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2023

S-TRACK Audio Product Manual

Fast response Service provider of Audio Fusion Information Technology

Web:www.s-track.com.cn

COMPANY PROFILE

2013

S-TRACK Was Established in 2013

S-TRACK was founded in 2013, is a national high-tech enterprise focusing on the audio visual field, dedicated to promote the digitalization, networking, intelligent process of AV industry, to provide professional AV one-stop service for partners in the field of smart government, smart education, smart finance, etc.

Served 2000+ Clients

At Home and Abroad

In the past three years, the sales of audio and video products are in the top 3 in the industry, serving 2000+ customers at home and abroad, with products in many segments of smart cities.

More than 20 Years of Technology

The Core Technology Team Has Focused on AV for Over 20 Years S-TRACK focus on AV exploration and product development, believe

that the convenience brought by technology is to make audio transmission clear, efficient and safe, with strong R & D capability and years of technical accumulation, with dozens of scientific and technical patents, for the rapid development of audio and video field and product implementation to provide a solid theoretical guarantee.

Over 200 Patents Software Publications

Have a Comprehensive Supply Chain System

It has more than 200 patent software works such as a digital audio processor DSP software for pick-up ring hearing aid system, a digital mixer, an audio speaker with multiple independent audio outputs, and network audio server control software.

Fast response Service provider of Audio Fusion Information Technology

COOPERATION PARTNERS

Texas Instruments | Analog Devices | Cirrus Logic | Dante | 重庆银行 BANK OF CHONGQING | 中国农业银行 AGRICULTURAL BANK OF CHINA

amazon | 中信银行 CHINA CITIC BANK | 浙江大学 ZHEJIANG UNIVERSITY | 北京理工大学 BEIJING INSTITUTE OF TECHNOLOGY | 中华人民共和国深圳海关 SHENZHEN CUSTOMS DISTRICT P.R.CHINA | 辽宁中医药大学

HONORS AND QUALIFICATIONS

S-TRACK 声菲特

Honor Certificates

2020 Guangdong Province Honour Contract and Credit Enterprise

National High-Tech Enterprise Certificate for Several Times

Shenzhen "Specialized and Unique" Enterprise ......

Invention Patents / Utility Model Patents

A Digital Mixer
A Mobile Audio with Easy Assembly
A Surround Type Microphone Device
A Portable On-Site Audio Processing Box
A Method of Long-Distance Audio Pickup
An Audio Device with An Independent Alarm System
An Audio Processing Device with Adaptive Sound Field
A Control Panel with A Positioning Mounting Structure
A Multi-Channel Audio Synchronous Transmission Circuit Invention ......

Test Certificates / Design Patents

Many Certifications at Home and Abroad:3C, RoHS, CE,FCC, GB
Desktop Matrix Microphone Design Patent
Digital Audio Processor Design Patent
Mobile Speaker Design Patent
......

Certificates / Software Copyrights

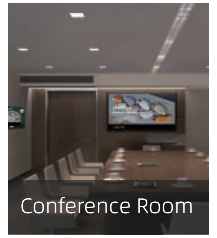
Environmental Management System Certification
Quality Management System Certification.......
S-TRACK Technology Distributed Networked Digital Audio Management System
S-TRACK Technology Multimedia Matrix Switcher Control Software
S-TRACK Technology All-Digital Conference System Control Software
S-TRACK Technology Digital Conference System Control Software
S-TRACK Technology Dante Audio Transmission Control Software
S-TRACK Technology Digital Amplifier Control Software
S-TRACK Technology USB Recording Control Software

DM12 Digital Mixer（Hippo）

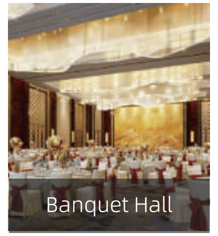
With innovative design and powerful DSP function, digital tuning system is perfectly integrated. The new generation of digital tuning station has excellent sound quality, reliable product quality and innovative user operation, which can meet the stringent needs of tuning engineers, and achieve the creativity and appeal of performers with the highest possible quality audio.

Product Features

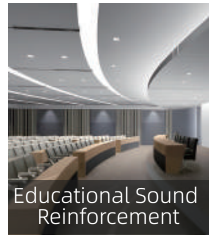
- Rich analog and digital interface
- Built-in USB recording and playback function
- Synchronize dual-system hot backup data in real time
- Compatible with multiple operating systems
- 7 "capacitive touch screen
- Supports 255 scenarios



Conference Room



Banquet Hall



Educational Sound Reinforcement

Video Conferencing

(Suitable for conference rooms, banquet halls, teaching sound reinforcement, video conferencing and other scenarios.)

Model	Hippo DM12
Input	Ch1-12 balanced XLR/TRS input; Two sets of 13/14~15/16 stereo TRS input; 1 set of S/PDIF digital input (coaxial and fiber interface); USB Sound Card (16x16 optional)
Output	MainLR bus output; Four AUX auxiliary outputs and one MonitorLR monitor output; 1 set of S/PDIF digital output (coaxial and fiber interface), 1 set of AES/EBU output.
Total harmonic distortion & noise	< 0.002% @ 18dBu A+ Right
Sample rate	48K
Background noise	-92dBu A+ Right
Screen	7 inch HD touch screen, 1024×600 resolution.
Frequency response (20~20KHz)	20HZ ~ 20K HZ ,±0.2dB
Quantization number	24bit
Maximum level (input)	+22dBu， Balanced
Maximum level (output)	+22dBu， Balanced
Phantom power supply	48V
Modular/digital dynamic range	110dB
Input-output dynamic range	108dB
Input impedance (balanced)	20KΩ
Output impedance (balanced)	100Ω
Channel isolation @1KHz	100dB
Operating temperature	0℃-55℃
Working power supply	220V/50hz
Power consumption	45W

Portable Digital Mixer (HIPPO)

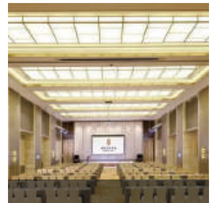
Model: D1608

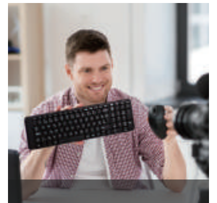
The product built-in USB recording playback function, while supporting APE\MP3\FLAC\WAV lossless audio format. With a simple digital interactive interface, professional mixing effect, both in a professional performance to play outstanding ability, but also to fully meet the inexperienced individual users to provide a powerful effect, more in line with the modern multi-function hall and various types of conference rooms on the professional requirements of audio, can be widely used in speeches, training, meetings, entertainment and gala events and other scenes.

