Dante Speaker: Dante DP 615POE

Description

Dante DP PoE Speaker can realize its multifunction the POE power supply or 24VDC, audio transmission and signal control via a RJ45 Ethernet port. It can obtain power supply from any Ethernet switch of 802.3 or above standard.

Over Dante™ Ethernet that transmits 24-bit audio (44.1K/48KHz) and its control signal can be routed through the Dante™ Controller. All these functions can perform remotely on the PC or Mac by Dante Controller Software.

Ordering Code

Dante DP 615:

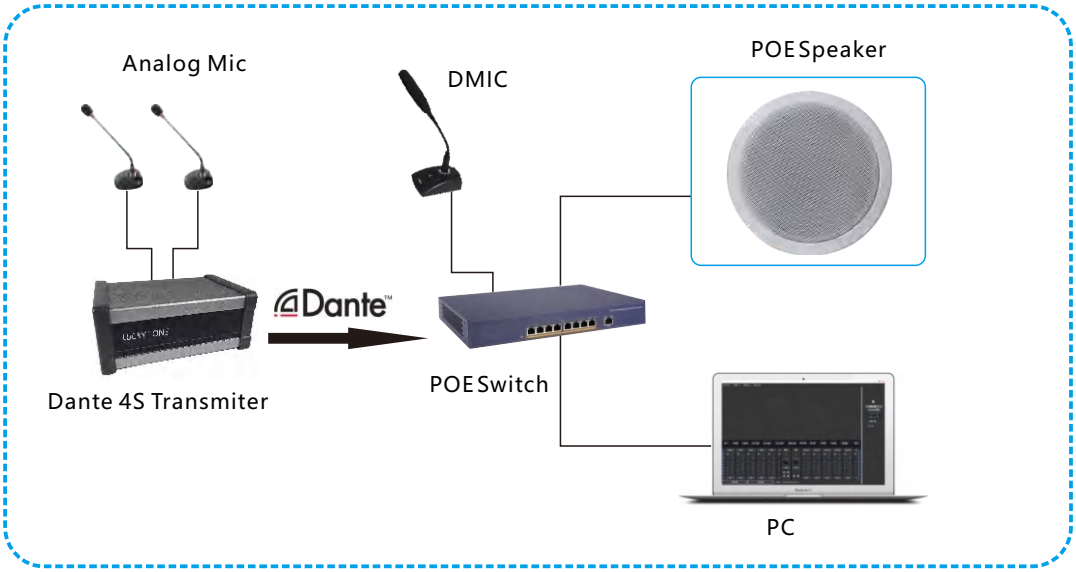
15W Dante 6-inch Metal Ceiling Speaker with External 24V DC power supply

Dante DP 615 POE:

15W Dante 6-inch Metal Ceiling Speaker with POE power supply

MODEL	Dante DP 615POE
Input Interface	RJ-45
Audio Input Signal	Dante™ | Balanced amplified signal
Cable	Cat-5 or above
Output Power	15W
Transmission Rate	100M/1000M bps
Frequency Range	80Hz - 18kHz
Sensitivity	91dB 1w/1m
Sound Pressure Level	106dB (peak)
Total Harmonic Distortion	<0.30%
Signal to Noise Ratio	>100dB
Noise Floor	<200uV
System Efficiency	>70%
Enclosure	Metal
Operation Temperature	-10 to 50 Degree
Color	RAL-9010
Dimension	Dia 200 x 90mm
Net Weight	0.95kg

Example Diagram



Dante Speaker: Dante DP 430

Description

Dante DP PoE Speaker can realize its multifunction the POE power supply or 24VDC, audio transmission and signal control via a RJ45 Ethernet port. It can obtain power supply from any Ethernet switch of 802.3 or above standard.

Over Dante™ Ethernet that transmits 24-bit audio (44.1K/48KHz) and its control signal can be routed through the Dante™ Controller. All these functions can perform remotely on the PC or Mac by Dante Controller Software.

Front

Rear

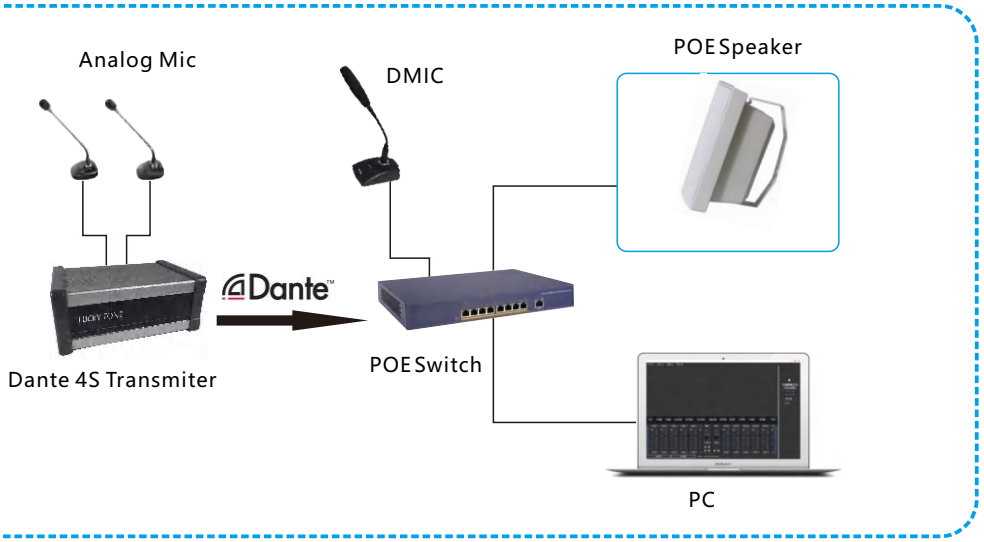
Ordering Code

DANTE DP 430:
30W Dante Wall Speaker with External 24V DC power supply
DANTE DP 430 POE:
30W Dante Wall Speaker with POE power supply
DANTE DP 430 POE With DSP:
30W Dante Wall Speaker with POE power supply and DSP

Side

MODEL	Dante DP 430
Input Interface	RJ-45
Audio Input Signal	Dante™ | Balanced amplified signal
Cable	Cat-5 or above
Output Power	30W
Transmission Rate	100M/1000M bps
Frequency Range	80Hz - 20kHz
Sensitivity	90dB 1w/1m
Sound Pressure Level	108dB (peak)
Total Harmonic Distortion	<0.30%
Signal to Noise Ratio	>100dB
Noise Floor	<200uV
System Efficiency	>70%
Enclosure	ABS
Operation Temperature	-10 to 50 Degree
Color	RAL-9010
Dimension	(High)295 x (Width)195 x (Depth)155mm
Net Weight	2.6kg

Example Diagram



Dante Speaker: DANTE WP 630 POE

Description

Dante DP PoE Speaker can realize its multifunction the POE power supply or 2 4VDC, audio transmission and signal control via a RJ45 Ethernet port. It can obtain power supply from any Ethernet switch of 802.3 or above standard. Over Dante™ Ethernet that transmits 24-bit audio (44.1K/48KHz) and its control signal can be routed through the Dante™ Controller. All these functions can perform remotely on the PC or Mac by Dante Controller Software.

Ordering Code

DANTE WP 630 POE:
Dante 6-inch Wall speaker with built-in 2x30W amplifier, POE power supply in Pair
DANTE WP 630:
Dante 6-inch Wall speaker with built-in 2x30W amplifier, 24VDC power supply in Pair

MODEL	DANTE WP 630 POE
Input Interface	RJ-45
Audio Input Signal	Dante™ | Balanced amplified signal
Cable	Cat-6 or above
Output Power	30W*2
Transmission Rate	100M/1000M bps
Frequency Range	80Hz - 20kHz
Sensitivity	90dB 1w/1m
Sound Pressure Level	108dB (peak)
Total Harmonic Distortion	<0.30%
Signal to Noise Ratio	>100dB
Noise Floor	<200uV
System Efficiency	>70%
Enclosure	ABS
Operation Temperature	-10 to 50 Degree
Color	RAL-9010
Dimension	215*284*190mm
Net Weight	3.5kg

PoE Dante Speaker: DP 180POE DSP

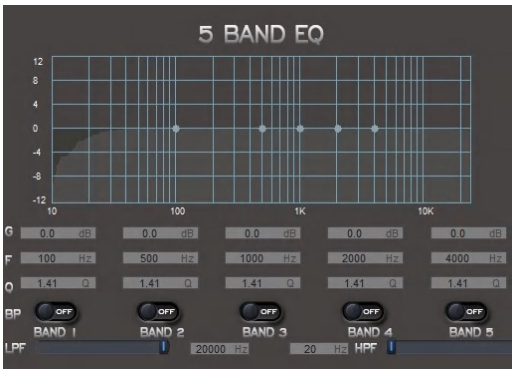
Description

DP 180POE DSP Speaker can realize its multifunction the POE power supply, audio transmission and signal control via a RJ45 Ethernet port. It can obtain power supply from any Ethernet switch of 802.3 or above standard. DP 180POE DSP adopts SSA inverter technology allowing that the high-quality enlargement of superior sound quality realizes on this compact speaker of low power design. The system built with plastic body and 6pcs 3 inch high efficient Neodymium driver.

