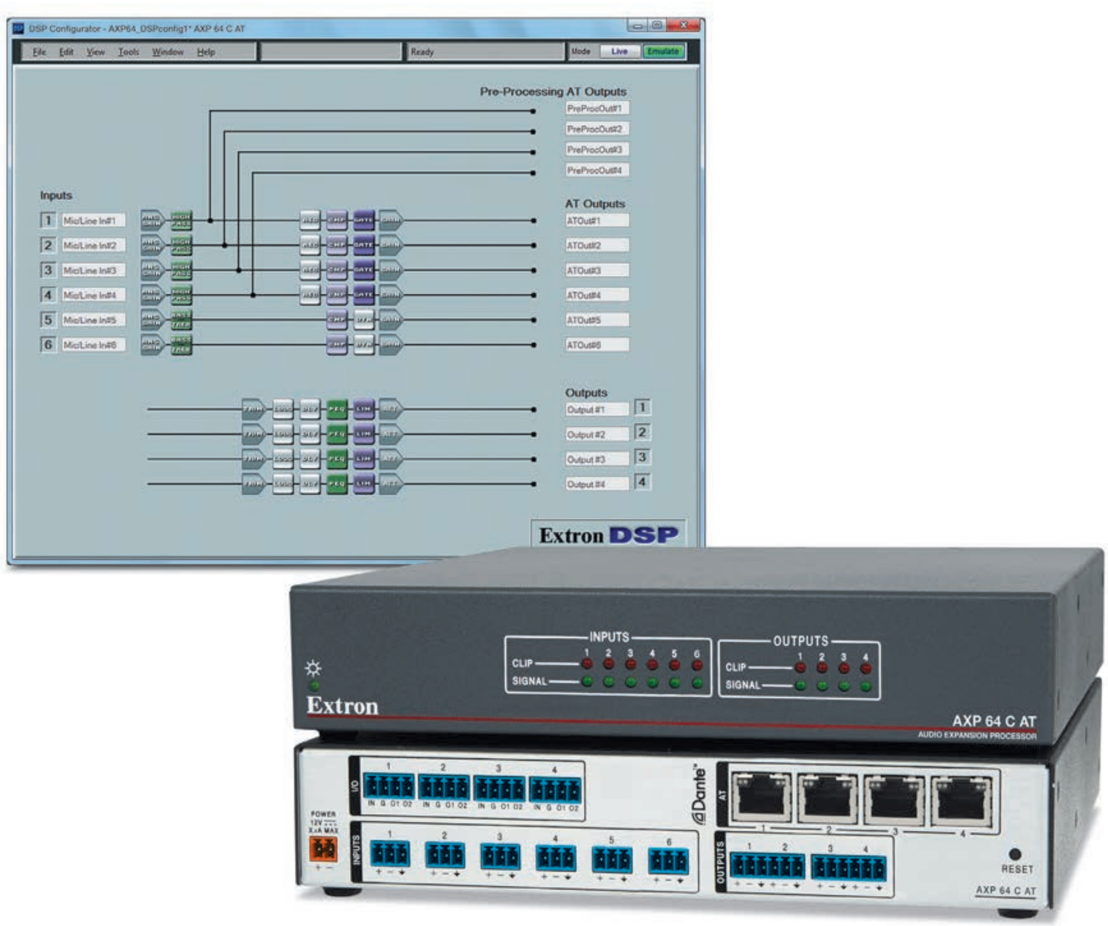
AXP 64 C AT

6 INPUT, 4 OUTPUT AUDIO EXPANSION PROCESSOR WITH AEC AND DANTE

ProDSP

Input and Output Channel Expansion with Processing for Extron DMP 128 Audio Systems

- Six mic/line level inputs, four with 48 volt phantom power
- Four independent channels of AEC - acoustic echo cancellation
- Four inputs with FlexInput capability to select between analog or Dante™ inputs
- Four Dante to line level outputs
- Built-in four-port Gigabit switch
- ProDSP™ audio signal processing:
  - 64-bit floating point digital signal processing engine
  - Fixed, low latency DSP processing