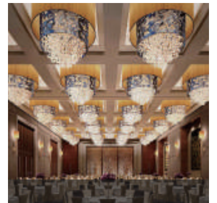
Product Features:

- Excess Pro DSP
- 4 Pro Effects Algorithms
- USB Recording Playback
- Support 30 Sets of Scene Presets
- 16 Analog Input Interface
- Support Major Operating Systems


Conference Room


Multifunctional Hall


Live Webcast


Banquet Hall

(Mainly applied to speech, lecture, training, presentation, report, academic exchange, meeting, discussion, karaoke entertainment joint activities and other scenes.)

Model	HIPPO D1608
Input	In1~8 balanced XLR/TRS combo input jacks; stero Input 9~16 TRS1/4 stereo input jacks; 2-way USB3.0 input sound card
Output	LR main bus output; 4-way AUX auxiliary output; 1-way TRS monitor
Screen	7" HD touch screen, 1024×600 resolution
Total Harmonic Distortion & Noise	< 0.002% @ 18dBu A+ Right
Frequency Response	20HZ~20K Hz,±0.2dB
Sampling Rate	48K
Signal to Noise Ratio	-90dBu
Maximum Level (Input)	+20dBu, Balanced
Maximum Level (Output)	+15dBu, Balanced
Input Impedance (Balanced)	20KΩ
Output Impedance (Balanced)	100Ω
Channel Isolation	@1KHz100dB
Power Consumption	30W
Number of Quantized Bits	24bit
Phantom Power	48V
Operating Power Supply	19/2A
Analog/Digital Dynamic Range	100dB
Digital/Analog Dynamic Range	100dB
Input to Output Dynamic Range	108dB
Size	410mm×253.5mm×69mm

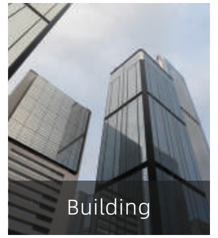
Digital Audio Processor（PUMA）

PUMA Digital network audio Processor is a free design of audio processing and control system products, using PUMA Designer software for system configuration, control and monitoring over the network, support for static or automatic address allocation. Adopt advanced DSP processing technology, built-in excellent conference application technology, these are by Sonfit years of market accumulation and research and development and will continue to update. Abundant I/O interface, the 1RU PUMA processor has multiple analog I/O input/output interfaces, USB A interface, USB B interface, RS232 interface and GPIO interface at the same time. It is a processor with high performance and strong function among similar products in the market, compatible with the existing and future Sonfit accessories.

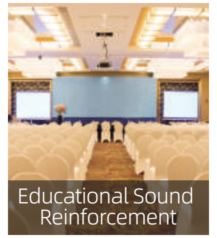
Product Features

- Super large system support
- Rich I/O interface
- Unified software platform
- Powerful DSP processing
- Open design
- Open interaction

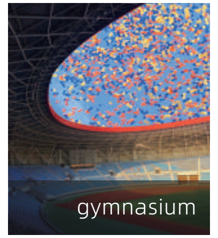
Large conference room



Building



Educational Sound Reinforcement



gymnasium

(Suitable for large conference room, building, multi-function hall, gymnasium and other large scene)

Model	PUMA
nput frequency response	20Hz-20kHz@+18dBu ±0.2dB
Input total harmonic distortion + noise	@1KHz, +18dBu sensitivity & +8dBu input, < 0.003%
Equivalent input noise	< -125dB
Cross talk between inputs @1kHz	>100dB
Input dynamic range	@ +18dB usensitivity , > 110 dB
Input common-mode noise suppression@60Hz	60dB
Input impedance (balanced)	2.4kΩ Nominal
Input sensitivity range	-39dBu ~ +18dBu
Phantom power supply	+48V direct current，The maximum output current is 8mA
Sampling rate	48kHz
AD/DA conversion	24bit
Output frequency response	20Hz-20kHz，±0.2dB
Output crosstalk@1kHz	>100dB Typical
Output dynamic range	>108dB
Output impedance (balanced)	100Ω
Maximum output level	18dBu/4dBu
Number of USB channels	2*2
Analog input/output channel	16/16
Dante Input/output channels	64/64
The dimensions(HWD)mm	（Product）44 x 483 x 260 / （Transport）115 x 585 x 345

Network Digital Audio Processor (PANDA)

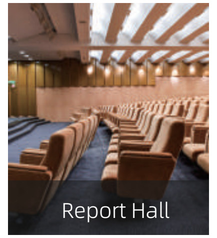
Model: 44S、88S、1616S、44N、88N、1616N、D88S、D1616S、D88N、D1616N

The digital audio processor is a freely designable audio processing and control system product. The front panel comes with a high-definition color screen to display the current working status of the device. It adopts advanced DSP processing technology with new algorithms of automatic mixing and feedback cancellation, which are targeted to solve various practical problems in application scenarios. Optional Dante module provides a high-bandwidth, low-latency, high-compatibility and low-cost solution for network audio transmission. The new UI software interface with integrated Dante control software makes debugging more convenient.1.

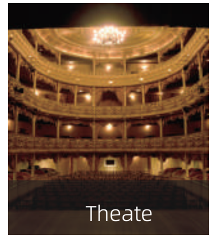
Product Features:

- OLED HD Color Screen
- High Performance DSP
- Optional Dante Module Available
- Real-Time Backup Security and Stability
- Intelligent Generation of Central Control Instructions
- USB Recording Playback

Conference Room



Report Hall



Theate

Multifunctional Hall

(Suitable for conference room, multi-function hall, theater, lecture hall, assembly hall and other scenes.)

Model	44S	88S	1616S	44N	88N	1616N	D88S	D1616S	D88N	D1616N
DSP Chip	Ti 456MHz FLOPS Dual Core									
USB2.0 Record/Playback	Support									
Central Command Set	Support									
Input Per Channel	Preamp, Signal Generator, Expander, Compressor, 5-Band Parametric EQ									
Output Per Channel	31-Band Graphic Equalizer, Delay, Crossover, Limiter									
Matrix Mixing	Matrix Mixing of Input and Output Signals, Mixing Component Control									
Auto Camera Tracking Function	Support									
Scene Preset	8-100 Sets of Scene Presets									
Chassis Size(W×D×H)	482*258*45mm									
Analog Input Channels	4	8	16	4	8	16	8	16	8	16
Analog Output Channels	4	8	16	4	8	16	8	16	8	16
Dante Input Channels	/	/	/	/	/	/	8	16	8	16
Dante Output Channels	/	/	/	/	/	/	8	16	8	16
Ethernet Control Port	1	1	1	1	1	1	1	1	1	1
Dante Interface (Main)	/	/	/	/	/	/	1	1	1	1
Dante Interface (Standby)	/	/	/	/	/	/	1	1	1	1
RS232/RS485	1/1	1/1	1/1	1/1	1/1	1/1	1/1	1/1	1/1	1/1
AFC/AEC/ANC	√/×/×	√/×/×	√/×/×	√/√/√	√/√/√	√/√/√	√/×/×	√/×/×	√/√/√	√/√/√
AM	√	√	√	√	√	√	√	√	√	√
AGC	√	√	√	√	√	√	√	√	√	√