Over Dante™ Ethernet that transmits 24-bit audio (44.1K/48KHz) and its control signal can be routed through the Dante™ Controller. DP 180POE DSP integrates versatile functions like gain control, mute control, 5-band equalizer control and DSP with limit control. All these functions can perform remotely on the PC or Mac by DSP Controller Software.

Features:

- Built-in Dante module
- Built-in power amplifier module and its system
- efficiency > 70%
  DSP controlling software
- PoE power supply
- Over-current protection, Thermal protection and Overload protection



Control software

Ordering Code

DP 180POE DSP: Column speaker with DSP and Dante

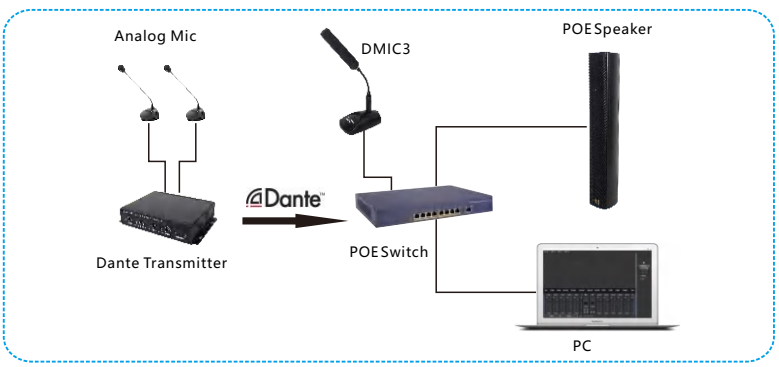
DP 180: Passive column speaker

Black as default and white color optional

MODEL	DP 180POE DSP	DP 180
Type	Column speaker with DSP and Dante	Passive column speaker
Input Interface	RJ-45	
Audio Input Signal	Dante™ | Balanced amplified signal	
Cable	Cat-5 or above	
Output Power	180W@6ohm,POE boost up to 60W depends on router power	
Bandwidth	100M/1000M bps	
Frequency Range	110Hz-19KHz(±3dB)	
Sensitivity	95dB (1w/1m)	
Max. SPL	120dB(continuous)	
Total Harmonic Distortion	<0.30%	
Signal to Noise Ratio	>100dB	
Noise Floor	<200uV	
System Efficiency	>70%	
Driver	6*3″ Lavoce driver	
Protective Circuitry	Over-current protection, Thermal protection, Overload protection.	
Cooler	Natural convection cooling	
Dimension	(Height)605 x (Width)90 x (Depth)120mm	
Net Weight	3kg	

Built-in Italian LAVOCE ferrite magnet steel basket drivers

Example Diagram



Speaker with DSP, POE power supply

Description

DP 90 DSP Speaker can realize its multifunction the POE power supply, audio transmission and signal control via a RJ45 Ethernet port. It can obtain power supply from any Ethernet switch of 802.3 or above standard. DP 90 DSP adopts SSA inverter technology allowing that the high-quality enlargement of superior sound quality realizes on this compact speaker of low power design.

Over Dante™ Ethernet that transmits 24-bit audio (44.1K/48KHz) and its control signal can be routed through the Dante™ Controller. DP 90 DSP integrates versatile functions like gain control, mute control, 5-band equalizer control and DSP with limit control. All these functions can perform remotely on the PC by DSP Controller Software.

Features:

- Built-in Dante module
- Built-in power amplifier module and its system efficiency > 70%
- DSP controlling software
- PoE power supply
- Over-current protection, Thermal protection and Overload protection

MODEL	DP 90 DSP
Type	Speaker with DSP and Dante
Input Interface	RJ-45
Audio Input Signal	Dante™ | Balanced amplified signal
Cable	Cat-5 or above
Output Power	90W, POE boost up to 60W depends on router power
Bandwidth	100M/1000M bps
Frequency Range	70Hz—18KHz(±3dB)
Driver	Lavoce Italy driver 6.5 inch coaxial
Sensitivity	91dB (1w/1m)
Max. SPL	117dB(continuous)
Total Harmonic Distortion	<0.30%
Signal to Noise Ratio	>100dB
Noise Floor	<200uV
System Efficiency	>70%
Protective Circuitry	Over-current protection, Thermal protection, Overload protection.
Cooler	Natural convection cooling
Dimension	(Height)305 x (Width)245 x (Depth)235mm
Package	(Height)369 x (Width)581 x (Depth)296mm
Net Weight	6kg

Subwoofer speaker with DSP, POE power supply

Description

DP 120 SUB DSP Speaker can realize its multifunction the POE power supply, audio transmission and signal control via a RJ45 Ethernet port. It can obtain power supply from any Ethernet switch of 802.3 or above standard. DP 120 SUB DSP adopts SSA inverter technology allowing that the high-quality enlargement of superior sound quality realizes on this compact speaker of low power design.
Over Dante™ Ethernet that transmits 24-bit audio (44.1K/48KHz) and its control signal can be routed through the Dante™ Controller. DP 120 SUB DSP integrates versatile functions like gain control, mute control, 5-band equalizer control and DSP with limit control. All these functions can perform remotely on the PC by DSP Controller Software.

Features:

- Built-in Dante module
- Built-in power amplifier module and its system efficiency > 70%
- DSP controlling software
- PoE power supply
- Over-current protection, Thermal protection and Overload protection

MODEL	DP 120 SUB DSP
Input Interface	RJ-45
Audio Input Signal	Dante™ | Balanced amplified signal
Cable	Cat-5 or above
Output Power	120W, POE boost up to 60W depends on router power
Bandwidth	100M/1000M bps
Frequency Response	50Hz-150Hz
Driver	1 x 8" woofer
Nominal Impedance	8ohm
Sensitivity	92dB
Max SPL	119dB
DSP	One way 5-band PEQ/mute control
Power Supply	PoE
Product Dimension	290x400x330mm
Shipping Dimension	834x493x507mm
Net Weight	12kg

Dante Desktop microphone with POE power supply

Description

DMIC Dante Microphone is a type of microphone that with built-in Dante module supporting POE power supply. It can be directly connected to the Dante network, providing you a rewarding experience that a extremely low latency, high fidelity and low cost solution for paging and announcement.

Features:

- ▶ Built-in Dante module
- ▶ Rotate gain control
- ▶ POE power supply
- ▶ Smart on/off switch

MODEL	DMIC 3
Input	Cat 5 connector
Transmission	Dante Network
Wire range	Cat-5 or above
Gooseneck size	300mm or made to order
Sensitivity	-45 dB
Capsule type	Uni directional
Frequency Response(20~20KHz)	±0.2dB
Power Supply	PoE IEEE802.3af
Signal to Noise Ratio	65dB, 1kHz@1Pa
Power	2.3W
Base Size	94*143*44mm
Weight	0.87kg

Dante 2 In 2 Out PCB Board

Dante 2 PCB

Dante 4 In 4 Out PCB Board

Dante 4 PCB

Features:

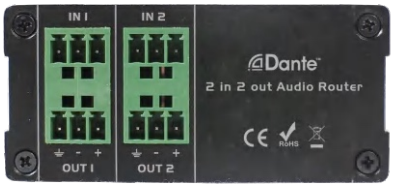
- Ideal interface to increase the number of microphones or AUX inputs
- 2/4 balanced line outputs can be used simultaneously, and all audio channels are used with Dante network.
- 2/4 balanced MIC/AUX inputs, and also 2/4 outputs using phoenix connector
- CAT5/6 connector to link with Dante network and controlled by Audinate

MODEL	Dante 2 PCB	Dante 4 PCB
Input	2 balanced/unbalanced input	4 balanced/unbalanced input
Input Impedance	Greater than1800 Ω	Greater than1800 Ω
Max. Input Level	+8dbu@ 0dB gain	+8dbu@ 0dB gain
Output	2 balanced/unbalanced output	4 balanced/unbalanced output
Output Impedance	200Ω balance	200Ω balance
Max. Output Level	+20dbu balance	+20dbu balance
Freq. Response	20~20KHz, ±1dB	20~20KHz, ±1dB
THD	Less than 0.02%	Less than 0.02%
Power Requirements	12V DC	12V DC
Dimensions	115x70x26 mm	115x70x26 mm

Dante Network High-Power Amplifier Module with DSP and POE

MODEL	POE 60
Channel	1
Input Interface	RJ-45
Cable	Cat-6 or above
Transmission Rate	100M/1000M bps
Rated Power	30W@8ohm, 50W/@4ohm
Peak Power	60W@8ohm, 100W/@4ohm
Output Connector	3-pin, 5.08mm European terminal block
Freq. Response	20~20KHz
Audio Control	Volume/Mute/Limiter/EQ/NoiseGate(option)
Protection Function	Over current protection / over temperature protection / overload / load short circuit protection
Power Requirements	PoE
Operation Temperature	0 to 40 Degree
Dimensions	148x68x22 mm

Dante 2 in 2 out Transmitter Dante 2by2i/ Dante 2by2i POE



Model with “POE” comes with a POE splitter

Features:

- ▶ Ideal interface to increase the number of AUX inputs
- ▶ 2 balanced line outputs can be used simultaneously, and all audio channels are used with Dante network.
- ▶ 2 balanced AUX inputs, and also 2 outputs using phoenix connector
- ▶ CAT5/6 connector to link with Dante network and controlledby Audinate
- ▶ Small size integrated mounting bracket which can be installed in very limited space

SPECIFICATIONS	Dante 2by2i/ Dante 2by2i POE	
Input	2 balanced/unbalanced input	
Input Impedance	10K Ohm	
Max. Input Level	+2dB	
Output	2 balanced/unbalanced output	
Output Impedance	10K Ohm	
Max. Output Level	+2dB	
Freq. Response	20~20KHz, ±1dB	
THD	0.01%	
Power Requirements	12V DC	12V DC or POE
Dimensions	123x75x35 mm	

Dante 4 in 4 out Transmitter Dante 4by4i/ Dante 4by4i POE

Model with “POE” comes with a POE splitter

Features:

- ▶ Ideal interface to increase the number of AUX inputs
- ▶ 4 balanced line outputs can be used simultaneously, and all audio channels are used with Dante network.
- ▶ 4 balanced AUX inputs, and also 4 outputs using phoenix connector
- ▶ CAT5/6 connector to link with Dante network and controlledby Audinate
- ▶ Small size integrated mounting bracket which can be installed in very limited space

SPECIFICATIONS	Dante 4by4i/ Dante 4by4i POE	
Input	4 balanced/unbalanced input	
Input Impedance	10K Ohm	
Max. Input Level	+2dB	
Output	4 balanced/unbalanced output	
Output Impedance	10K Ohm	
Max. Output Level	+2dB	
Freq. Response	20~20KHz, ±1dB	
THD	0.01%	
Power Requirements	12V DC	12V DC or POE
Dimensions	123x75x35 mm	

Dante 4 in 4 out Transmitter Dante 4by4/Dante 4by4POE

Features:

- Ideal interface to increase the number of microphones or AUX inputs
- 4 balanced line outputs can be used simultaneously, and all audio channels are used with Dante network.
- 4 balanced MIC/AUX inputs, and also 4 outputs using phoenix connector
- CAT5/6 connector to link with Dante network and controlledby Audinate
- Small size integrated mounting bracket which can be installed in very limited space
- The volume of each channel can be adjusted independently.

Model with “POE” comes with a POE splitter

SPECIFICATIONS	Dante 4by4/Dante 4by4POE	
Input	4 balanced/unbalanced input	
Input Impedance	Greater than1800 Ω	
Max. Input Level	+8dbu@ 0dB gain	
Output	4 balanced/unbalanced output	
Output Impedance	200Ω balance	
Max. Output Level	+20dbu balance	
Freq. Response	20~20KHz, ±1dB	
THD	Less than 0.02%	
Power Requirements	12V DC	12V DC or POE
Dimensions	133 x 100 x 26 mm	

Dante 8-in 8-out Transmitter Dante 448

Features:

- Built-in DSP enables signal routing, EQ, limiting, noise gating and programming
- Integrated with Dante audio network port
- 4 Analog input and output;
- 4 Digital input and output for Dante network

  Support software control and programming through RS-232 port
- Built-in 8x8 matrix enables seamless mixing and routing for Analog and Digital signals.

SPECIFICATIONS	Dante 448
Input	8 balanced/unbalanced input (XLR)
Input Impedance	Greater than 2000 Ω
Max. Input Level	+4dbu@ 0dB gain
Output	8 balanced/unbalanced output (XLR)
Output Impedance	400Ω
Max. Output Level	+4dbu balance
Freq. Response	20~20KHz, ±1dB
S/N Ratio	Greater than 95dB
THD	Less than 0.05%
Power Requirements	12V DC
Dimensions	44(H) x 178(W) x 220(D) mm

DCON 2 Dante Wall Control Panel for DANTE DSP 0404

Features:

- Standard 86 back box installation
- 3.5-inch resistive touch screen
- Support TF card upgrade
- Rs485 interface
- Support scene switching, 8 input and output volume adjustment

Dante Bluetooth Wall panel

Dante Bluetooth Wall panel

Description:

Dante BW 0202 is a wall panel for converting between Bluetooth and Dante signal, achieving two-way stereo transmission. It is widely used for teleconference, audio media transmission, etc.

Bluetooth Dante

Features:

- CAT5/6 connector to link with Dante network and controlled by Audinate;
- Small size integrated mounting bracket which can be installed in very limited space;
- Adopt advanced V5.0 Bluetooth technology;
- Simple to pair bluetooth by pressing a button;
- Compatible with a variety of mobile phones, tablets, computers and other devices;
- Support PoE power supply;
- Integrated power supply and audio transmission in one body
- Convert 1 Bluetooth Audio signal to 2 Dante Network Audio channels

Model	Dante BW 0202
Dante Audio Channel	2
Bluetooth Interface	Bluetooth 5.0, stereo input and output
Dante Interface	RJ45 interface, stereo input and output
Sampling Rate	48KHz, 24bit
Noise floor	-90dB
Power consumption	2W
THD	Less than 0.005%
Freq. Response	20~20KHz
Power Requirements	POE
Operating temperature	-10℃- 40℃
Working humidity	5-95%
Dimensions	123.5×79×35mm
Net weight	156g

Dante Bluetooth Wall panel

Bluetooth Dante

Description:

Dante BW 0404 is a wall panel for converting among analog, Bluetooth and Dante signal. It is widely used for teleconference, audio long-distance transmission and other applications.

Features:

- CAT5/6 connector to link with Dante network and controlled by Audinate;
- Small size integrated mounting bracket which can be installed in very limited space;
- Adopt advanced V5.0 Bluetooth technology;
- Simple to pair Bluetooth by pressing a button;
- Compatible with a variety of mobile phones, tablets, computers and other devices;
- With RCA and 3.5mm TRS stereo input, TRS priority;
- With one 3.5mm TRS stereo output;
- Support PoE power supply;
- Integrated power supply and audio transmission in one body;
- With Bluetooth call bridge function for conference purposes
- Convert 1 Bluetooth Audio signal to 2 Dante network audio channel;
- Convert 1 aux or 3.5 jack stereo to 2 Dante network audio channel.
- Receiving Dante network audio signal via local 3.5 jack

Model	Dante BW 0404
Dante Audio Channel	4
Input Interface	Bluetooth 5.0, stereo, RCA*2, 3.5mm TRS*1
Output Interface	Bluetooth 5.0, stereo, 3.5mm TRS*1
SNR	>100dB
Noise floor	-90dB
Power consumption	2W
THD	Less than 0.005%
Freq. Response	20~20KHz
Power Requirements	POE
Operating temperature	-10℃ - 40℃
Working humidity	5-95%
Dimensions	123.5×109.1×30mm

Dante Panel 2 Dante Wall Panel 2*2

Description:

Dante Panel series wall-mounted interface machines boast analog input and Dante digital input, analog line output and Dante digital output. The analog inputs support phantom power and manual preamp-gain adjustment. It is suitable for conference room and other professional applications. It available in black and white color.

Model	Dante Panel 2
Sampling	48k@24bit
Phantom power	DC 48V
Frequency response	20Hz-20KHz, ±0.7
THD+N	≤0.005%@1k, ±0.7
Digital-to-analog/analog-to digital dynamic range(A-weighted)	105dB
Input/output impedance(balance-type)	20kΩ/100Ω
EIN(A-weighted)	≤-126dBu
Channel isolation	100dB@1kHz
Common mode rejection (CMR)	80dB@80Hz
Max. input/output level	8dBu/12dBu
Background noise	-93dBu
System delay	≤3ms
Operating power supply	PoE
Operating temperature	0-40℃

8 In 8 out DSP audio processor with Dante DSP-880ND

Description:

The DSP-880ND audio processor combines Auto Mixer, Matrix Mixer, Feedback suppressor and Dante audio networking in one purpose-built DSP processor. The DSP-880ND includes 8 mic/line analog inputs, 8 analog outputs to meet the needs of a wide range of audio processing applications such as boardroom, medium/large conference rooms, courtrooms, auditoriums and multi-purpose halls.

Features:

- 8-channel analog input and 8-channel analog output, Microphone or Line in switch
- +48V switch per input channel
- AFC in each input
- AUTO-MIX and MATRIC-MIX
- 31 PEQS per input and 10 PEQS per output
- USB free driver connect, RS232/485 connector for center control

SPECIFICATIONS	DSP-880ND
Input Channels	8 balanced. Mic/line level
Connectors	3.81 mm Phoenix Contact, 12-pin
Input Impedance	11.5KΩ
Maximum Input Level	14dBu/Line, -7dBu/Mic
Phantom Power	+48VDC, 10mA, selectable per input
Output Channels	8 balanced, line level
Output Impedance	150Ω
Frequency Response	20Hz-20kHz(+-0.5dB)/Line 20Hz-20kHz(+-1.5dB)/Mic
THD+N	-90dB(@12dBu,1kHz,A-wt)/Line -86dB(@-7dBu,1kHz,A-wt)/Mic
SNR	105dB(@12dBu,1kHz,A-wt)/Line 95dB(@-7dBu,1kHz,A-wt)/Mic
USB	Micro-B type, free driver
RS232	Serial port communication
TCPIP	RJ-45, Optional
LED Status Indicators	Power, Link, +48V, Input and Output Audio signal
Mains Voltage	AC100V~~~240V 50/60 Hz
Dimensions	44mmx483mmx210mm
Net Weight	2.8kg
Operating Temperature	-20℃~80℃
Cooling System	Active, Fan inside
Integrated DSP	
Signal Processor	Signal Processor32-bit fixed/floating-point DSP 300MHz
Audio Latency	Audio Latency< 1ms
A/D and D/A Converters	A/D and D/A Converters24-bit
Sample Rate	48KHz

8 In 8 out DSP audio processor DSP-880N

Description:

The DSP-880N audio processor combines Auto Mixer, Matrix Mixer, and Feedback suppressor in one purpose-built DSP processor. The DSP-880N includes 8 mic/line analog inputs, 8 analog outputs to meet the needs of a wide range of audio processing applications such as boardroom, medium/large conference rooms, courtrooms, auditoriums and multi-purpose halls.

