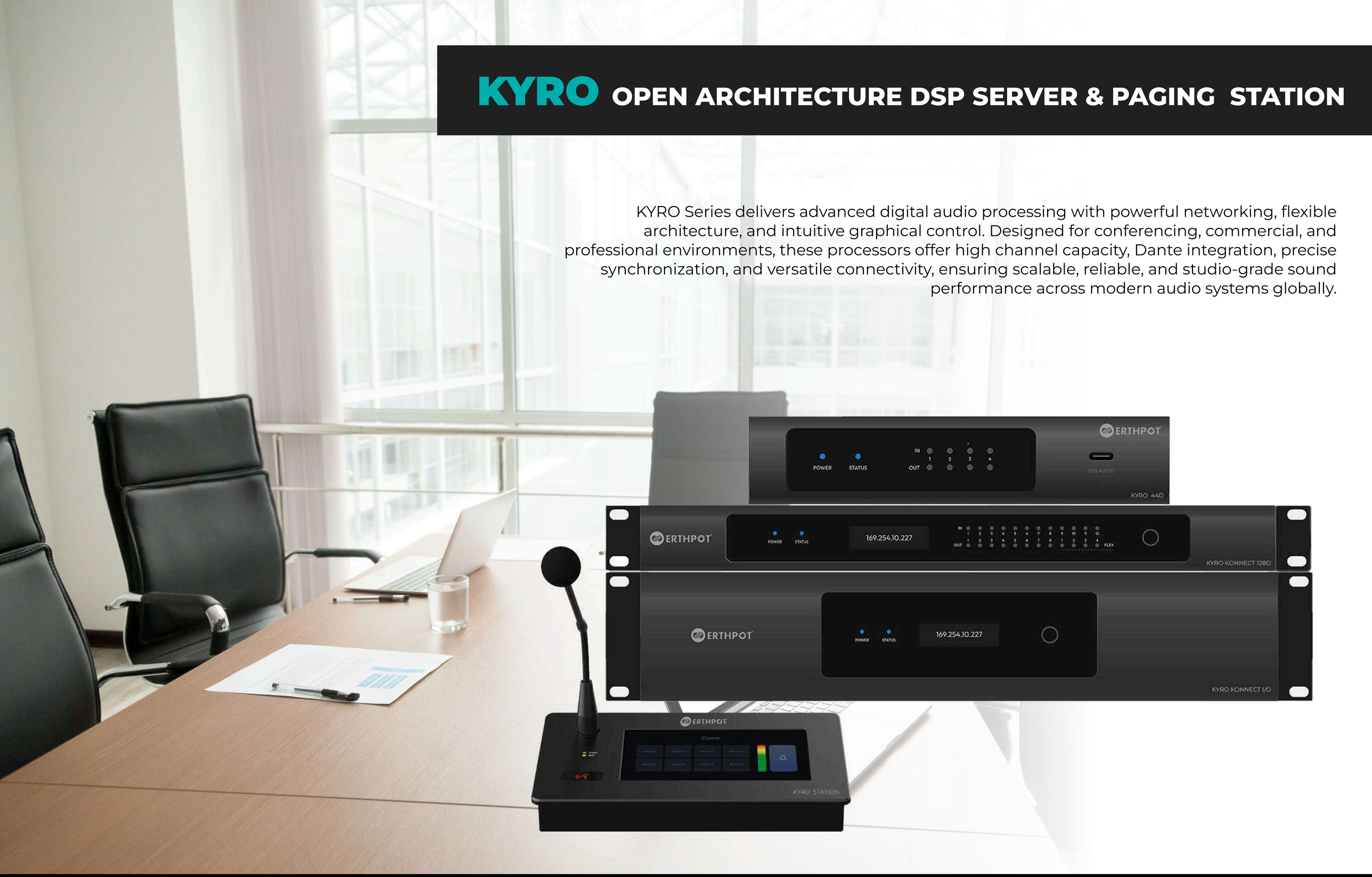
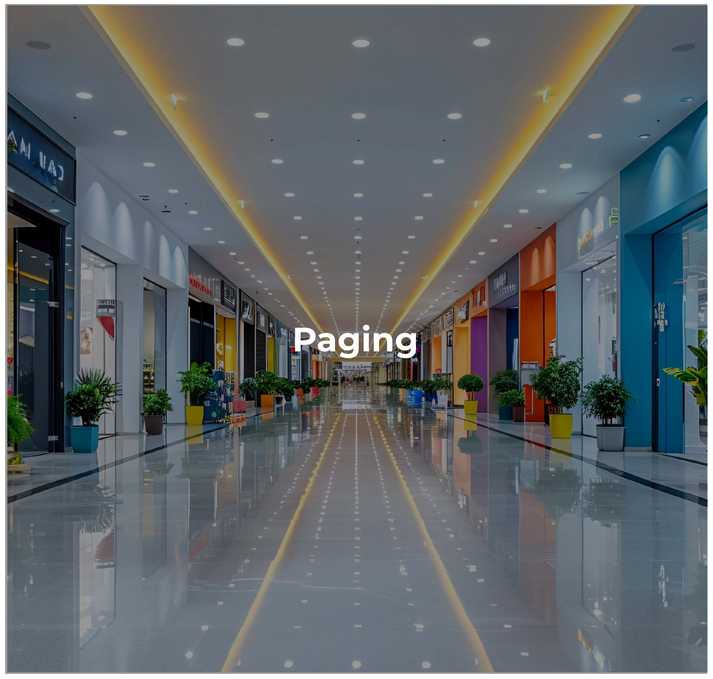
KYRO OPEN ARCHITECTURE DSP SERVER & PAGING STATION