Amplifier Audio Processor (LION V)

Model：V44N、V88N

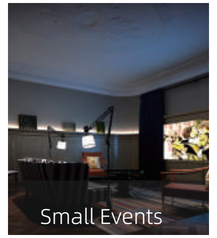
LION Amplifier Audio Processor integrates a dual-channel 145W digital amplifier module and dual-channel UHF wireless receiver mode, signal processing and power amplification integrated in a device to work, shortening the signal path, simplifying the engineering wiring, can quickly build an audio system in a limited space. Support RS232 control port with power supply. Optional the latest feedback cancellation (AFC), echo cancellation(AEC), noise cancellation (ANC) algorithm, client support for Windows, IOS, Android systems, support for browser-side control, fully compatible with IE, CHROME, FIREFOX.

Product Features:

- 51dB Ultra-Wide Gain Amplification
- High-Speed Floating Point DSP
- Wireless Microphone
- AFC/AEC/ANC
- Dual-Channel Class-D Amplifiers
- USB Recording / Playback

Conference Room

Exhibition Hall



Small Events

Classroom Interaction

(Suitable for conference rooms, classrooms, training rooms, local sound reinforcement, interactive teaching and other scenarios.)

Model	V44N	V88N
DSP Chip	Ti 456MHz FLOPS Dual Core	
USB2.0 Recording / Playback	Support	
Central Control Command Set Generator	Support	
Input Per Channels	Preamp, Signal Generator, Expander, Compressor, 5-Band Parametric EQ	
Output Per Channels	31-Band Graphic Equalizer, Delay, Crossover, Limiter	
Matrix Mixing	Matrix Mixing of Input and Output Signals, Mixing Component Control	
Auto Camera Tracking Function	Support	
Scene Preset	8-100 Groups of Scene Presets	
Amplifier Module	Built-in 8Ω 2×150W Class D Digital Amplifier	
Wireless Microphone Module	Built-in UHF Dual-channel Wireless Microphone Receiver (Optional Handheld, Lavalier Wireless Microphone Combination)	
Cabinet Size (W×D×H)	482*258*45mm	
Input Gain Amplification	-24~27dB，3dB Level One Total 17 Levels	
Analog Input Channels	4	8
Analog Output Channels	4	8
Ethernet Control Port	1	1
RS232	1	1
WIFI Online	√	√
AFC/AEC/ANC	√/√/√	√/√/√
AM	√	√
AGC	√	√

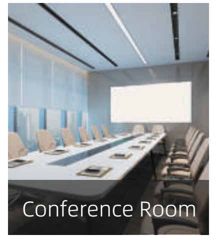
Amplifier Audio Processor (LION)

Model：44N、88N

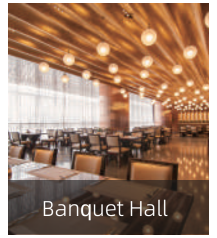
LION Amplifier Audio Processor integrates a dual-channel 145W digital amplifier module, signal processing and power amplification integrated in one device to work, shortening the signal path and simplifying the engineering wiring. Support with power RS232 control port. Optional the latest feedback cancellation (AFC), echo cancellation (AEC), noise cancellation (ANC) algorithm. Client support Windows, IOS, Android system, support browser-side control method, fully compatible with IE, CHROME, FIREFOX.Translated with www.DeepL.com/Translator (free version)

Product Features:

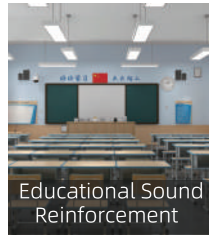
- 51dB Ultra-Wide Gain Amplification
- High-Speed Floating Point DSP
- 100 Groups of Scene Presets
- AFC/AEC/ANC
- Dual-Channel Class-D Amplifiers
- USB Recording /Playback



Conference Room



Banquet Hall



Educational Sound Reinforcement

Video Conferencing

(Suitable for conference rooms, banquet halls, teaching sound reinforcement, video conferencing and other scenarios.)

Model	44N	88N
DSP Chip	Ti 456MHz FLOPS Dual Core	
USB2.0 Recording / Playback	Support	
Central Control Command Set Generator	Support	
Input Per Channels	Preamp, Signal Generator, Expander, Compressor, 5-band Parametric EQ, Auto Gain, AM Auto Mix function	
Output Per Channels	31-band Graphic EQ, Delay, Crossover, High/Low pass filter, Limiter	
Matrix Mixing	Input and Output Signal Matrix Mixing, Mixing Component Control	
Auto Camera TrackingFunction	Support	
Scene Presets	100 Groups of Scene Presets	
Amplifier Module	Built-in 8Ω 2×150W Class D Digital Amplifier	
Input Gain Amplification	-24~27dB，3dB Level One Total 17 Levels	
MIC/LINE I/O Support	Support	
48V Phantom Power Switch Per Channel	Support	
Analog Input Channels	4	8
Analog Output Channels	4	8
Ethernet Control Port	1	1
RS232	1	1
WIFI on-line	√	√
AFC/AEC/ANC	√/√/√	√/√/√
AM	√	√
AGC	√	√

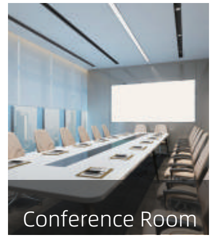
Network Digital Audio Processor (SWIFT)

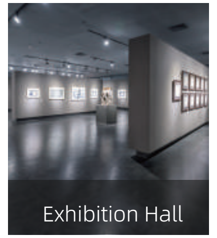
Model: 88S、1616S、D88S

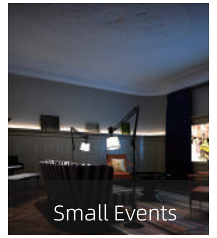
The product adopts advanced DSP audio processing technology, built-in automatic mixing console and feedback cancellation module, as well as the central control code random generation, automatic power failure protection memory, one key reset and many other functions. The simple and easy-to-understand graphical software control interface brings customers a quick and real-time operating experience, and with other audio products of S-TRACK, it can meet the needs of more scenarios of applications. Client support for Windows, Android systems, support browser-side control mode, fully compatible with IE, CHROME, FIREFOX.