Features:

- 8-channel analog input and 8-channel analog output, Microphone or Line in switch
- +48V switch per input channel
- AFC in each input
- AUTO-MIX and MATRIC-MIX
- 31 PEQS per input and 10 PEQS per output
- USB free driver connect, RS232/485 connector for center control

SPECIFICATIONS	DSP-880N
Input Channels	8 balanced. Mic/line level
Connectors	3.81 mm Phoenix Contact, 12-pin
Input Impedance	11.5KΩ
Maximum Input Level	14dBu/Line, -7dBu/Mic
Phantom Power	+48VDC, 10mA, selectable per input
Output Channels	8 balanced, line level
Output Impedance	150Ω
Frequency Response	20Hz-20kHz(+-0.5dB)/Line 20Hz-20kHz(+-1.5dB)/Mic
THD+N	-90dB(@12dBu,1kHz,A-wt)/Line -86dB(@-7dBu,1kHz,A-wt)/Mic
SNR	105dB(@12dBu,1kHz,A-wt)/Line 95dB(@-7dBu,1kHz,A-wt)/Mic
USB	Micro-B type, free driver
RS232	Serial port communication
TCPIP	RJ-45, Optional
LED Status Indicators	Power, Link, +48V, Input and Output Audio signal
Mains Voltage	AC100V~~~240V 50/60 Hz
Dimensions	44mmx483mmx210mm
Net Weight	2.8kg
Operating Temperature	-20℃~80℃
Cooling System	Active, Fan inside
Integrated DSP	
Signal Processor	Signal Processor32-bit fixed/floating-point DSP 300MHz
Audio Latency	Audio Latency< 1ms
A/D and D/A Converters	A/D and D/A Converters24-bit
Sample Rate	48KHz

16 In 16 out DSP audio processor with Dante DSP-1616ND

Description:

The DSP-1616ND audio processor combines Auto Mixer, Matrix Mixer, Feedback suppressor and Dante audio networking in one purpose-built DSP processor. The DSP-1616ND includes 16 mic/line analog inputs, 16 analog outputs to meet the needs of a wide range of audio processing applications such as boardroom, medium/large conference rooms, courtrooms, auditoriums and multi-purpose halls.

Features:

- 16-channel analog input and 16-channel analog output, Microphone or Line in switch
- +48V switch per input channel
- AFC in each input
- AUTO-MIX and MATRIC-MIX
- 31 PEQS per input and 10 PEQS per output
- USB free driver connect, RS232/485 connector for center control

SPECIFICATIONS	DSP-1616ND
Input Channels	16 balanced. Mic/line level
Connectors	3.81 mm Phoenix Contact, 12-pin
Input Impedance	11.5KΩ
Maximum Input Level	14dBu/Line, -7dBu/Mic
Phantom Power	+48VDC, 10mA, selectable per input
Output Channels	16 balanced, line level
Output Impedance	150Ω
Frequency Response	20Hz-20kHz(+-0.5dB)/Line 20Hz-20kHz(+-1.5dB)/Mic
THD+N	-90dB(@12dBu,1kHz,A-wt)/Line -86dB(@-7dBu,1kHz,A-wt)/Mic
SNR	105dB(@12dBu,1kHz,A-wt)/Line 95dB(@-7dBu,1kHz,A-wt)/Mic
USB	Micro-B type, free driver
RS232	Serial port communication
TCPIP	RJ-45, Optional
LED Status Indicators	Power, Link, +48V, Input and Output Audio signal
Mains Voltage	AC100V~~~240V 50/60 Hz
Dimensions	44mmx483mmx210mm
Net Weight	2.8kg
Operating Temperature	-20℃~~80℃
Cooling System	Active, Fan inside
Integrated DSP	
Signal Processor	Signal Processor32-bit fixed/floating-point DSP 300MHz
Audio Latency	Audio Latency< 1ms
A/D and D/A Converters	A/D and D/A Converters24-bit
Sample Rate	48KHz

16 In 16 out DSP audio processor DSP-1616N

Description:

The DSP-1616N audio processor combines Auto Mixer, Matrix Mixer, and Feedback suppressor in one purpose-built DSP processor. The DSP-1616N includes 16 mic/line analog inputs, 16 analog outputs to meet the needs of a wide range of audio processing applications such as boardroom, medium/large conference rooms, courtrooms, auditoriums and multi-purpose halls.

Features:

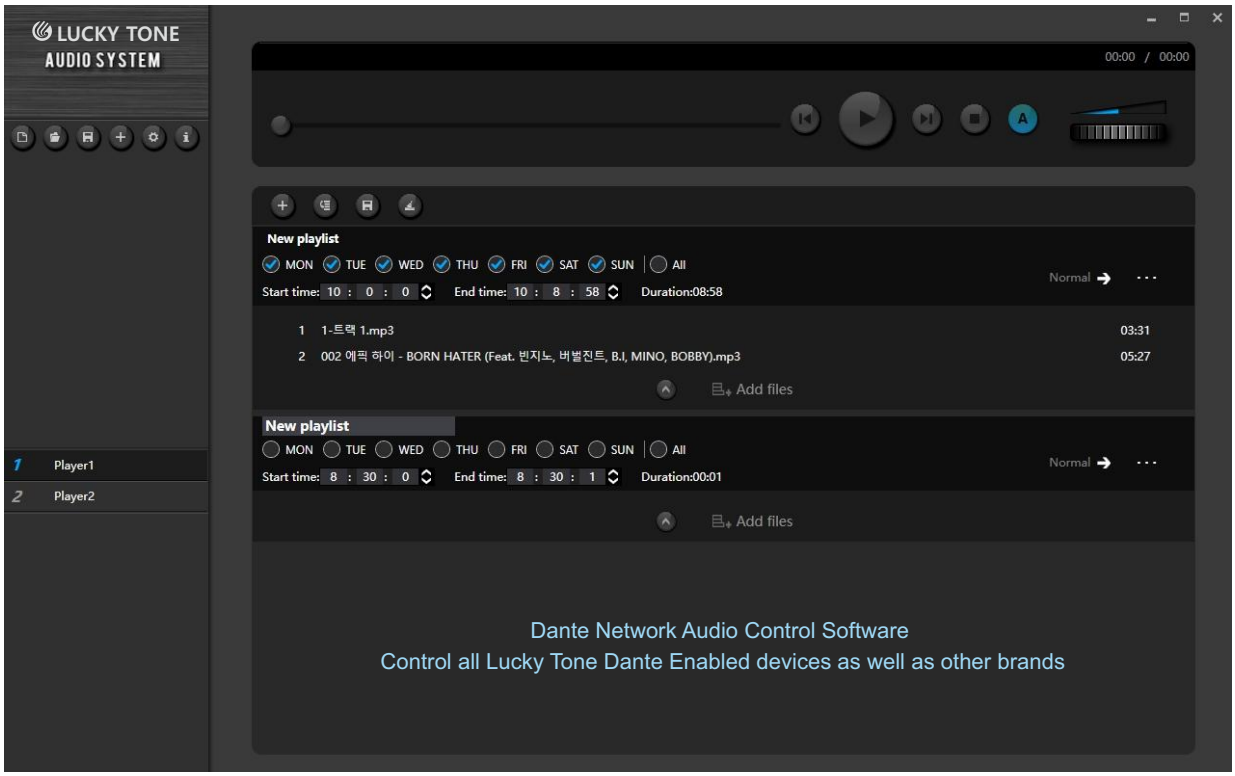
- 16-channel analog input and 16-channel analog output, Microphone or Line in switch
- +48V switch per input channel
- AFC in each input
- AUTO-MIX and MATRIC-MIX
- 31 PEQS per input and 10 PEQS per output
- USB free driver connect, RS232/485 connector for center control