KYRO Series delivers advanced digital audio processing with powerful networking, flexible architecture, and intuitive graphical control. Designed for conferencing, commercial, and professional environments, these processors offer high channel capacity, Dante integration, precise synchronization, and versatile connectivity, ensuring scalable, reliable, and studio-grade sound performance across modern audio systems globally.





FEATURES

Flexible Analog I/O

Supports multiple analog signal inputs and outputs, enabling versatile multi-speaker system configurations.

High-Performance DSP Processing

Built-in 24-bit/48kHz DSP chip with premium A/D and D/A converters ensures high-fidelity signal handling.

Intuitive User Interface

Interactive LCD display with a quick-access, embedded GUI interface for efficient system navigation and control.

Advanced FIR Filtering

512-tap custom FIR filter with stable, linear-phase performance for precise audio tuning.

Third-Party FIR Compatibility

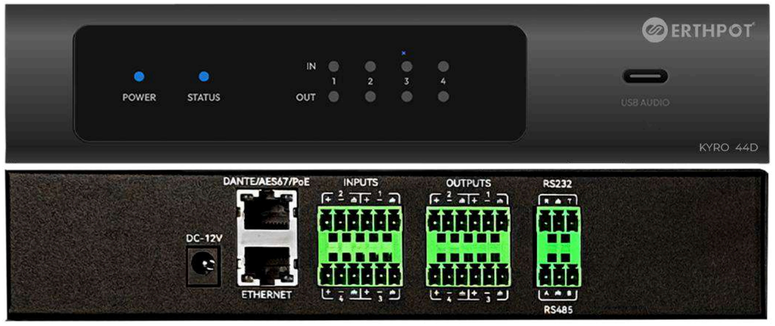
Supports import of external FIR parameters via a graphical interface for customized sound optimization.

Real-Time Control

Graphical interface enables accurate filter adjustments and real-time system monitoring.

KYRO 44D

Semi-Open architecture DSP



FEATURES

. ADI SHARC platform with semi-open architecture.
. Integrated management of multiple devices offers open user interface and supports good sound quality (at 24 bit/48kHz.)
. Each channel has its own independent adaptive feedback suppression which can find and suppress audio feedback automatically. Full-duplex Adaptive Echo Cancellation (AEC) and Active Noise Cancellation (ANS).
. Gain sharing Auto Mixer Control (AMC).
. Automatic Gain Control (AGC).
. Full-featured audio mixing matrix supports adjustable input level.
. Provides up to 16 presets, each can work independently.
. Supports audio channel duplicating, linking, and grouping.
. Supports RS-232/485 and UDP central control.
. Provides the flexibility to assign UDP port number, with the control commands available through the control software. With USB type-C for PC connect.
. Built-in dual channel USB sound card can be used for audio recording, audio broadcasting, and video conference.
. Supports both PoE power supply and DC 12V power adapter.
. Supports iOS and Windows operating system.

OVERVIEW

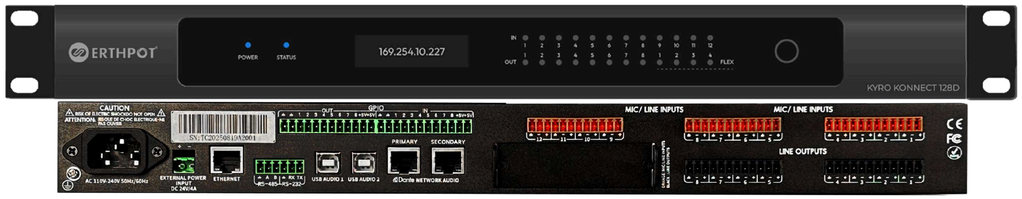
The ERTHPOT KYRO 44D is a compact 1U half-rack audio processor built on an ADI SHARC platform. It offers a semi-open architecture for integrated device management through a flexible user interface. The KYRO 44D provides high-quality 24bit/48kHz audio. It includes 4 analog input/output channels and 4x4 Dante input/output channels. A key feature is its built-in camera tracking. The device also has a comprehensive audio mixing matrix with adjustable input levels and supports up to 16 independent presets. It enables versatile audio channel management, including duplicating, linking, and grouping functions. For control, the KYRO 44D supports RS-232/485 and UDP central control, with flexible UDP port assignment and software-based commands. Connectivity includes a USB Type-C port for PC connection, featuring a built-in dual-channel USB sound card for audio recording, broadcasting, and video conferencing. The KYRO 44D can be powered via Power over Ethernet (PoE) or a 12V DC adapter and is compatible with iOS and Windows.