Extron Electronics
INTERFACING, SWITCHING AND CONTROL

Introduction

The Extron **AXP 64 C AT** is an audio expansion processor with six mic/line inputs and four line outputs for remote connectivity to a Dante-enabled Extron DMP 128 audio system. It is used to place six mic/line sources onto a Dante audio network, and route any four Dante channels from the network to an amplifier or other destination. The AXP 64 C AT features Extron ProDSP with gain, filtering, and dynamics processing for all inputs and outputs. Four of the inputs also include AEC, 48 volt phantom power, and dedicated control ports for mic control. FlexInputs offer the additional capability to process Dante channels in place of the first four local inputs. A single Ethernet cable from one AXP 64 C AT, or several linked units, to a central equipment rack greatly reduces the effort and expense of pulling one cable for each endpoint.

Remote Input and Output Expansion With Processing

The AXP 64 C AT accepts six channels of balanced or unbalanced analog audio, four with 48 volt phantom power. The AXP 64 C AT includes four independent channels of AEC with advanced algorithms for fast convergence and optimal intelligibility in conferencing applications. For remote processing of Dante audio network signals, the first four inputs can access any Dante channel on the network. The AXP 64 C AT is truly unique in that it incorporates DSP for each input channel. This makes it much simpler and less expensive to expand input capacity by avoiding the need to install a remote break-in box plus additional DSP hardware at the central rack, which may otherwise be underutilized.

The AXP 64 C AT can convert four Dante channels and route them with dedicated filtering, dynamics, and delay processing to the four line level outputs. This enables audio from the Dante network to be sent to a sound reinforcement system or other audio destination.

A sound system designer can incorporate several AXP 64 C AT or AXP 50 C AT audio expansion processors to create a large mixing matrix with up to 56 remote inputs and 24 outputs per DMP 128 AT. This greatly simplifies scalability and the audio cabling infrastructure. Utilizing the Dante network also avoids a long analog mic cable run for each mic channel and the possibility of introducing noise into the system. For further convenience, multiple AXP Series and DMP 128 AT units can be linked over their integrated four-port Gigabit switches.

FlexInput Channels

The four AEC-enabled inputs on the AXP 64 C AT offer FlexInput capability to route a Dante channel in place of a local mic/line input. The flexible input selection enables the AXP 64 C AT to process audio from remote wireless microphones, wallplates, and other sources from anywhere on the Dante network.

Acoustic Echo Cancellation at the Source

The AXP 64 C AT includes four independent channels of Extron AEC for conferencing applications. An AEC reference signal is provided over the Dante network by a DMP 128 C AT. AEC is essential for effective room-to-room conversations, ensuring clear, natural communication for all participants.

Flexible Signal Routing

The internal DSP architecture of the AXP 64 C AT sends both pre- and post-processing input feeds for the first four input channels onto the Dante network. The last two channels offer post-processing feeds to the Dante network. This provides the flexibility to limit the signal processing on mic input signals for recording, assistive listening, or voice reinforcement feeds, while applying full processing and AEC to a conferencing application. Filter blocks are situated prior to the pre-processing outputs, applying low frequency roll-off to attenuate breath and wind noise and other EQ settings, while bypassing the AEC. For conferencing applications, AEC processing, together with dynamics and EQ, are an absolute necessity to ensure optimal audio quality to the far end.

The AXP 64 C AT is housed in a compact 1U, half rack width enclosure that can be installed underneath a conference table or inside a credenza near microphones, sources, and remote audio destinations using the optional Extron UTS Series or MBU 123 mounting hardware.

Features

ProDSP

The **AXP 64 C AT** features Extron ProDSP, a powerful digital signal processing platform based on a 64-bit floating point DSP engine. ProDSP provides an extensive array of digital processing tools for audio system design, configuration, and optimization. The DSP Configurator™ Software is the user interface to ProDSP for full control and management of the AXP 64 C AT and all of its DSP functions, including level control, AEC, filters, and dynamics.