Product Features:

- Feedback Cancellation Algorithm
- High Speed Floating Point DSP
- Intelligent Software Control
- Channel Copy|Paste | Linked Control Function
- Automatic Power Failure Protection Memory
- 100 Groups of Scene Presets


Conference Room


Exhibition Hall


Small Events


Classroom Interaction

(Applicable to scenes suitable for conference rooms, exhibition halls, small events, classrooms, etc.)

Model	88S	1616S	D88S
DSP Chip	Ti 456MHz FLOPS Dual Core		
Central Control Command Set Generator	Support		
Input Per Channels	Preamp, Signal Generator, Expander, Compressor, 5-band Parametric EQ, Auto Gain, AM Auto Mix function		
Output Per Channels	31-Band Graphic EQ, Delay, Crossover, High/Low Pass Filter, Limiter		
Matrix Mixing	Input and Output Signal Matrix Mixing, Mixing Component Control		
Scene Preset	100 Groups of Scene Presets		
Phantom Power	48V		
Preamp Amplification	42dB (6dB per step, 7 steps in total)		
Channel Isolation	104dB @1KHz，4dBu		
Cabinet Size (W×D×H)	482*258*45(mm)		
Analog Input Channels	8	16	8
Analog Output Channels	8	16	8
Dante Input Channels	/	/	4
Dante Output Channels	/	/	4
Ethernet Control Port	1	1	1
RS485	1	1	1
GPIO	2	2	2
AFC	√	√	√
AM	√	√	√
AGC	√	√	√

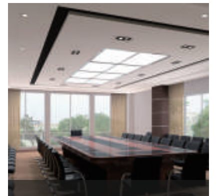
10 IN 4 OUT MINI Processor (ABOX)

Model: 1004N

ABOX MINI processor, equipped with excellent performance of digital audio processing technology, including full-band adaptive noise cancellation, feedback cancellation algorithm, echo cancellation algorithm and a variety of modules, using a stable two-way simultaneous speech (Double Talk) detection method, even in strong background noise and non-linear distortion environment can effectively eliminate excess echo, fast and accurate tracking of environmental noise changes and maintain good output sound quality.

Product Features:

- 51dB Ultra-Wide Gain Amplification
- High-Speed Floating Point DSP
- 10-channel Input Signal Interface
- AFC|AEC|ANC Algorithms
- PNP Plug-and-play Easy Configuration
- 48V Phantom Power


Conference Room


Court


Command Center


Classroom Interaction

(Suitable for conference rooms, courts, command centers, classroom interaction, etc.)

Model	1004N
DSP Chip	Ti 456MHz FLOPS Dual Core
Input Per Channels	Preamp, Expander, Compressor, 5-band Parametric EQ, Auto Gain, Auto Mix
Output Per Channels	31-Band Graphic EQ, Delay, Crossover, Limiter
Matrix Mixing	Matrix Mixing of Input and Output Signals
Gain Adjustment	Support
Input Gain Amplification	42dB(6dB per step, 8 steps in total)
Balanced/Unbalanced I/O	Support
Independent 48V Phantom Power Switch	Support
Channel Isolation	100dB @1KHz
Cabinet Size (W×D×H)	180*168*42(mm)
EIN (A weighted)	<-120dBu
THD+N:MIC Channels	0.005%@4dBu; LINE Channel: 0.01%@4dBu
Analog MIC Input	8
Analog LINE Input	2
Analog Output	4
Ethernet Control Port	1
USB Upgrade	1
AFC/AEC/ANC	√/√/√
AM	√
AGC	√

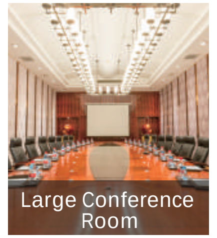
Speaker Manager (DOLPHIN)

Model: 26、48、D26、D48

Dolphin Series Speaker Manager supports multiple analog signal input and output, multiple Dante signal input, built-in high-performance DSP processing chip, comes with 31-segment graphic equalizer, 15-segment parametric equalizer, gain control, compressor, delay and other functions, with stability and linear phase of 512-order custom FIR filter for multi-speaker scene signal distribution management; professional PC control software, connected via USB or RJ45 cable to achieve PC software control settings.

Product Features:

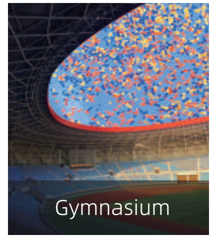
- TFT Interactive LCD
- High Performance DSP
- Channel Copy Function
- Channel Interbinding Function
- 15-Band Parametric Equalizer
- Built-in Dante Chip



Large Conference Room

Exhibition Hall

Multifunctional Hall



Gymnasium

(Suitable for large conference rooms, exhibition halls, multi-purpose halls, stadiums and other occasions where multiple speakers need to be managed.)

Model	26	48	D26	D48
DSP Chip	DSP Processing Chip, 24bit, 48kHz Processing Capability			
Screen	TFT Interactive LCD, Quick Reference Embedded GUI Interface			
Filter	512-Order Custom FIR filter with Stability and Linear Phase			
Input Per Channels	Mute, gain, delay (0-180ms), 31-band Graphic Equalizer, Expander, Compressor, Speaker, Gain function			
Output Per Channels	Delay (0-60ms), crossover (Butterworth/Bessel/Linkwich filter type 12dB/24dB/36dB/48dB filter slope, FIR filter), 15-band Parametric EQ, Gain, Limiter function.			
Input and Output Channel Copy Function	Support			
Channel Inter-binding Function	Support			
Scene Preset	Support Multi-group Scene Save, Recall and Import, Export Functions			
Cabinet Size (W×D×H)	440x208x44mm			
Analog Input Channels	2	4	2	4
Analog Output Channels	6	8	6	8
Dante Input Channels	/	/	2	4
Network Port	/	/	RJ45	RJ45
Frequency Response	20 - 20KHz，±0.2 dBu			
Maximum Input Level	+20 dBu，Balanced Input			
Maximum Output Level	+20 dBu，Balanced Output			
Bi-directional RS232 Interface	√			
THD+N	<0.002% @ 4 dBu			
Channel Isolation	105 dB @ 1 KHz			
Operating Temperature	-10-50℃			

Network Audio Interface Box (OSTRICH)

Model：D88、D1616

(Main Screen)

(Output Interface)

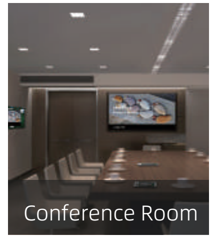
(Configuration Interface)

(Input Interface)

OSTRICH Series products based on Dante international common protocol audio interface equipment, can realize the analog audio signal and Dante network digital audio interconversion, through Dante protocol routed to any Dante audio transmission network equipment. The device uses high-quality AD / DA signal conversion chip, comes with TFT screen, embedded GUI software control, provides 48V phantom power supply, can be accessed to the traditional analog audio equipment, such as mixers, amplifiers, processors and so on.

