
- Phantom - special mode for mix-minus systems
- Override - dominant in auto mixing activity
- Background - subordinate in auto mixing activity

The *Phantom* mode allows the channel to participate in the auto mixing algorithm at any crosspoint, but not deliver the actual audio signal to the output. This is used to combine zones for room-wide auto mixing activity in a mix-minus reinforcement system. The auto mixing action is common to all zones, but the audio signal routing to the loudspeakers remains as is it configured in the setup.

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

Low Latency

The throughput latency of a single master board is only 1.33 ms, regardless of how much processing is being used. Each additional PCB adds only 0.125 ms.

200 inputs can be handled with only 4.33 ms latency (1.33 ms for the master PCB plus 24 additional PCBs at 0.125 ms each.

Other examples include:

- 264 inputs with only 5.33 ms total latency
- 328 inputs with only 6.33 ms total latency
- 456 inputs with only 8.33 ms total latency
- Unlimited maximum with 1 ms added for each additional 64 inputs (8 boards)

Latency is not affected by the amount of processing being used at any stage in the signal chain.

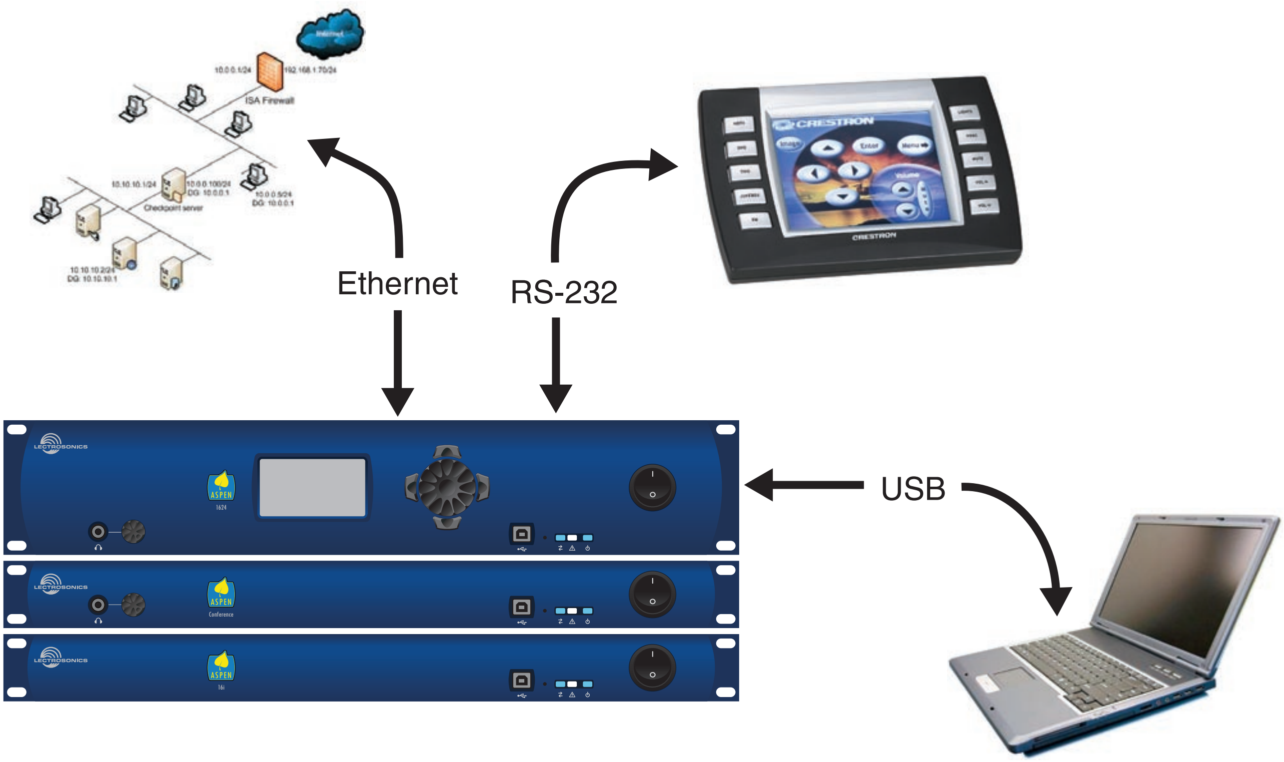
Every input is automatically synchronized to eliminate phase differences between the inputs included in the final mixes.

Single Point of Control

All ASPEN models support simultaneous use of Ethernet, RS232 and USB ports for setup, monitoring, diagnostics and control.

Installers and operators can use the software GUI to monitor the state of the processor via the USB port to verify that commands sent from the 3rd party controller (over RS232) are working correctly.

Remote monitoring and setup can be conducted via a network connection and from remote sites over the internet.



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