SPECIFICATIONS	DSP-1616N
Input Channels	16 balanced. Mic/line level
Connectors	3.81 mm Phoenix Contact, 12-pin
Input Impedance	11.5KΩ
Maximum Input Level	14dBu/Line, -7dBu/Mic
Phantom Power	+48VDC, 10mA, selectable per input
Output Channels	16 balanced, line level
Output Impedance	150Ω
Frequency Response	20Hz-20kHz(+-0.5dB)/Line 20Hz-20kHz(+-1.5dB)/Mic
THD+N	-90dB(@12dBu,1kHz,A-wt)/Line -86dB(@-7dBu,1kHz,A-wt)/Mic
SNR	105dB(@12dBu,1kHz,A-wt)/Line 95dB(@-7dBu,1kHz,A-wt)/Mic
USB	Micro-B type, free driver
RS232	Serial port communication
TCPIP	RJ-45, Optional
LED Status Indicators	Power, Link, +48V, Input and Output Audio signal
Mains Voltage	AC100V~~~240V 50/60 Hz
Dimensions	44mmx483mmx210mm
Net Weight	2.8kg
Operating Temperature	-20℃~~80℃
Cooling System	Active, Fan inside
Integrated DSP	
Signal Processor	Signal Processor32-bit fixed/floating-point DSP 300MHz
Audio Latency	Audio Latency< 1ms
A/D and D/A Converters	A/D and D/A Converters24-bit
Sample Rate	48KHz

8 Channel Programmable Timer Dante Player8/Dante player16

Features:

- ▶ 8/16 audio channels can play simultaneously.
- ▶ Each channel supports independent scheduled broadcast and manual broadcast.
- ▶ Scheduled broadcast can be configured on daily, weekly and monthly basis. Manual broadcast supports loop mode and random mode.
- ▶ With super-fast loading speed, the brand-new playback engine does not occupy CPU for playback, and the playback of all zone is fully synchronized.
- ▶ Supports audio formats and video formats such as MP3, WAV, WMA, WMV, RM, RMVB, MPEG1, MPEG2, DIVX and more.
- ▶ The playback engine supports audio fade-in and fade-out effects;
- ▶ The machine prompts 30 seconds in advance to avoid accidental shutdown;
- ▶ Supports high priority broadcast override
- ▶ Supports Dante virtual sound card



4 In 2 Out Automatic DSP Audio Processor

Description:

Developed for normalized recording and broadcasting, remote interaction and conference sound reinforcement system, DSP-0402+ is a digital audio processor with adaptive feedback and “far side” speech echo cancellation. It has built-in AFC, AEC, ANS, AGC, ANC and Rs232.

Automatic DSP Matrix, No adjustment is required.

MODEL	DSP-0402+
Input Channels(analog)	4
Output Channels(analog)	2
Processor	ADI SHARC 21489@450 MHz SIMD
DSP Handling Capacity	400 MIPS, 1.6 GFLOPS
Sampling Rate	48K/24bit
Frequency Response	(20~20kHz):±0.3dB
THD+N	<-94dB @17dBu
Input Dynamic Range	110dB
Output Dynamic Range	112dB
Input gain	0- 50dBu
Phantom Power	+48V/10mA max
Maximum level	+18dBu
Channel isolation @1kHz	108dB
Input Impedance (balanced connection)	5.4KΩ
Output Impedance (balanced connection)	600Ω
System Latency	<3ms
Power Supply	DC12V
Dimension	215x182x44mm
Weight	1.2KG

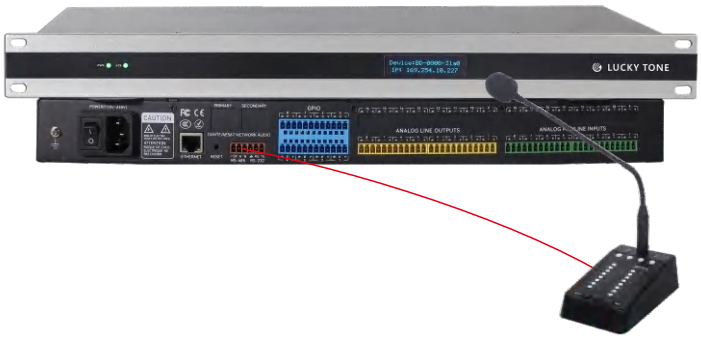
4 In 4 Out Configurable Digital Audio Processor

Features:

- Built-in 2 in 2 out USB sound card, supporting music playback, recording and soft video conferencing, such as: ZOOM, Tencent Meeting, DingTalk Meeting, etc;
- Supports AFC (feedback suppression), AEC (echo cancellation), ANS (noise suppression), ANC (noise gain compensation), AGC (automatic gain), gain sharing, threshold automatic mixing.
- 8-band equalizer providing 5 filter options: Parametric, Lowshelf, Highshelf, Lowpass, Highpass.
- Provides customizable user interface, and support up to 30 devices to be managed under the same UI.
- Supports RS232, RS485 and UDP central control.

Model	DSP-440ND+
Processor	ADI SHARC 21489@450 MHz SIMD
DSP processing power	400 MIPS, 1.6 GFLOPS
Sample Rate	48K/24bit
Number of analog input and output channels	4x4
Number of Dante input and output channels	4x4
Input Gain	0/6/12/18/24/30/36/42/48 dB
Phantom Power	+48V/10mA max
Frequency Response	20~20kHz: ±0.5dB
Max. Level	+18dBu
THD+N	<-100dB @4dBu
Input Dynamic Range	110dB
Output Dynamic Range	112dB
Channel Isolation @1kHz	108dB
Input Impedance (balanced connection)	5.4KΩ
Output Impedance (balanced connection)	600Ω
System Latency	<3ms
Power Supply	DC12V/PoE48V
Dimension	215x162x44mm
Weight	2KG

8 In 8 out DSP Audio Processor



Description:

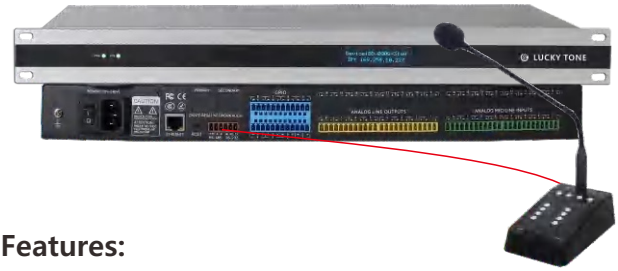
DSP-880N2+ uses the industry-leading chip ADI DSP SHARC 21489, and the highest operating frequency can reach 450M Hz.
The input and output modules of DSP can be customized according to the site conditions. At the same time, it has independent AFC/ AEC/ ANS/ AGC/ gain sharing automatic mixing, threshold automatic mixing and other processing modules.
Meet the needs of audio processors and transmission in various places, such as conference rooms, multi-function halls, conference centers, auditoriums, administrative centers, etc.

Features:

- With industry-leading chip ADI SHARC 21489 , the working frequency can reach 450M Hz;
- Customizing operation software makes the configuration more flexible, and it can control Different DSP.
- Provides operation interface for customers to realize centralized control of multiple devices. And it can control third-party's equipment through DUP RS232, RS485;
  AFC (feedback suppression), AEC (echo cancellation), ANS (noise suppression), ANC (noise gain compensation), AGC (automatic gain), gain sharing, threshold automatic mixing, dodger and other processing modules;
  Each channel has independent adaptive feedback suppression, automatically find the feedback point, and automatically suppress;
- Full-featured matrix mixing, the input mixing level can be adjusted;
- There are 16 presets, each preset works independently;
- 8 GPIOs can independently configure with input or output, and they can be used as independent ADC when configuring with input;
- Support channel copy, LINK and group functions;
- Support RS232&UDP central control, UDP port can be set freely, and you can check the control software code;
- 2 types of wall panels are optional, and they are available to software programming.A

SPECIFICATIONS	DSP-880N2+
Input Channels(analog)	8
Output Channels(analog)	8
Input	5/ 8/ 12 section PEQ (optional), AFC/AEC/ANS/AGC/Auto Mixer
Output	Frequency divider: 5, 8, 12 segment PEQ (optional); Delayer; Limiter
Input gain	0/10/20/30/40/43 dB
THD+N	0.003% @4dBu
Frequency response	20~20kHz ±0.2dB
Maximum level	+24dBu
Phantom power	+48V
Dynamic range (analog channel)	113dB
Self-Noise (A-Weighting-analog)	-89dBu
Common mode rejection ratio @60Hz	80dB
Channel isolation @1kHz	108dB
Input impedance (balanced connection)	9.4KΩ
Output impedance (balanced connection)	102Ω
System delay	<3ms
Power supply	AC110~240V,50Hz/60Hz
Dimension	482 x 260 x 45mm
Weight	3kg

8 In 8 out DSP Dante Audio Processor



Description:

DSP-880ND2+ uses the industry-leading chip ADI DSP SHARC 21489, and the highest operating frequency can reach 450M Hz. The input and output modules of DSP can be customized according to the site conditions. At the same time, it has independent AFC/ AEC/ ANS/ AGC/ gain sharing automatic mixing, threshold automatic mixing and other processing modules.
Meet the needs of audio processors and transmission in various places, such as conference rooms, multi-function halls, conference centers, auditoriums, administrative centers, etc.