Product Features:

- High Quality
- Low Latency
- Long Distance
- Easy Networking
- Strong Anti-Interferenc
- SAD/DA Conversion Chips



Conference Room

Multifunctional Hall

Opera House

Theatres

(Suitable for large conference rooms, multi-function halls, theaters, theaters, extended processor input and long-distance signal access, etc...)

Model	D88	D1616
Chip	AD/DA Signal Conversion Chip	
Input	Phantom Power, Channel Mute, Sensitivity Adjustment	
Output	Channel Mute	
Indicator Light	Channel Signal Indicator, Clipping Overload Indicator, Phantom Power Indicator	
Control Method	Edit Knob, Control Button, Return Button Control, with TFT Screen, Embedded GUI Software Control	
Phantom Power	Support 16-way 48V Phantom Power, Access to Condenser Microphone	
Sensitivity	0dB ~ 51dB,3dB Per Level	
Number of Quantization Bits	24bit	
Sampling Rate	48k	
Frequency Response	20HZ ~ 20K HZ ,±0.5dB	
Background Noise	-90dBu	
THD+N	＜0.002% @ 4dBu	
Analog/Digital Dynamic Range	118dB	
Digital/Analog Dynamic Range	123dB	
Channel Isolation	@1kHz 100dB	
Product Size	486mm×156mm×110mm	
Analog Inputs	8	16
Analog Outputs	8	16
Dante Inputs	8	16
Dante Outputs	8	16

Bluetooth Transmission Panel (OSTRICH)

Model: BD22、BD44

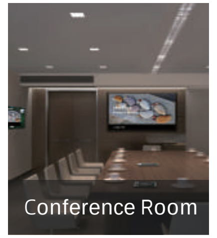
(BD22) (BD44)

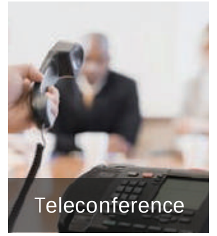
Product Features:

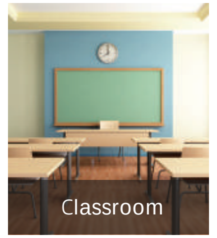
- Easy to Use
- Good Compatibility
- Bluetooth 5.0
- Integrated Power Supply and Transmission
- Stereo I/O
- Bluetooth Call Bridging Function

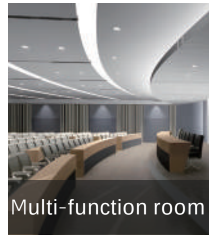
Ostrich BD22 has Bluetooth and Dante interconversion, bi-directional stereo transmission function, through the Bluetooth interface and cell phones, iPad and other devices connected, the received audio through Dante converted to network digital signal transmission.

Ostrich BD44 is a Dante Audio Interface (4x4) and Bluetooth Audio Interface (2X2), Analog Audio Interface (RCA or TRS) interconversion device, support Bluetooth wireless audio interface and analog audio interface.Unified PoE power supply, power, control and audio data are connected by network cable transmission, you can open Dante Controller control software on the PC, the routing configuration of audio, you can complete the system build.


Conference Room


Teleconference


Classroom


Multi-function room

Suitable for teleconferencing, audio long-distance transmission, audio media transmission and other application scenarios.

Model	BD22	BD44
Input Interface	Dante *2, Bluetooth 5.0, Stereo	Dante *4, Bluetooth 5.0, Stereo, RCA*2, 3.5mm TRS*1
Output Interface	Dante *2, Bluetooth 5.0, Stereo	Dante *4, Bluetooth 5.0, Stereo, 3.5mm TRS*1
Frequency Response	20Hz-20kHz	20Hz-20kHz
Sampling Rate	48KHz	48KHz
THD+N	<0.005%	<0.005%
Background Noise	-90dB	-90dB
Signal to Noise Ratio	>100dB	>100dB
Power Consumption	2W	2W
Operating Temperature	-10℃ -40℃	-10℃ -40℃
Operating Humidity	5-95%	5-95%
Size(LxWxH)	123.5mm×79mm×35mm	123.5mm×109.1mm×30mm
Weight	156g	310g

Dante Signal Conversion Interface (OSTRICH)

Model: DI22、DO22、DU22

Product Features:

- Lightning and Surge Protection
- Small and Exquisite
- Microsecond Latency
- Support PoE Power Supply
- Dante Software Support
- 3 Types of Signal Conversion

Model	DI22	DO22	DU22
Maximum Signal Level (Balanced)	+18dBu	+18dBu	/
Frequency Response	20Hz-20kHz,±0.5dB	20Hz-20kHz,±0.5dB	/
Impedance	20kΩ(Balanced) 10kΩ(Unbalanced)	110Ω (Balanced)	/
Dynamic Range	>100dB	>100dB	/
Signal to Noise Ratio	>100dB	>100dB	/
THD+N	<0.003%@4dBu	<0.003%@4dBu	/
Connector	RJ45&1XLR-F RJ45&2XLR-M	RJ45&1XLR-M RJ45&2XLR-M	RJ45 and USBA Series
Power Supply	PoE IEEE802.3af Standard	PoE IEEE802.3af Standard	USB
USB	/	/	Specification Level USB 2.0 Devices
Dante Device Latency	1, 2 or 5ms (Configurable with Dante Controller)		
Network Transmission	Dante IP AES67 RTP Audio		

Product Introduction

OSTRICH DI22 supports two analog audio signal inputs and two Dante network audio signal outputs.
OSTRICH DO22 supports two channels of Dante network audio signal access and two channels of analog audio signal output.
OSTRICH DU22 supports USB signal to Dante signal and Dante signal to USB signal output for bi-directional channel signal transmission.

Dante Signal Interface Box (DACO)

Model: DACO 88、DACO 1616

Product Features:

- AoIP Audio Network
- High Quality Low Latency Transmission
- 48V Phantom Power
- RJ45 Network Interface
- Bi-directional RS232 interface
- Web Control Su pport

Model	DACO 88	DACO1616
Analog Input and Output Channels	8/8	16/16
Dante I/O Channels	8/8	16/16
Sampling Rate	48KHz	
Frequency Response	20~20KHz, ±0.5dB	
Phantom Power (Per Input)	48V	
EIN (A Weighted)	≤-125dBu	
Input Gain Amplification	-6~36dB,6dB level, total seven levels	
Analog/Digital Dynamic Range	(A-weighted)114dB	
Digital/Analog Dynamic Range	(A-weighted)120dB	
Channel Isolation	107dB@1kHz	
Input Impedance (Balanced)	20kΩ	
Output Impedance (Balanced)	100Ω	
Cabinet Size (WxDxH)	483*258*45(mm)	

Product Introduction

DACO series network audio interface machine to achieve network digital audio signal and analog audio signal interconversion, support the maximum 16-way input and 16-way output analog and Dante channel, is the analog audio system for digital upgrade, expansion of the excellent choice, suitable for large conference rooms, multi-function hall, theater, theater, expand the processor input and long-distance signal access.