Highest Quality Converters Plus Floating Point DSP

The AXP 64 C AT features studio grade ADCs - analog-to-digital converters and DACs - digital-to-analog converters using professional quality 24-bit resolution and 48 kHz sampling, fully preserving the integrity of the original audio signal.

The processing power of the 64-bit floating point DSP engine allows for simultaneous audio processing algorithms within the same channel, and across multiple channels, without compromising sound quality. This powerful DSP engine also delivers very wide dynamic range to prevent clipping and fully maintain signal quality.

Fixed Yet Flexible DSP Architecture

The DSP Configurator Software features a fixed layout of DSP processing blocks for each input. Each block in the Graphical User Environment represents a Gain, Dynamics, Filter, or AEC algorithm within the DSP engine. While this architecture is fixed, each block offers flexible options and customizable parameters. For example, the Filter block contains several selectable filters, each of which can be customized as parametric EQ, low pass, high pass, bass, or treble. Each processing block can be selectively bypassed.

Emulate and Live Modes

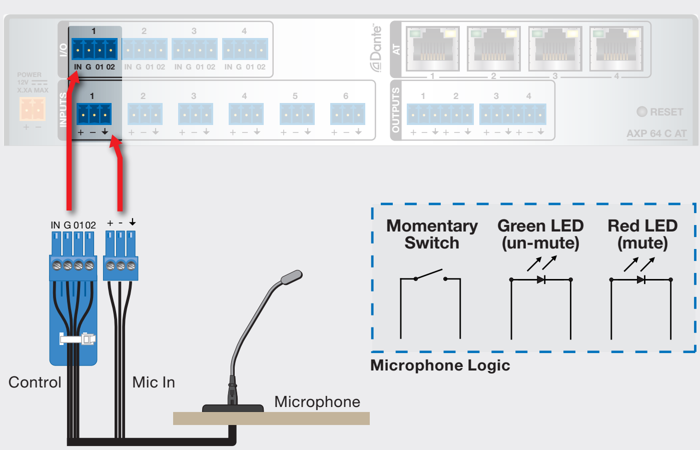
The DSP Configurator Software features an Emulate mode, which allows for complete audio system design while working offline on a PC. When connected to the AXP 64 C AT, Live mode enables real-time control of all settings, file updates, and archiving, plus active metering of all input channels. In Live mode, integrators can "push" all or part of a configuration to the AXP 64 C AT from the PC, while preserving the existing file. Emulate and Live modes give audio system designers the flexibility to create an entire project from their PC in advance of installation, and then, once they are on-site, use the same software to provide accurate system setup and final optimization.

Control

Control and configuration are conveniently available via the built-in four-port Gigabit switch. Any port can be designated for control only. The AXP 64 C AT includes digital I/O ports for the first four inputs, which allow for external triggering such as mic activation and muting, and illuminating mic status LEDs.

Digital Port with Dual Outputs

The AXP 64 C AT provides digital input, ground, and dual digital outputs at each of the first four mic/line inputs for convenient microphone interfacing. This allows button-triggered mic activation and muting, as well as dual mic LED tally back, and is ideal for integration of conference microphones that have dual status LEDs.



Key Features

DANTE TECHNOLOGY OVERVIEW

Dante technology from Audinate provides digital audio distribution over standard local area networks. Dante allows high resolution audio channels to be transported uncompressed across a switched Ethernet data network using standard TCP/IP protocols, while meeting the stringent quality requirements of professional audio. Dante was built on the IEEE 1588 Time Precision Protocol standard to derive a precise clocking mechanism for synchronization. As a result of this, latency of 1 ms is maintained across 10 network switch hops using Dante in a Gigabit Ethernet network, even when using existing Ethernet switch technology. Digital audio signals are converted to packets at the edge of the network, then processed and transported to other Dante-enabled devices.