Features:

- With industry-leading chip ADI SHARC 21489 , the working frequency can reach 450M Hz;
- Customizing operation software makes the configuration more flexible, and it can control Different DSP.
- Provides operation interface for customers to realize centralized control of multiple devices. And it can control third-party's equipment through DUP RS232, RS485;
- AFC (feedback suppression), AEC (echo cancellation), ANS (noise suppression), ANC (noise gain compensation), AGC (automatic gain), gain sharing, threshold automatic mixing, dodger and other processing modules;
- Each channel has independent adaptive feedback suppression, automatically find the feedback point, and automatically suppress;
- Full-featured matrix mixing, the input mixing level can be adjusted;
- There are 16 presets, each preset works independently;
- 8 GPIOs can independently configure with input or output, and they can be used as independent ADC when configuring with input;
- Support channel copy, LINK and group functions;
- Support RS232&UDP central control, UDP port can be set freely, and you can check the control software code;
- 2 types of wall panels are optional, and they are available to software programming.

SPECIFICATIONS	DSP-880ND2+
Input Channels(analog)	8
Output Channels(analog)	8
Dante audio channel	Dante Ultimo
Input	5/ 8/ 12 section PEQ (optional), AFC/AEC/ANS/AGC/Auto Mixer
Output: Frequency divider	Frequency divider: 5, 8, 12 segment PEQ (optional); Delayer; Limiter
Input gain	0/10/20/30/40/43 dB
THD+N	0.003% @4dBu
Frequency response	20~20kHz ±0.2dB
Maximum level	+24dBu
Phantom power	+48V
Dynamic range (analog channel)	113dB
Dynamic range (Dante channel)	115dB
Self-Noise (A-Weighting-analog)	-89dBu
Self-Noise (A-Weighting-Dante)	-91dBu
Common mode rejection ratio @60Hz	80dB
Channel isolation @1kHz	108dB
Input impedance (balanced connection)	9.4KΩ
Output impedance (balanced connection)	102Ω
System delay	<3ms
Power supply	AC110~240V,50Hz/60Hz
Dimension	482 x 260 x 45mm
Weight	3kg

8 Zone Paing Microphone Station

Features:

8 ZONE Remote Microphone Station for DSP-880N+,DSP-880M+,DSP-880ND+

- Controls
  Zone selector switches for Z1~Z8. Talk & Chime on/off.
  Level control volume for MIC & Chime.
- Connectors
  RS485 communication with DSP-880N+,DSP-880M+,DSP-880ND+.
  DC input.
- Indicators
  Power, Busy, Signal, Clip.

MODEL	DSP M8 _
Rated Output Level / Impedance	0dB / 600 Ω Balanced
Input sensitivity / Impedance	-36dB / 20kΩ Balanced
Frequency Response (-1dB)	80Hz - 15kHz
T.H.D	< 0.1%
Hum & Noise	> 90 dB
Number of Zone	8 Zones
Weight (Net)	1.2 kg
Dimensions (W x H x D)	115 x 60 x 195mm (only body)

12 In 8 Out DSP Audio Processor

Features:

- AI machining learning can intelligently eliminate environmental noises such as table knock, page turn, fan noise, etc., to realize the separation of non-human voice signals.
- Machining learning (100,000 hours): the number of network nodes can be dynamically adjusted, achieving online learning and featuring low latency.
- The processing module is configurable and can be freely replaced.
- Built-in one-in-one-out USB sound card, supports music playback, recording and video conferences (such as ZOOM, Tencent conference, etc.);
- Supports AFC (feedback suppression), AEC (echo cancellation), ANS (noise suppression), Ducker, ANC (noise gain compensation), AGC (automatic gain), gain sharing, threshold automatic mixing.
- 12-band equalizer providing 5 filter options: Parametric, Lowshelf, Highshelf, Lowpass, Highpass.
- Provides customizable user interface, and support up to 30 devices to be managed under the same UI.
- Support camera control
- 8 GPIOs
- Supports RS232, RS485 and UDP central control.

MODEL	DSP-1280N2+AI
Input Channels(analog)	12
Output Channels(analog)	8
Processor	ADI SHARC 21489@450 MHz SIMD
DSP Handling Capacity	400 MIPS, 1.6 GFLOPS
Sampling Rate	48K/40bit
Frequency Response	(20~20kHz):±0.3dB
THD+N	<-94dB @17dBu
Input Dynamic Range	110dB
Output Dynamic Range	112dB
Input gain	0/3/6/9/12/15/18/21/24/27/30/33/36/39/42/45/48 dBu
Phantom Power	+48V/10mA max
Maximum level	+18dBu
Channel isolation @1kHz	108dB
Input Impedance (balanced connection)	5.4KΩ
Output Impedance (balanced connection)	600Ω
System Latency	<3ms
Power Supply	AC110~240V,50Hz/60Hz
Dimension	482 x 260 x 45mm
Weight	4KG

16 In 16 out DSP Audio Processor

Description:

DSP-1616N2+ uses the industry-leading chip ADI DSP SHARC 21489, and the highest operating frequency can reach 450M Hz.
The input and output modules of DSP can be customized according to the site conditions. At the same time, it has independent AFC/ AEC/ ANS/ AGC/ gain sharing automatic mixing, threshold automatic mixing and other processing modules. Meet the needs of audio processors and transmission in various places, such as conference rooms, multi-function halls, conference centers, auditoriums, administrative centers, etc.

Features:

- With industry-leading chip ADI SHARC 21489 , the working frequency can reach 450M Hz;
- Customizing operation software makes the configuration more flexible, and it can control Different DSP.
- Provides operation interface for customers to realize centralized control of multiple devices. And it can control third-party's equipment through DUP RS232, RS485;
- AFC (feedback suppression), AEC (echo cancellation), ANS (noise suppression), ANC (noise gain compensation), AGC (automatic gain), gain sharing, threshold automatic mixing, dodger and other processing modules;
- Each channel has independent adaptive feedback suppression, automatically find the feedback point, and automatically suppress;
- Full-featured matrix mixing, the input mixing level can be adjusted;
- There are 16 presets, each preset works independently;
- 8 GPIOs can independently configure with input or output, and they can be used as independent ADC when configuring with input;
- Support channel copy, LINK and group functions;
- Support RS232&UDP central control, UDP port can be set freely, and you can check the control software code;
- 2 types of wall panels are optional, and they are available to software programming.

SPECIFICATIONS	DSP-1616N2+
Input Channels(analog)	16
Output Channels(analog)	16
Input	5/ 8/ 12 section PEQ (optional), AFC/AEC/ANS/AGC/Auto Mixer
Output	Frequency divider: 5, 8, 12 segment PEQ (optional); Delayer; Limiter
Input gain	0/10/20/30/40/43 dB
THD+N	0.003% @4dBu
Frequency response	20~20kHz ±0.2dB
Maximum level	+24dBu
Phantom power	+48V
Dynamic range (analog channel)	113dB
Dynamic range (Dante channel)	115dB
Self-Noise (A-Weighting-analog)	-89dBu
Self-Noise (A-Weighting-Dante)	-91dBu
Common mode rejection ratio @60Hz	80dB
Channel isolation @1kHz	108dB
Input impedance (balanced connection)	9.4KΩ
Output impedance (balanced connection)	102Ω
System delay	<6ms
Power supply	AC110~240V,50Hz/60Hz
Dimension	482 x 260 x 45mm
Weight	3kg

8 In 8 out DSP Audio Processor

Description:

DSP-880M2+ uses the industry-leading chip ADI DSP SHARC 21489, and the highest operating frequency can reach 450M Hz.
The input and output modules of DSP can be customized according to the site conditions. At the same time, it has independent AFC/ AEC/ ANS/ AGC/ gain sharing automatic mixing, threshold automatic mixing and other processing modules.
Meet the needs of audio processors and transmission in various places, such as conference rooms, multi-function halls, conference centers, auditoriums, administrative centers, etc.

Features:

- With industry-leading chip ADI SHARC 21489 , the working frequency can reach 450M Hz;
- Customizing operation software makes the configuration more flexible, and it can control Different DSP.
- Provides operation interface for customers to realize centralized control of multiple devices. And it can control third-party's equipment through DUP RS232, RS485;
  AFC (feedback suppression), AEC (echo cancellation), ANS (noise suppression), ANC (noise gain compensation), AGC (automatic gain), gain sharing, threshold automatic mixing, dodger and other processing modules;
  Each channel has independent adaptive feedback suppression, automatically find the feedback point, and automatically suppress;
- Full-featured matrix mixing, the input mixing level can be adjusted;
- There are 16 presets, each preset works independently;
- 8 GPIOs can independently configure with input or output, and they can be used as independent ADC when configuring with input;
- Support channel copy, LINK and group functions;
- Support RS232&UDP central control, UDP port can be set freely, and you can check the control software code;
- 2 types of wall panels are optional, and they are available to software programming.A

DSP-880M2+/DSP-440M2+
(Simple version without Ducker and Camera control)

SPECIFICATIONS	DSP-880M2+/DSP-440M2+
Input Channels(analog)	8/4
Output Channels(analog)	8/4
Input	5/ 8/ 12 section PEQ (optional), AFC/AEC/ANS/AGC/Auto Mixer
Output	Frequency divider: 5, 8, 12 segment PEQ (optional); Delayer; Limiter
Input gain	0/10/20/30/40/43 dB
THD+N	0.003% @4dBu
Frequency response	20~20kHz ±0.2dB
Maximum level	+24dBu
Phantom power	+48V
Dynamic range (analog channel)	113dB
Self-Noise (A-Weighting-analog)	-89dBu
Common mode rejection ratio @60Hz	80dB
Channel isolation @1kHz	108dB
Input impedance (balanced connection)	9.4KΩ
Output impedance (balanced connection)	102Ω
System delay	<3ms
Power supply	AC110~240V,50Hz/60Hz
Dimension	482 x 260 x 45mm
Weight	3kg

16 In 16 out DSP Audio Processor

Description:

DSP-1616M2+ uses the industry-leading chip ADI DSP SHARC 21489, and the highest operating frequency can reach 450M Hz.
The input and output modules of DSP can be customized according to the site conditions. At the same time, it has independent AFC/ AEC/ ANS/ AGC/ gain sharing automatic mixing, threshold automatic mixing and other processing modules. Meet the needs of audio processors and transmission in various places, such as conference rooms, multi-function halls, conference centers, auditoriums, administrative centers, etc.