TECHNICAL SPECIFICATIONS

Processor	ADI SHARC 21489
Sampling rate/quantization bits	48K/24bit
No. of analog input and output channels	4x4
No. of DANTE input and output channels	4x4
Input gain	0/6/12/18/24/30/36/42/48dB
Phantom power	+48V/10mA max
Frequency response	±0.5dB(20~20kHz
Maximum level	+18dBu
THD+N	<-95dB @17dBu
Input/Output dynamic range	113dB
Camera Tracking	Support
Channel isolation @1kHz	108dB
Input impedance (balanced connection)	5.4KΩ
Output impedance (balanced connection)	600Ω
System delay	<3ms
Working power supply	DC 12V or PoE 48V
Dimensions (width x depth x height)	215x162x44mm
Shipping weight	2KG

KYRO KONNECT 128D

Open architecture DSP server



FEATURES

. Equipment type: audio processing server.
. Drag-and-drop DSP module (non-fixed), unified management.
. 12 microphone/line level analog audio inputs, 8 line level analog audio outputs.
. Supports customization of 4 line level analog input/output channel cards, with a maximum of 16 inputs and 8 outputs or 12 inputs and 12 outputs; USB channel: Dual USB interface, supports two computers to be used at the same time, with a maximum of 4 inputs and 4 outputs.
. Audio playback: built-in 4 multi-track players, supports scheduled playback, parallel 16-track playback, supports MP3 and WAV audio files.
. Media storage capacity: up to 64 GB.
. Network channel: 128×128 Dante network channel, supports the number of streams 128*128.
. AEC channel: up to 12 assignable and routable AEC processor modules.
. Feedback suppressor channel: 144 channels.
. Dynamic Dynamic range: ≥118dB.
. Total harmonic distortion: ≤0.002%.
. Control port: supports 8 GPI and 8 GPO channels, 1 RS-232 interface, 1 RS-485 interface.
. User interface: The graphical user interface can be edited as needed.
. Power supply: built-in universal power module and 24V 4A DC external power supply, dual power supply backup.
. The front panel has input and output signal indicators, OLED display, and touch page turning buttons.

OVERVIEW

KYRO KONNECT 128D from the ERTHPOT KYRO product line is a next-generation digital audio matrix processor designed for high-performance audio systems. It features 12 mic/line inputs, 8 line outputs, and 4 customization I/O channels (expandable up to 16×8 or 12×12 configuration), along with dual USB support. The processor supports 128×128 Dante network audio transmission for seamless, ultra-low latency digital audio over IP networks. Built with flexible drag-and-drop DSP architecture, up to 12 AEC modules, 144 feedback suppression channels, and a dynamic range of ≥118 dB, it ensures precise and reliable audio performance. With an interactive graphical control interface, dual power supply backup, and built-in media playback (up to 64 GB storage), the KONNECT 128D is ideal for conference rooms, council halls, and medium-sized professional audio environments requiring advanced processing and networked audio integration.

TECHNICAL SPECIFICATIONS

Processor Chipset	ADI SHARC 21569
Channel isolation	108dB
Sampling rate/number of quantization bits	48K/24bit
Input impedance (balanced connection)	5.4KΩ
Input gain -adjustable in 6 dB increments	0/4/12/18/24/30/36/ 42/ 48dB/54dB
Output impedance (balanced connection)	600Ω
Phantom power	+48V/10mA max
Working power supply	AC 100-240V, 50/60Hz and DC24V 4A
Frequency response (20~20kHz)	±0.3dB
Maximum level	+22dBu
Dante Primary/Secondary	RJ451000 Mbps
THD+N	≤0.002%
Dante Channel	128x128
Input/Output dynamic range	118dB
Product dimensions (width × height × depth)	482 x 260 x 45mm
Information screen	2.08-inch flip display