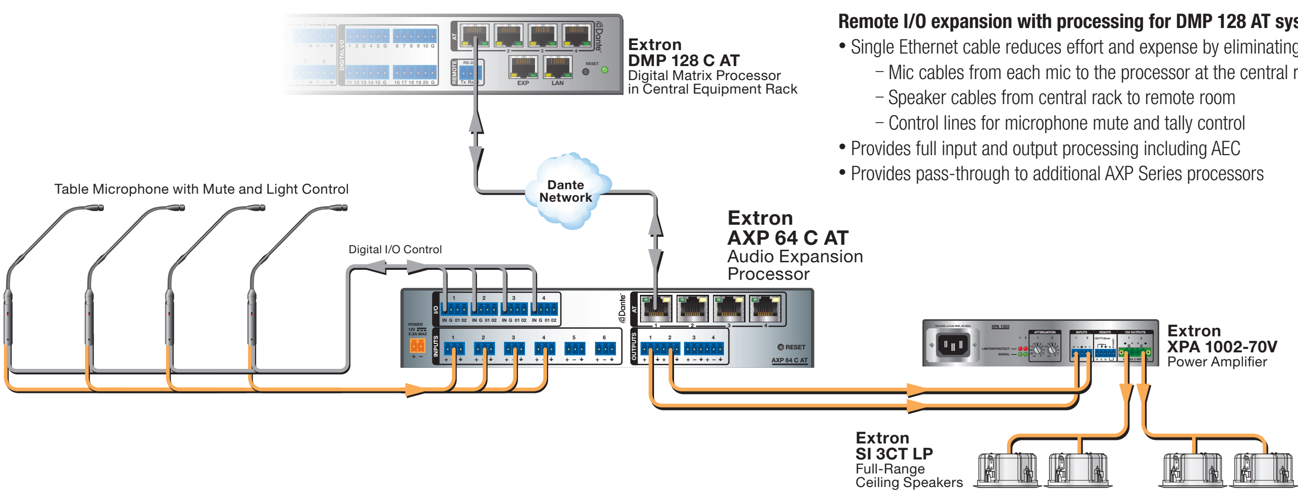
A network with Dante enabled devices can be shared with ordinary data traffic such as e-mail. Audio channels can be transported as unicast or multicast to make the most efficient use of available bandwidth.

With Dante, AXP 64 C AT and DMP 128 AT processors can share multiple channels of high resolution digital audio with each other over a local area network. They can be directly linked using their built-in four-port Gigabit Ethernet switches or by connecting into a network infrastructure. Dante technology distributes up to 512x512 audio channels at 24-bit/48 kHz over a single Gigabit Ethernet link with extremely low latency. Both Dante and the AXP 64 C AT processor's four-port switch are AVB - Audio Video Bridging ready.

INPUT AND OUTPUT EXPANSION USING DANTE

The AXP 64 C AT is a true system expander for a DMP 128 AT, with capability beyond a simple break-in box, digital snake, or break-out box. The built-in DSP processing, coupled with the capabilities and flexibility of Dante networking, create a powerful distributed approach to adding microphones to a system or for routing source audio to systems in various rooms. Traditionally, expanding a system requires that each input or output channel be matched with DSP at the rack. This often necessitates the addition of one or more processors to the system, with only partial utilization of their I/O and DSP capacity.

The AXP 64 C AT greatly reduces equipment, cabling, and installation costs by allowing integrators to remotely locate and process microphone and program channels, insert them onto the Dante network, and receive them as expansion inputs to one or more DMP 128 AT processors, or other Dante-capable device. To further simplify the audio cabling infrastructure, the AXP 64 C AT converts four channels from the Dante network to line level audio signals, processes them, and routes the optimized audio to local sound reinforcement systems.