Features:

- With industry-leading chip ADI SHARC 21489 , the working frequency can reach 450M Hz;
- Customizing operation software makes the configuration more flexible, and it can control Different DSP.
- Provides operation interface for customers to realize centralized control of multiple devices. And it can control third-party's equipment through DUP RS232, RS485;
- AFC (feedback suppression), AEC (echo cancellation), ANS (noise suppression), ANC (noise gain compensation), AGC (automatic gain), gain sharing, threshold automatic mixing, dodger and other processing modules;
- Each channel has independent adaptive feedback suppression, automatically find the feedback point, and automatically suppress;
- Full-featured matrix mixing, the input mixing level can be adjusted;
- There are 16 presets, each preset works independently;
- 8 GPIOs can independently configure with input or output, and they can be used as independent ADC when configuring with input;
- Support channel copy, LINK and group functions;
- Support RS232&UDP central control, UDP port can be set freely, and you can check the control software code;
- 2 types of wall panels are optional, and they are available to software programming.

DSP-1616M2+(Simple version without Ducker and Camera control)

SPECIFICATIONS	DSP-1616M2+
Input Channels(analog)	16
Output Channels(analog)	16
Input	5/ 8/ 12 section PEQ (optional), AFC/AEC/ANS/AGC/Auto Mixer
Output	Frequency divider: 5, 8, 12 segment PEQ (optional); Delayer; Limiter
Input gain	0/10/20/30/40/43 dB
THD+N	0.003% @4dBu
Frequency response	20~20kHz ±0.2dB
Maximum level	+24dBu
Phantom power	+48V
Dynamic range (analog channel)	113dB
Dynamic range (Dante channel)	115dB
Self-Noise (A-Weighting-analog)	-89dBu
Self-Noise (A-Weighting-Dante)	-91dBu
Common mode rejection ratio @60Hz	80dB
Channel isolation @1kHz	108dB
Input impedance (balanced connection)	9.4KΩ
Output impedance (balanced connection)	102Ω
System delay	<6ms
Power supply	AC110~240V,50Hz/60Hz
Dimension	482 x 260 x 45mm
Weight	3kg

16 In 16 Out DSP Dante Audio Processor

Description:

DSP-1616ND2+ uses the industry-leading chip ADI DSP SHARC 21489, and the highest operating frequency can reach 450M Hz.
The input and output modules of DSP can be customized according to the site conditions. At the same time, it has independent AFC/ AEC/ ANS/ AGC/ gain sharing automatic mixing, threshold automatic mixing and other processing modules.
Meet the needs of audio processors and transmission in various places, such as conference rooms, multi-function halls, conference centers, auditoriums, administrative centers, etc.

Features:

- With industry-leading chip ADI SHARC 21489 , the working frequency can reach 450M Hz;
- Customizing operation software makes the configuration more flexible, and it can control Different DSP.
- Provides operation interface for customers to realize centralized control of multiple devices. And it can control third-party's equipment through DUP RS232, RS485;
- AFC (feedback suppression), AEC (echo cancellation), ANS (noise suppression), ANC (noise gain compensation), AGC (automatic gain), gain sharing, threshold automatic mixing, dodger and other processing modules;
- Each channel has independent adaptive feedback suppression, automatically find the feedback point, and automatically suppress;
- Full-featured matrix mixing, the input mixing level can be adjusted;
- There are 16 presets, each preset works independently;
- 8 GPIOs can independently configure with input or output, and they can be used as independent ADC when configuring with input;
- Support channel copy, LINK and group functions;
- Support RS232&UDP central control, UDP port can be set freely, and you can check the control software code;
- 2 types of wall panels are optional, and they are available to software programming.

SPECIFICATIONS	DSP-1616ND2+
Input Channels(analog)	16
Output Channels(analog)	16
Dante audio channel	Dante Ultimo
Input	5/ 8/ 12 section PEQ (optional), AFC/AEC/ANS/AGC/Auto Mixer
Output	Frequency divider: 5, 8, 12 segment PEQ (optional); Delayer; Limiter
Input gain	0/10/20/30/40/43 dB
THD+N	0.003% @4dBu
Frequency response	20~20kHz ±0.2dB
Maximum level	+24dBu
Phantom power	+48V
Dynamic range (analog channel)	113dB
Dynamic range (Dante channel)	115dB
Self-Noise (A-Weighting-analog)	-89dBu
Self-Noise (A-Weighting-Dante)	-91dBu
Common mode rejection ratio @60Hz	80dB
Channel isolation @1kHz	108dB
Input impedance (balanced connection)	9.4KΩ
Output impedance (balanced connection)	102Ω
System delay	<6ms
Power supply	AC110~240V,50Hz/60Hz
Dimension	482 x 260 x 45mm
Weight	3kg

32 In 32 out DSP Dante Audio Processor

Description:

DSP-3232ND2+ uses the industry-leading chip ADI DSP SHARC 21489, and the highest operating frequency can reach 450M Hz.
Meet the needs of audio processors and transmission in various places, such as conference rooms, multi-function halls, conference centers, auditoriums, administrative centers, etc.

Features:

Features:

- With industry-leading chip ADI SHARC 21489 , the working frequency can reach 450M Hz;
- Provides operation interface for customers to realize centralized control of multiple devices.
- UDP, RS232, RS485 can control third-party's equipment,with full-featured matrix mixing, the input mixing level can be adjusted;
- There are 16 presets, each preset works independently;
- 8 GPIOs can independently configure with input or output, and they can be used as independent ADC when configuring with input and it supports channel copy, LINK and group functions;
- Support RS232&UDP central control, UDP port can be set freely, and you can check the control software code;
- 2 types of wall panels are optional, and they are available to software programming.

SPECIFICATIONS	DSP-3232ND2+
Dante audio channel	Dante Ultimo
Input	8 section PEQ (optional), Expander, Compressor
Output	Frequency divider, 8 segment PEQ, Delayer, Limiter
Input gain	0/10/20/30/40/43 dB
THD+N	0.003% @4dBu
Frequency response	20~20kHz ±0.2dB
Maximum level	+24dBu
Phantom power	+48V
Dynamic range (analog channel)	113dB
Dynamic range (Dante channel)	115dB
Self-Noise (A-Weighting-analog)	-89dBu
Self-Noise (A-Weighting-Dante)	-91dBu
Common mode rejection ratio @60Hz	80dB
Channel isolation @1kHz	108dB
Input impedance (balanced connection)	9.4KΩ
Output impedance (balanced connection)	102Ω
System delay	<6ms
Power supply	AC110~240V,50Hz/60Hz
Dimension	482 x 260 x 45mm
Weight	3kg

64 In 64 out DSP Dante Audio Processor

Description:

DSP-6464ND2+ uses the industry-leading chip ADI DSP SHARC 21489, and the highest operating frequency can reach 450M Hz.
Meet the needs of audio processors and transmission in various places, such as conference rooms, multi-function halls, conference centers, auditoriums, administrative centers, etc.

Features:

- With industry-leading chip ADI SHARC 21489 , the working frequency can reach 450M Hz;
- Provides operation interface for customers to realize centralized control of multiple devices.
- UDP, RS232, RS485 can control third-party's equipment,with full-featured matrix mixing, the input mixing level can be adjusted;
- There are 16 presets, each preset works independently;
- 8 GPIOs can independently configure with input or output, and they can be used as independent ADC when configuring with input and it supports channel copy, LINK and group functions;
- Support RS232&UDP central control, UDP port can be set freely, and you can check the control software code;
- 2 types of wall panels are optional, and they are available to software programming.

SPECIFICATIONS	DSP-6464ND2+
Dante audio channel	Dante Ultimo
Input	8 section PEQ , Expander, Compressor
Output	Frequency divider, 8 segment PEQ, Delayer, Limiter
Input gain	0/10/20/30/40/43 dB
THD+N	0.003% @4dBu
Frequency response	20~20kHz ±0.2dB
Maximum level	+24dBu
Phantom power	+48V
Dynamic range (analog channel)	113dB
Dynamic range (Dante channel)	115dB
Self-Noise (A-Weighting-analog)	-89dBu
Self-Noise (A-Weighting-Dante)	-91dBu
Common mode rejection ratio @60Hz	80dB
Channel isolation @1kHz	108dB
Input impedance (balanced connection)	9.4KΩ
Output impedance (balanced connection)	102Ω
System delay	<6ms
Power supply	AC110~240V,50Hz/60Hz
Dimension	482 x 260 x 45mm
Weight	3kg

Open architecture Dante 64 In 64 out DSP audio processor

Description:

The DSP-6464 LD+ intelligent digital audio processing server uses two processing chips with a main frequency of 1GHz, which is the core device of the thermal computing system platform. The thermal computing system platform is based on the real-time Linux operating system. All input channels can be configured with AEC cards to build a 96-channel AEC input channel system. The system server supports 64*64 network transmission, and the control function is intuitive. It is suitable for various applications such as medium and large-scale conference applications, theme parks, airports, railway stations, and large supermarkets.

