LUCKY TONE

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
- Built-in Lua script, providing flexible extension and customization functions for users
- Dual power redundancy (AC/DC)

SPECIFICATIONS	DSP-6464 LD+
Network channel	64*64
Local channel	two 4-channel AEC input cards, microphone / line input
Audio I/O capability	6 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

LUCKY TONE

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. The system has built-in Lua script language, which can be written in standard C language. 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 5.81mmm phoenix holder
Voltage	220VAC-240VAC50Hz/24VDC
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

4 Channel MIC/Line 48V phantom Power Input module for Open architecture DSP-6464LD+ or DSP-3232LD+

Description:

The analog input card provides functions, including 4-channel mic/line input, +48V phantom power, 84dB gain level control and 58dB Mic Amp. DSP-AI4 LD+ adopts plug-in balanced inputs. Control functions for each input include: gain level control, Mic Amp, mute, and signal inversion.

Features:

- 4 mic inputs/line inputs
- +48V phantom power
- Dynamic range of 118dB
- Easy for installation
- Control and configure via software

SPECIFICATIONS	DSP-AI4 LD+
Dynamic range	＞118dB
Frequency response (±0.2dB)	20Hz~20kHz
Input impedance	5.5k Ω
Channel Crosstalk	<-112dB
THD+N	<0.002%
Common Mode Rejection Ratio (@0dBu)	>91dBu
Maximum input level (@1% distortion)	+22dBu
Connectors	four 3-pin Euro breakaway terminals

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

Description:

The AEC input card provides 4-channel AEC (Acoustic Echo Cancellation), phantom power, 84dB gain level, and 58dB Mic Amp (Microphone Amplifier). DSP-AEC4 LD+ provides AEC broadband processing. Each channel of AEC input card has independent input and direct output for local sound reinforcement.

Features:

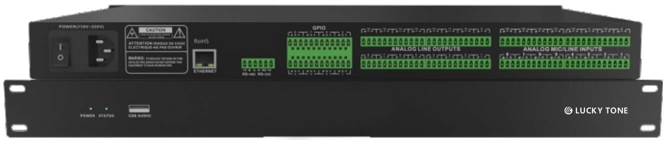
- 4-channel AEC input card with DSP processing capability
- With additional DSP processing power to serve the system
- Easy for installation
- Control and configure via software

SPECIFICATIONS	DSP-AEC4 LD+
Dynamic range	>118dB
Frequency response	(±0.2dB) 20Hz~20kHz
Input impedance	5.5k Ω
Channel Crosstalk	<-112dB
THD+N	<0.002%
Common mode rejection ratio (@0dBu)	>91dBu
Maximum input 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-880 LD+
Type	Network interface, D/A conversion
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