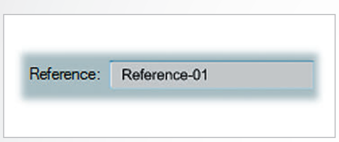
Remote I/O expansion with processing for DMP 128 AT systems

- Single Ethernet cable reduces effort and expense by eliminating:
  - Mic cables from each mic to the processor at the central rack
  - Speaker cables from central rack to remote room
  - Control lines for microphone mute and tally control
- Provides full input and output processing including AEC
- Provides pass-through to additional AXP Series processors

ACOUSTIC ECHO CANCELLATION

In conferencing applications, hearing the talker's voice returned as an echo is disruptive to natural communication. AEC processing prevents far end audio, as reproduced in the near end, from being returned back to the remote talker as echo, ensuring clear, natural conversations. However, AEC processing can be challenged by conditions such as double-talk, when talkers from both ends are speaking simultaneously, and when near end talkers use wireless microphones. Extron AEC delivers fast echo cancelling optimized for these challenging conditions.

The DSP Configurator Software simplifies AEC and noise cancellation setup with a user-friendly interface that provides real-time metering for ERL - echo return loss, ERLE - echo return loss enhancement, and TER - total echo reduction levels. Guided alerts appear whenever ERL is outside of the optimal range for echo cancellation. Optional settings include fine adjustments for NLP - non-linear processing to maximize AEC performance in acoustic environments with significant sonic reflections or reverberation. Each AEC processor in the AXP 64 C AT includes selectable noise cancellation. Extron AEC features advanced algorithms that deliver fast echo canceler convergence for optimal intelligibility, even in challenging conditions.



AEC Dialog Close-Up
An AEC reference can be individually routed to each of the four independent AEC processors of the AXP 64 C AT.

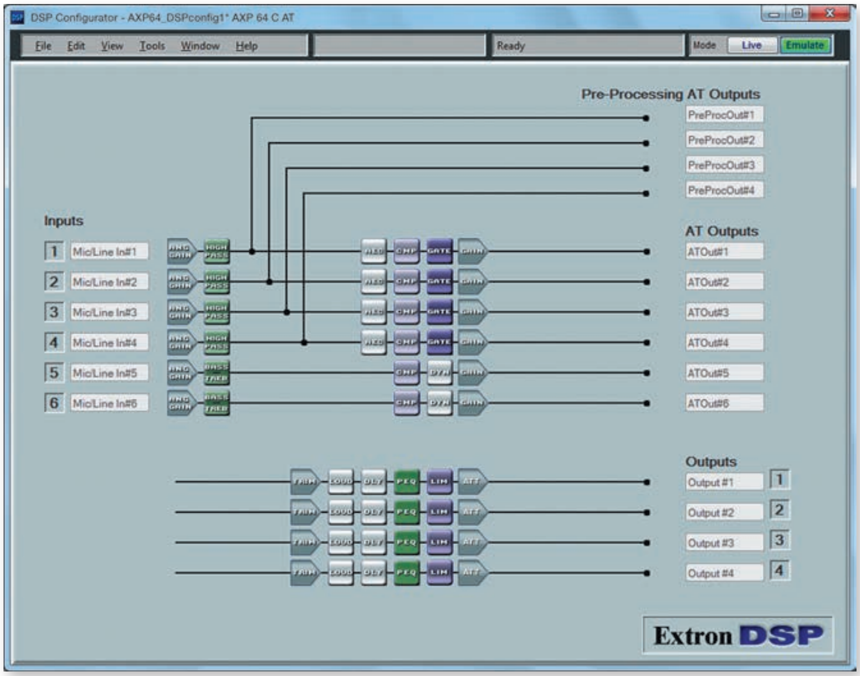
Selecting the AEC Reference

Audio from the far end is reproduced by near end loudspeakers so that listeners can hear the far end talkers. However, this audio can return to the far end via the near end mics, DSP, and codec. To prevent this, AEC processing in the near end analyzes two important signals, the far end audio coming from the conferencing codec or phone input – also known as the AEC reference, and the same audio after being played over the near end speakers into the acoustic space and picked up by the mics. These two signals are analyzed in order to create and apply an adaptive filter to cancel out the far end audio captured at the mic.