KYRO KONNECT I/O

Open architecture DSP server

FEATURES

. Equipment type: Optional board plug-in audio processing server.
. Drag-and-drop DSP module (non-fixed), unified management.
. This machine has 8 expandable card slots, and can be equipped with 4-channel microphone/ line input card, 4-channel line output card, 4-channel AEC (acoustic echo cancellation) microphone/line input card, 2-channel analog phone card, 2-channel VOIP card, 2x2USB sound card.
. Transmission method: equipped with dual network hot backup transmission.
. AEC channel: local 20 channels, supports adding AEC card, can support 80 channels of AEC.
. Audio playback: Built-in 16-channel player, supports MP3 and WAV audio files.
. Dante network channels: 128x128 channels of Gigabit network transmission with network transmission backup.
. AEC: Supports custom addition of independent AEC modules, with a maximum of 16 independent AEC.
. Playback/recording: 4-channel audio playback supports timed playback of WAV and MP3, and 1-channel lossless recording; Built-in white noise, pink noise and other signal generators.
. Graphical user interface that can be edited as needed, supporting control on the software side, web side, and mobile side.
. Built-in Lua script: provides flexible expansion and customization functions. Dual power supply redundancy (AC/DC)

OVERVIEW

KYRO KONNECT I/O, the flagship model from the ERTHPOT product line, is a high-performance network audio processing server powered by three 1GHz processors and a real-time Linux OS.
It supports 128×128 network audio transmission, up to 32 local analog channels, expandable card slots (analog I/O, AEC, USB), and advanced AEC configurations. With dual host backup, intelligent redundancy, and seamless mirror fail over, it ensures reliable, uninterrupted operation.
Featuring drag-and-drop DSP, customization control interfaces, Lua scripting, and integration with wall controllers and touch screens, it is ideal for large-scale applications such as airports, railway stations, theme parks, and commercial complexes.

TECHNICAL SPECIFICATIONS

Audio/IO capacity	8 audio/IO card slots
USB storage capacity	64GB
Audio/IO capacity Dante network channel capacity	128 × 128
Dynamic range	> 118 dB
Frequency response (± 0.2 dB)	20 Hz to 20 kHz
Phantom power	+48V/10mA max
Frequency response	(20~20kHz):±0.3dB
Input impedance	5.5 kΩ
Signal-to-noise ratio	< -112 dB
THD + N	≤ 0.002%
Common mode rejection ratio (@ 0 dBr)	> 91 dBu
Maximum input level (@ 1% distortion)	+ 22 dBu
Dante Primary/Secondary	RJ45 1000 Mbps
Voltage	220 VAC - 240 VAC, 50 Hz
AC main power supply AC main power supply	IEC connector
Product dimensions (width × height × depth)	483 × 88 × 364 mm
Transport weight	8.5 KG

KYRO STATION Paging Microphone

OVERVIEW

The modern choice for intelligent and seamless communication

FEATURES

7-inch Android Touch Screen
The paging station integrates a 7-inch Android touch screen, functioning as a control interface. User interface can be designed and tailored according to the audio processor for the purpose of system control and monitoring.

Built-in Microphone Preamp
Each paging station comes standard with a condenser gooseneck microphone, driven by 48V phantom power.And a 0-30dB mic preamp gain knob is located on the rear panel to achieve an optimal level based on the signal indicator from the front.

Digital Transmission
The paging station supports Dante audio transmission, enabling microphone signals to be transmitted to audio processor or mixing console for routing via Dante. It can be utilized in paging or intercom systems due to its flexibility.

Dual Power Supply
The paging station supports both PoE and 12DC power supply, offering flexible deployment based on requirements.

Analog Output Switching
The Source toggle on the rear panel of the paging station allows switching between analog and digital. When set to the A (analog) position, the microphone signal is output through the analog terminal on the rear panel of the paging station. When switched to the D (Digital) position, the analog terminal outputs the Dante received signal while microphone keeps paging through the Dante transmit channel.

Dual-mode Paging
Mode 1, the microphone button is Push-To-Talk by default, press and hold it to speak. Mode 2, triple-tap enters the Lock-on mode, enabling hands-free speaking. When finish paging ,press the button again can revert to the PTT mode.

TECHNICAL SPECIFICATIONS

System	Android 10.00
Resolution	7"RGB 1024*600
Processor	A133 1.6GHz Cortex A53 4 core 64bit
RAM	2GB
Storage	16GB
Dante Channel	1x1
Analog Output	1 Balanced Output (Source-selectable Output Signal)
Mic Gain	0-30 dB 1dB step
Mic Polar Pattern	Hypercardioid
Frequency Response	20Hz~20kHz(0/-0.6 dB)
Maximum Level	+13dBu
Sensitivity	-40dB (1V \ 1 Pa)
Sampling Rate	48 kHz
THD+N	≤0.003% (1kHz, A-weighted)
Input/Output Dynamic Range	≥104dB
SNR	≥103dB
Noise	≤-91dBu (1kHz, +13dBu A-weighted)
Output Impedance	600Ω
Power Supply	12V DC or PoE (IEEE 802.3af) power supply
Dimension (W x D x H)	260x155x65mm (No microphone included)