2 In 2 Out Dante Wall Interface (DPANEL)

Model: Dpanel

Product Features:

- Input Phantom Power Switch
- Input Gain Adjustment
- Dante Module
- 2-in-1 XLR Input Connector
- Standard 120 Type Pre-embedded Box
- Analog/Dante Signal Transfer

Model	DPANEL
Sampling Rate	48KHz
THD+N	≦0.005% @4dBu
Phantom Power	48V
Frequency Response	20Hz~20K Hz，±0.5dB
Common Mode Rejection	80dB @80 Hz
Background Noise	-90dBu
Maximum Output Level	20dBu
Maximum Line Input	11dBu
Channel Isolation	100dB @1k Hz
Input Impedance (Balanced)	20kΩ
Output Impedance (Balanced)	100Ω
Operating Power	PoE Power Supply

Product Introduction

DPANEL series wall interface machine, which has analog input and Dante digital input, analog line output and Dante digital output, analog input support phantom power and manual preamp gain adjustment, suitable for conference room, multi-function hall, reserved for hidden input and output interface.

4 In 4 Out Network Audio Interface Machine (DBOX)

Model: 44-H

Product Features:

- AoIP Audio Network
- PC|PAD Side Software Control
- TI Dual-Core High-Speed Floating Point DSP
- AFC,AEC,ANC Algorithms
- Bi-directional RS485 Interface
- POE Method Power Supply

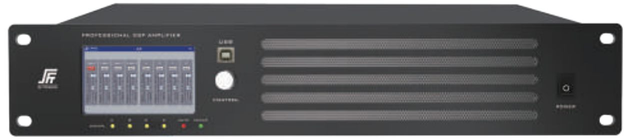
DModel	DBOX 44-H
Inputs Per Channels	Preamp, Signal Generator, Expander, 5-band Parametric EQ, AFC, AEC, ANC
Outputs Per Channels	Speaker Manager (31-band Graphic Equalizer, Limiter)
Analog Channels	4-channel balanced input + 4-channel balanced outpu
Dante Channels	4-channel input + 4-channel output
Channel Isolation	100dB@1kHz，4dBu
Maximum Input Level	10dBu
Maximum Output Level	14dBu
Sampling Rate	48kHz
Input Impedance	20 KΩ
Output Impedance	100 Ω
Input Gain Adjustment	6～36dB, 6dB level total 8 levels
Operating Power	AC110V-220V,50Hz/60Hz；POE48V；DV12V
Size (WXDXH)	142X157*46mm

Product Introduction

DBOX series network audio interface machine to achieve network digital audio signal and analog audio signal interchange, 4 × 4 is the analog multimedia system for digital upgrade, expansion of the excellent choice for large conference rooms, multi-function hall, theaters, theaters, conference centers, extended processor input and long-distance signal access, distributed processing to reduce the burden of audio processing in the server room.

DSP Network Amplifier (WHALE)

Model: D4600、D4750、D4900



Model	Whale D4600	Whale D4750	Whale D4900
8Ω Stereo Power	4X600W	4X750W	4X900W
Signal-to-noise Ratio	>112dB		
Damping Coefficient	> 1000@ 8Ω		
Total Harmonic Distortion	<0.1%(20Hz-20 kHz 1W)		
Frequency Response	20Hz-34KHz(+0/-0.3dB,1W/8Ω)		
Level Adjustment	Front Panel Potentiometer, from Negative Infinity to 0dB		
Cooling Method	Stepless Speed Control Fan, Airflow from Front to Back		
Amplifier Protection Method	Short Circuit, Circuit Breaker, DC Voltage, Overheat, Overvoltage, RF, Ultra Low Frequency Protection		
Power Supply Specifications	AC voltage 180-240VC (90V-120VAC) 50-60Hz		
Cabinet Size(H*W*L)	88mmX490mmX438mm		
Package Size(H*W*L)	155mmX600mm×550mm		
Net weight	13.5kg		
Gross weight	16.5kg		

Product Features:

- High Performance DSP Processor
- Color Touch Scree
- Dante Network Transmission Module
- Overcurrent Protection
- Overheat Protection
- Multi-scene Save Call

Product Introduction

4-channel DSP Network Amplifier with high-performance ADI DST processor, high-fidelity low-noise 24-bit, A/D and D / A conversion, embedded Dante network transmission module, support 4 Dante signal / 4 analog signal input, 4 groups of digital power amplification output, and provide USB and RJ45 interface and computer connection, each output channel with 8-segment parametric equalization, crossover, delay, limiter, gain, polarity.

Digital Amplifiers (WHALE)

Model: 2150、2350、2650、4150、4350、8150

Model	Whale 2150	Whale 2350	Whale 2650
Output (8Ω, 1kHz)	2x150W	2x350W	2X650W
Output (4Ω, 1kHz)	2x250W	2x500W	2x950W
Bridge output power per 2 channels	1x450W	1x950W	1X1800W
Frequency Response	20Hz~20kHz ±3dB		
Conversion Rate	20V/us		
Cabinet Size	483 x 89 x 249 (mm)		
Model	**Whale 4150**	**Whale 4350**	**Whale 8150**
Output (8Ω, 1kHz)	4x150W	4x350W	8x150W
Output (4Ω, 1kHz)	4x220W	4x500W	8x250W
Bridge output power per 2 channels	2x450W	2x950W	4x500W
Frequency Response	20Hz~20kHz ±3dB		
Conversion Rate	20V/us		
Cabinet Size	483 x 89 x 249 (mm)		580 x 130 x 420 (mm)

Product Features:

- Independent Cooling System
- Delayed Start System
- Automatic Limiting Output
- Overcurrent Protection
- Over Temperature Protection
- Short Circuit Protection

Product Introduction

The digital amplifier back panel is equipped with dual-channel, single-channel output conversion, SPEAKON output connector, balanced input interface, large-capacity switching power amplifier, cooling fan with advanced infinitely variable speed circuit control, and automatic limiting output, short circuit, overload, over temperature, power-on delay and other protection functions to protect the speaker from damage due to impact.