The AXP 64 C AT provides the flexibility to use the AEC reference signal from any DMP 128 AT processor over the Dante network. The AEC reference can be independently routed to each of the four channels of AEC processing.

DSP Configurator

EASY-TO-USE DSP CONFIGURATOR SOFTWARE FOR FAST SETUP



Intuitive Graphical User Environment – The DSP Configurator Software features a Graphical User Environment that offers a clear view of all inputs, outputs, and audio processing blocks in a single window. This allows a designer or installer to quickly view the entire configuration without having to access multiple windows or menus.

Extron ProDSP includes all the essential DSP tools needed to set up and fine-tune audio systems. Within the DSP Configurator Software, these tools, or processing blocks, allow for control and management of gain, dynamics, filtering, AEC, and more. Extron digital audio processors serve as the central point for sound system setup and optimization. The easy-to-use configuration approach to setting up these processors saves time during system tuning.

For the AXP 64 C AT, Gain, Filter, and Dynamics blocks are available for each input and output. AEC blocks are also available for the first four inputs. Selecting any of these blocks opens a dedicated pop-up window with a range of options and customizable parameters. Multiple windows can be open at the same time. Input levels can be monitored by simply opening any of the pre-processing Gain windows.

To further expedite configuration, Extron Building Blocks can be applied to any of the inputs. Each Building Block includes gain and processor settings optimized for a specific type of input device, such as microphones, with preset levels, filters, dynamics, and more. Building Blocks can also be customized and saved, or created by capturing the current gain and processor settings for an input.

EXTENSIVE ARRAY OF DSP TOOLS

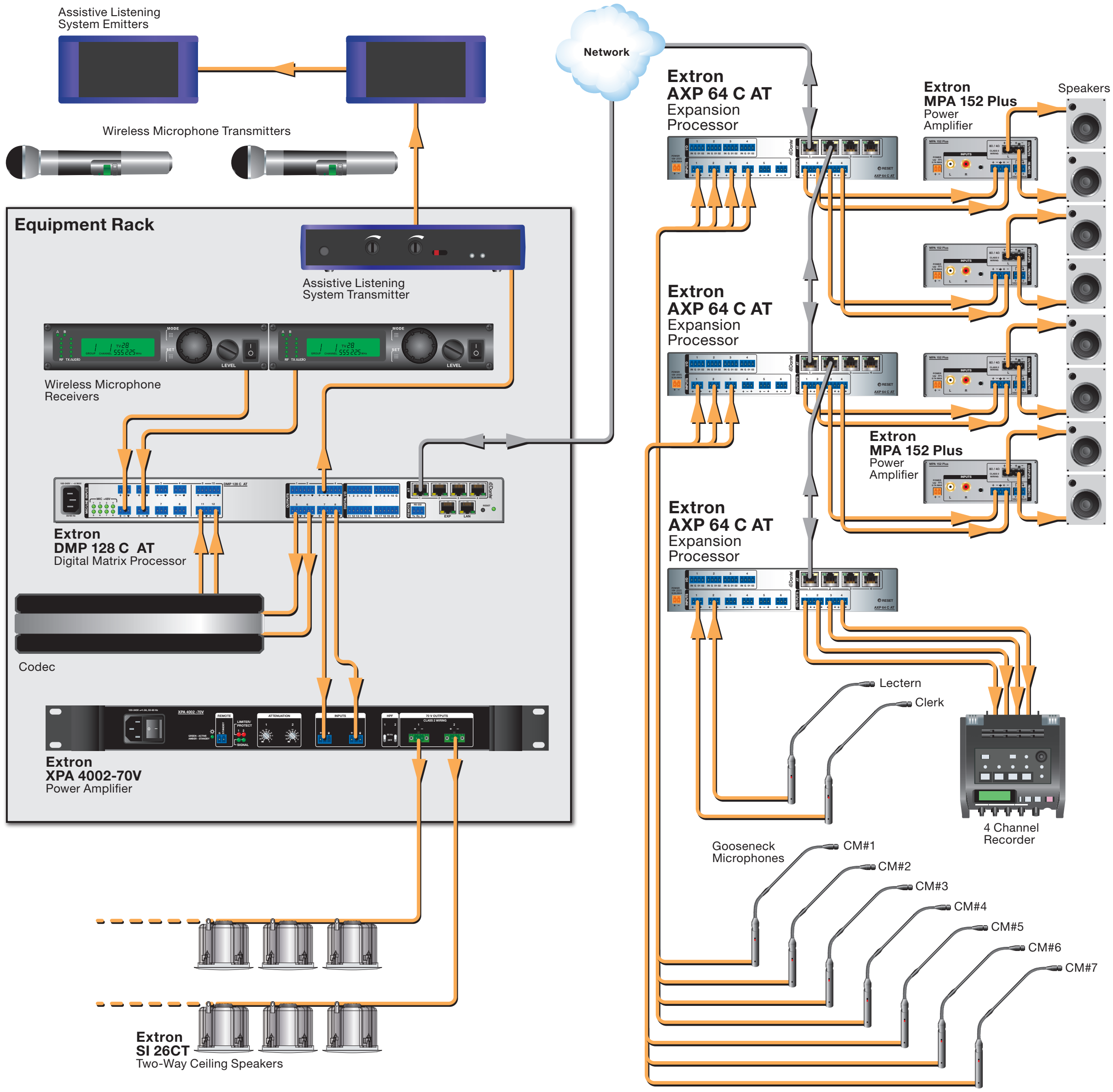
AEC	4 independent acoustic echo cancellers with selectable noise reduction
GAIN STAGES	2 gain stages across input paths; 2 gain stages across output paths
DYNAMICS	AGC - Automatic Gain Control Compressor Limiter Noise gate
FILTERS	5 filters per input High pass Low pass Band pass Shelving Parametric EQ
DELAY	Up to 200 ms; available on all outputs