Features:

- Gigabit network transmission, with network transmission backup
- 8 AEC channels, and supports user-defined addition of independent AEC modules, Support up to 32 channels
- Dual host backup (Active/Standby) mode
- 16-channel audio player, 64G memory, playback format WAV MP3 supports scheduled playback
- Provides operation interface for customers to realize centralized control of multiple devices
- Dual power redundancy (AC/DC)

SPECIFICATIONS	DSP-6464 LD+
Network channel	64*64
Audio I/O capability	8 audio I/O card slots
Multitrack player capability	16 tracks
Media storage capacity	64G
Dynamic range	>118dB
Frequency Response (±0.2dB)	20Hz~20kHz
Input Impedance	5.5k ohm
THD+N	<0.002%
Maximum input level	+22dBu
RS232/485	6Pin3.81mm Phoenix
GPIO	12Pin381mm Phoenix
Dante Primary	RJ45 1000Mbps
Dante Secondary	RJ45 1000Mbps
AC Main Power	IEC Connector
DC backup power supply	24VDC 2A 2pin 5.81mmm Phoenix
Voltage	220VAC-240VAC, 50Hz/24VDC
Current	4A max@100VAC (actual current depends on specific configuration, such as I/O card
Product size (WxHxD)	483x88x364mm
Shipping Package Dimensions	618x153x473mm
Shipping weight	10KG

Open architecture Dante 32 In 32 out DSP audio processor

Description:

The DSP-3232 LD+intelligent digital audio processing server adopts two processing chips with a main frequency of 1GHz, which is the core device of the thermal computing system platform. The server has network backup and dual-host backup functions. Meet the needs of audio processors and transmission in various places, such as conference rooms, multi-function halls, conference centers, auditoriums, administrative centers, etc.

Features:

- Gigabit network transmission, network transmission backup
- Independent AEC modules can be added, Support up to 32 channels
- Dual host warm backup (Active/Standby) mode
- Player: 16-channel audio player, 64G memory, WAV/MP3 with scheduled playback
- Provides operation interface for customers to realize centralized control of multiple devices.
- Dual Power Redundancy (AC/DC)

SPECIFICATIONS	DSP-3232 LD+
Network Channel	32*32
Audio I/O capability	8 audio I/O card slots
Multi-track player capacity	16 tracks
Media storage capacity	64G
GPIO	12Pin 3.81mm Phoenix Block
Dante Primary	RJ45 1000Mbps
Dante Secondary	RJ45 1000Mbps
AC main power	IEC connector
DC standby power	24VDC 2A 2pin 5.81mm
Voltage	220VAC-240VAC
Product Dimensions (WxHxD)	483x88x364mm
Shipping package size	618x153x473mm
Shipping weight	10KG

4 Channel Output module for Open architecture DSP-6464LD+ or DSP-3232LD+

Description:

The analog output card provides 4-channel line output. DSP-AO4 LD+ employs electronically balanced outputs and plug-in connectors. Control functions for each output include: gain level and mute control.

Features:

- 4 balanced line outputs
- Easy for installation
- Control and configure via software

SPECIFICATIONS	DSP-AO4 LD+
Dynamic range	>118dB
Frequency response (±0.2dB)	20Hz~20kHz
Output impedance	102 Ω
Channel Crosstalk	<-112dB
THD+N	<0.002%
Maximum output level (@1% distortion)	+22dBu
Interface	four 3-pin Euro breakaway terminals

Open architecture Dante 8 In 8 out DSP audio processor

Description:

The input and output modules of DSP can be customized according to the site conditions. At the same time, it has independent AFC/ AEC/ ANS/ AGC/ gain sharing automatic mixing, threshold automatic mixing and other processing modules. Meet the needs of audio processors and transmission in various places, such as conference rooms, multi-function halls, conference centers, auditoriums, administrative centers, etc.

Features:

- Customizing operation software makes the configuration more flexible, and it can control Different DSP.
- Provides operation interface for customers to realize centralized control of multiple devices. And it can control third-party's equipment through DUP RS232, Rs485;
- AFC (feedback suppression), AEC (echo cancellation), ANS (noise suppression), ANC (noise gain compensation), AGC (automatic gain), gain sharing, threshold automatic mixing, dodger and other processing modules;
- 8 GPIOs can independently configure with input or output, and they can be used as independent ADC when configuring with input;
- Support RS232&UDP central control, UDP port can be set freely, and you can check the control software code
- High-quality 24bit A/D and D/A converter
- Dual network backup
- Dual Power Redundancy (AC/DC)

SPECIFICATIONS	DSP-880LD2+
Type	Network interface, D/A conversion
AEC	Built-in 4 AEC channels
Channels	8 channels mic (with phantom power)/ Line input, 8 channels line output
Network port	2 Gigabit Ethernet ports
Control ports	8 GPIO channels and 1 general purpose serial port Rs232
Frequency Response	20Hz~20kHz
Frequency of sample	48kHz
Dynamic Range	118dB
THD+N	<0.002%
Phantom Power	+48V DC 10mA
Gain	6dB/step (0db-45dB)
AD/DA	123dB
OPA(operational amplifier)	113 dB
Input Impedance (Balanced)	8kΩ
Output impedance (balanced)	207Ω
SNR	> 90dB
Maximum input and output level	+20dBu/10dBv

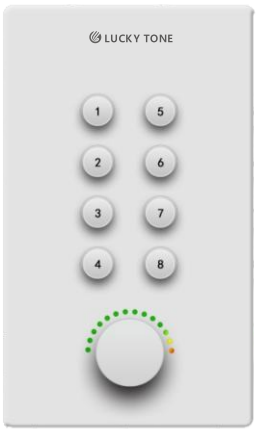
Technical parameters	Audio information processing server		Processing interface machine
	DSP-6464 LD+	DSP-3232 LD+	DSP-880 LD+
DSP type	Drag-and-drop DSP processing architecture	Drag-and-drop DSP processing architecture	Drag-and-drop DSP processing architecture
Number of card slots	8	8	\
Built-in AEC channel	0	0	4
Max AEC processing power	64	32	4
Network audio channel	64x64	32x32	8x8
GPIO	8x8	8x8	8
RTA real-time analyzer	√	√	\
Dante I/O channel	√	√	√
Expandable I/O interface box	√	√	√
IOS control interface	√	√	√
PC computer control	√	√	√
Logic control	√	√	√

Technical parameters	Input Interface Card	Input Interface Card	AEC Input Interface Card
	DSP-AI4 LD+	DSP-AO4 LD+	DSP-AEC4 LD+
Number of channels	4ch	4ch	4ch
Frequency influence	20Hz~20kHz	20Hz~20kHz	20Hz~20kHz
Phantom power	48VDC	-	48VDC
Dynamic range	>118dB	>118dB	>118dB
Total harmonic distortion	<0.002%	<0.002%	<0.002%

DCON-7 Dante Wall Control Panel

Features:

- ▶ There are eight buttons and one knob;
- ▶ Support customized functions
- ▶ Use UDP communication protocol;
- ▶ POE power supply.



Model	DCON-7
Programmable buttons	8 (support user-defined)
Control knob	1
Control protocol	UDP communication protocol
Address setting method	TCP/IP
Signal cable length	100 meters
Power supply	POE power supply
Connection port	RJ-45
Power consumption	<100mw
Color	White
Material	Aluminum alloy sandblasting

DCON-8 Dante Wall Control Panel

Features:

- ▶ There is one knob.
- ▶ Support customized functions.
- ▶ Use UDP communication protocol.
- ▶ POE power supply.
- ▶ Support 32 menus.

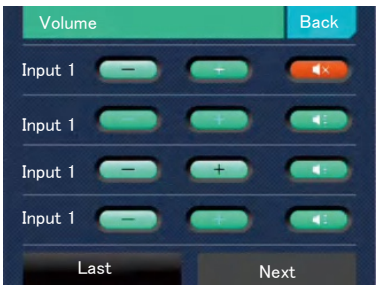
Model	DCON-8
Control knob	1
Control protocol	UDP communication protocol
Address setting method	TCP/IP
Signal cable length	100 meters
Power supply	POE power supply
Connection port	RJ-45
Power consumption	<100mw
Color	White
Material	Aluminum alloy sandblasting

Dante Wall Control Panel

DCON 3 Dante Wall Control Panel For DSP-880N, 880ND, 1616N, 1616ND
DCON 6 Dante Wall Control Panel For DSP 880N+, 880ND+, and 880M+

Features:

- Standard 86 back box installation
- 4 inch Capacitive touch screen
- Support TF card upgrade
- Rs485 interface
- Support scene switching, 8 input and output volume adjustment



Dante DSP Controller in audio and video conference