Dante POE Ceiling Speaker (EAGLE)

Model: XDP6、XDP8

Model	EAGLE XDP6	EAGLE XDP8
System Type	1 X 6.5"+3 X 1.5" coaxial unit	1 X 8"+3 X 1.5" coaxial unit
Size	240x240(mm)	281x251(mm)
Net Weight	3.5kg	5.5kg
Sound Pressure Level	110dB	111dB
Sensitivity	90dB	92dB
Power Rating	30W	90W
Transmission Protocol	Dante	Dante
Power Supply Method	PoE Power Supply	PoE Power Supply
Frequency Response	80-20KHz	75-20KHz
Coverage Angle	100°	90°*90°
Input Interface	RJ-45	RJ-45
Opening Size(mm)	210	250
Opening Depth(mm)	260	280

Product Features:

- Built-in Amplifier Module
- Support PoE Power Supply
- Dante Transmission
- Software with Full Function Contro
- Multifunctional DSP Chips
- Gain/Silence Leveling Function

Product Introduction

Ceiling speakers, generally installed in the ceiling, used in background music and conference systems, the advantage is that the system can be better hidden after installation of the speaker. The speaker is mainly used in airports, hotels, conference rooms, convention centers, restaurants, schools, hospitals, shopping malls, stores and various background music sound reinforcement places.

Dante Active Speakers (EAGLE)

Model: DY6、DY8、DY10、DY12

Model	DY6	DY10
Sensitivity	(1W/1m)89dB	(1W/1m)90dB
Unit Configuration	6.5"(165 mm) woofer1" (25 mm) tweeter driver	10"(250 mm) Ferrite Bass 1.5"(34 mm) Ferrite Tweeter Driver
Frequency Response	(±10dB)100Hz-18kHz	(±3dB)50Hz-20kHz
Power Rating	50W	120W
Coverage Angle	(H x V) 80° x 80°	(H x V) : 80°x 80°
Active Amplifier Module	100W	200W
Model	**DY8**	**DY12**
Sensitivity	(1W/1m)89dB	(1W/1m)98dB
Unit Configuration	8"(200 mm) Ferrite Bass 1"(25 mm) Ferrite Treble Driver	12"(300 mm) Ferrite Bass1.75" (44 mm)Ferrite Treble Driver
Frequency Response	(±3dB)55Hz-20kHz	(±3dB)50Hz-20kHz
Power Rating	100W	350W
Coverage Angle	(H x V) : 80° x 80°	(H x V) : 80°x 60°
Active Amplifier Module	100W	600W

Product Features:

- Dante Transmission Technology
- ClassD Amplifier Technology
- Support Analog Signal Input
- No Compression, Loss
- Network IP-Based Architecture
- No Latency

Product Introduction

EAGLE DY series speakers use the world's top Dante protocol audio transmission technology. Products using ClassD amplifier technology, through a standard cat5e the same line can simultaneously achieve power supply and transmission, without additional power amplification equipment, to achieve sound without compression, loss and delay; network IP-based architecture design to achieve networked, digital.The speaker is mainly used in airports, hotels, conference rooms, convention centers, restaurants, schools, hospitals, shopping malls, stores all kinds of sound reinforcement places.

Dante PoE Speakers (EAGLE)

Model: DP6

Product Features:

- SSA Inverter Technology
- Support POE Power Supply
- Dante Audio Network
- Lossless Transmission

Model	DP6
Frequency Response	70Hz -21kHz
Output Power	15.4-30W (determined by poe power supply device output)
Coverage Angle	180°
Sound Pressure Level	118dB
Sensitivity	92dB(1w@1m)
Speaker	woofer 6.5 inch x 1, tweeter 1 inch imported agnesium film + horn drive standard impedance
Standard Impedance	8Ω
System Composition	6.5", two-way, two-unit built-in crossover single-driver full-range box

Product Introduction

EAGLE DP6 network PoE speaker can achieve the power supply, audio transmission and signal control functions through only a low-cost Category 5 twisted pair cable. It can get power supply from any 802.3at standard Ethernet switch, using SSA inverter technology, making low power design, but also can get excellent sound quality of efficient amplification.

Dante PoE Column (EAGLE)

Model: ZDP34、ZDP38

Product Features:

- Built-in Amplifier Module
- Full-Function Control of Software
- Dante Transmission
- Software for Full Function Control
- Multifunctional DSP Chip
- Gain/silence Level Function

Model	ZDP34	ZDP38
System Composition	4x2.75" Full-range unit	8x2.75" Full-range unit
Frequency Response	220Hz-20kHz，-3dB	220Hz-20kHz，-10dB
Sensitivity	84dB(300Hz-18kHz)	88dB(300Hz-18KHz)
Maximum SPL	107dB/113dB(Peak)	110dB/116dB(peak)
Power	30W Power Rating	60W Rated Power
Directivity	Vertical 40°/horizontal 140°	Vertical 25°(2kHz-16kHz)(±10°) Horizontal 160°(1kHz-4kHz)(±20)
Cabinet	Square Box, Laminate	Square Box, Laminate
Installation	2 Hanging Points	8 Hanging points
Surface Treatment	Black Polyurethane Coating	Black Polyurethane Coating
Steel Mesh	Black Plastic Powder Coating, 1.0mm Steel Perforated Panel	Black Plastic powder coating, 1.0mm steel perforated plate
Connector	RJ45 Network Port	RJ45 Network Port
System Type	Lacquered Wood Speaker	Lacquered Wood Speaker
Speaker Size(W×D×H)	104x130x333mm	104x130x610mm

Product Introduction

The speaker has a built-in amplifier module, supports PoE power and Dante transmission, single channel, 60W @ 4 ohms; all functions can be controlled by Dante-Controller on a PC or Mac process, and has an integrated DSP containing gain control, mute control, level, temperature display, 5-band equalizer control and limit control functions.

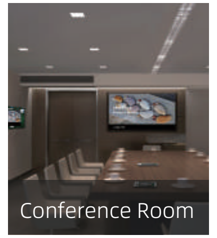
Multimodal Conference Host (YH)

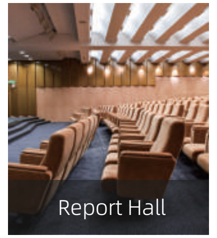
Model: YH3000、YH3000Z、YH3000F

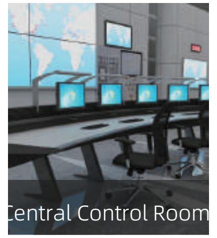
The system adopts high-performance digital processing technology, with line power "hot-swappable" function, six microphone management mode and super anti-mobile phone interference capability, never generate noise when calling, support automatic detection, host software upgradeconference unit audio input can be realized separately or mixed output, can realize the partition output audio function. Up to 120 units can be supported. Optional feedback suppression function, provide network-based voice transcription interface. 3.5inch color capacitive touch screen, can be centralized control and management of all conference functions, Chinese \ English menu display.