Application

CITY COUNCIL CHAMBER

A high performance audio system for a city council chamber necessitates a microphone and dedicated speaker at each council bench seat, plus audience sound reinforcement and other functions. With multiple mic inputs, speaker zones, audio recording, conferencing with AEC, and an assistive listening system to integrate, the challenge arises in creating a suitable system design that is not overly complex or expensive. An ideal solution is an Extron DMP 128 C AT digital matrix processor with Dante on a standard IP network with several AXP 64 C AT expansion processors.

The DMP 128 C AT is installed in the central equipment rack and serves as the master DSP matrix for the audio system. The compact size of the AXP 64 C AT units enables them to be situated inside the millwork for the council bench, in close proximity to the microphones as well as the discreetly installed, quarter rack width MPA 152 Plus amplifiers driving each speaker as a separately mixed audio zone. The combination of local DSP with AEC at the microphones and speakers, plus a Dante network of master and expansion audio processors results in an efficient, properly scaled system design that eliminates the added expense and effort of running numerous audio microphone, line, and speaker cables back to the central rack. A large bundle of cables is reduced to just a single network cable.



Specifications

AUDIO	
Gain	-6 dB unbalanced output, 0 dB balanced output
Frequency response	20 Hz to 20 kHz, ±0.2 dB
THD + Noise	<0.01%, 20 Hz to 20 kHz, at maximum level
S/N	>105 dB, 20 Hz to 20 kHz, at maximum balanced output, unweighted
Crosstalk	≤-90 dB @ 20 Hz to 20 kHz, fully loaded
Volume control	-100 dB to +12 dB in 0.1 dB steps
AUDIO INPUT	
Number/signal type	6 mono, mic/line, balanced/unbalanced
Connectors	(6) 3.5 mm captive screw connectors, 3 pole
Impedance	>10k ohms unbalanced/balanced
Nominal level	-60 dBV, +4 dBu, -10 dBV, adjustable via input gain
Maximum level	>+21 dBu, at rated THD+N when input gain is set to 0 dB
Equivalent input noise	<-120 dBV (1 μVrms) at 40 dB gain
Input gain adjustment (mic)	-18 dB to +80 dB, in 0.1 dB steps, adjustable per input
CMRR	>60 dB typical
DC phantom power	+48 VDC ±10%, inputs 1-4, can be switched on or off
NOTE: 0 dBu = 0.775 Vrms, 0 dBV = 1 Vrms, 0 dBV ≈ 2 dBu	
AUDIO OUTPUT	
Number/signal type	4 mono (or 2 stereo), balanced/unbalanced
Connectors	(2) 3.5 mm captive screw connectors, 6 pole
Impedance	50 ohms unbalanced, 100 ohms balanced
Gain error	±0.1 dB channel to channel
Maximum level (Hi-Z)	>+21 dBu balanced, +15 dBu unbalanced
AUDIO PROCESSING	
A/D conversion	24 bit, 48 kHz sampling
AT PORTS	
Transmission type	Dante over TCP/IP; AVB ready
Connectivity	(4) RF-45 connectors, 4-port 1 Gbps switch to Dante interface
Inputs	10 channels Rx, 5 channels AEC reference only
Outputs	10 channels Tx, pre- and post-processing
Audio format	24 bit, 48 kHz sampling, uncompressed
Latency	Deterministic, based on user selections: 0.25 ms, 1.0 ms (default), 5.0 ms
COMMUNICATIONS	
Ethernet host port	4 RJ-45 female, 1 can be configured for control only
Ethernet data rate	Gigabit, half/full duplex with autodetect
Ethernet default settings	Link speed and duplex level = autodetected DHCP = on
Program control	Extron DSP Configurator control/configuration program for Windows® Extron Simple Instruction Set (SIS™), Telnet

GENERAL	
Power supply	External Input: 100-240 VAC, 50-60 Hz Output: 12 VDC, 1 A, 12 watts
Power consumption	
Device	TBD
Device and power supply	TBD
Temperature/humidity	Storage: -40 to +158 °F (-40 to +70 °C) / 10% to 90%, noncondensing Operating: +32 to +122 °F (0 to +50 °C) / 10% to 90%, noncondensing
Cooling	Convection, no vents
Thermal dissipation	
Device	TBD BTU/hr
Device and power supply	TBD BTU/hr
Mounting	
Rack mount	Yes, with optional 1U rack shelf
Furniture mount	Yes, with optional under-desk mounting kit
Enclosure type	Metal
Enclosure dimensions	1.7" H x 8.75" W x 9.5" D (1U high, half rack wide) (4.3 cm H x 22.2 cm W x 24.1 cm D) (Depth excludes connectors.)
Product weight	1.9 lbs (0.9 kg)
Shipping weight	4 lbs (2 kg)
Vibration	ISTA 1A in carton (International Safe Transit Association)
Regulatory compliance	
Safety	CE, c-UL, UL
EMI/EMC	CE, C-tick, FCC Class A, ICES, VCCI
Environmental	Complies with the appropriate requirements of RoHS, WEEE
Warranty	3 years parts and labor
NOTE: All nominal levels are at ±10%.	

Model	Version Description	Part number
AXP 64 C AT	6 In, 4 Out Expansion Processor	60-1499-01
DMP 128 C AT	12x8 ProDSP Proc. w/AEC and Dante	60-1178-10
DMP 128 C P AT	12x8 ProDSP Proc. w/AEC, POTS and Dante	60-1179-10
DMP 128 AT	12x8 ProDSP Processor w/Dante	60-1211-10
AXP 50 C AT	5 In Expansion Processor	60-1325-01

For complete specifications, please go to www.extron.com
Specifications are subject to change without notice.

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