Product Features:

- High-Speed RISC Processor
- CD Grade Sound Quality Effect
- Color Capacitive Touch
- Custom Unit Numbering
- Support Camera Tracking
- Treatment Antistatic


Conference Room


Report Hall


Central Control Room


Large Conference Room

(Suitable for small, medium and large conference rooms,lecture halls, central control rooms, multi-functional halls, and other scenes ...)

Model	YH3000	YH3000Z	YH3000F
Input Channel	Noise Gate \ Gain \ Phase \ Mute \ Time Delay and 31-segment Parametric Equalization, etc.		
Output Channel	High and Low pass filters\delay\phase\mute\voltage limiter and 10-segment parametric equalization, etc.		
Input Interface	4-channel 6PDIN+1-channel XLR (with Phantom Power)		
Output Interface	3 way Phoenix Terminal + 1 way XLR		
Control Panel	3.5-inch Capacitive Touch Screen, resolution 480 * 320, Chinese and English		
Camera Tracking Function	Support		
Maximum Supported Units	4-channel Conference unit input port, support up to 120 units		
Balanced and Unbalanced Inputs	Support		
Balanced and Unbalanced Outputs	Support		
Material	Aluminum Alloy panel and sheet metal body, built-in anti-static treatment, can resist 8000V static electricity		
Enclosure Size	483x356x89（mm）		
Communication Port	RJ45*1 / RS232*2 / RS485*1		
Automatic Gain Attenuation	2 mics on, 1dBu reduction; 4 mics on, 2dBu reduction		
Master Gain Control	40×1dB and off (mute)		
Sampling	48K		
Frequency Response	20Hz-20kHz		
Operating Power Supply	AC100-240V,50-60Hz		
Power Consumption	Stand-alone 25W, maximum with load 350W		
Feedback Cancellation (AFC)	/	/	√
Voice Transcription Interface	/	√	/

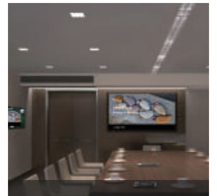
Multimodal Conference Unit (NAJA)

Model：S303P、S303、S301P-L、S301-L

The system unit adopts multimodal connection, providing three output interfaces of XLR analog, Dante digital and six-core shielded wire, with super-cardioid pickup effect and the best pickup distance of 40-80CM; built-in high-performance embedded CPU, 3.5-inch color touch screen display, supporting functions of sign-in \voting \voting \rating, etc., supporting dynamic dial display time, accepting short messages, sending commands to the background send commands to the backstage to achieve front and backstage communication.

Product Features:

- High Fidelity Microphone
- Powered " Hot-swapping" Function
- Color Touch Screen
- Supercardioid Pickup Effect
- Support Camera Tracking
- ID Flexible Number


Conference Room


Courts


Multifunctional Hall

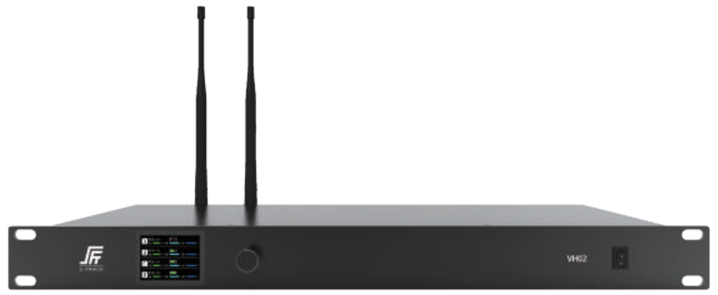

Large Conference Room

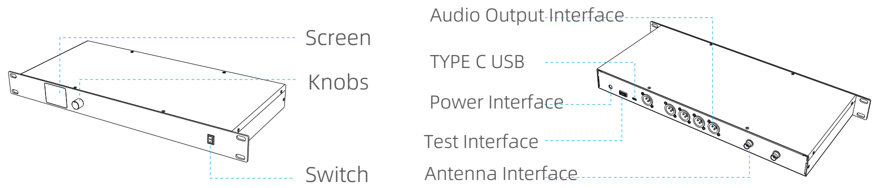
(Suitable for small, medium and large conference rooms, courts, multi-function rooms, and other scenes ...)

Model	NAJA S303P	NAJA S301P-L	NAJA S303	NAJA S301-L
Pickup Head	14 14 mm Capacitive Gold-plated Film Microphone			
Maximum Sound Pressure Level	110dB(3%T.H.D.@1kHz，0dBSPL=2x10Pa)			
Camera Tracking Function	Camera Tracking Function			
Equivalent Noise Level	＜25dBSPL（A）			
Overload Harmonic Distortion	<1%			
Headphone Load	32Ω-2KΩ			
Headphone Volume	10mW			
Headphone Output Connector	Ø 3.5mm Stereo jack			
Base Size	149mmx94mmx55mm			
Pointing Characteristics	Supercardioid			
Camera Tracking Function	Support			
Unit Interface	6 Core Shielded Wire, DANTE, XLR Analog Output	6 Core Shielded Wire	6 Core Shielded Cable, DANTE, XLR Analog Output Connector	6 Core Shielded Wire
Frequency Response	20Hz-20kHz;125(-6dB)-14kHz(-3dB)	100-16kHz	20Hz-20kHz;125(-6dB)-14kHz(-3dB)	100-16kHz
Sensitivity	-34dB±2dB	-40dB	-34dB±2dB	-40dB
Maximum Power Consumption	＜3.5W	＜2.5W	＜2.5W	＜1W
Mic Pole Size	232mmx37mmx25mm	42cm	232mmx37mmx25mm	42cm
Optimal Pick Up Distance	40-80CM	10-50CM	40-80CM	10-50CM
Control Screen	3. 5 Inch color touch screen		/	
Screen Resolution Dial Function	640x480		/	
Dial Functions	Check-in\voting\voting\rating and other functions, support dynamic dial display time, a variety of dial styles can be selected		/	

All Digital Wireless Conference System (Host)

Model: VH02

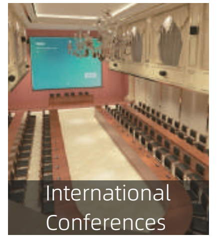


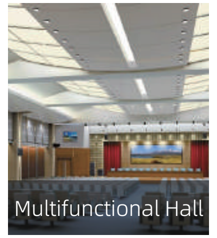


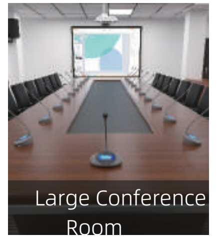
This product uses the fourth generation of digital U-band wireless communication band, with stronger bandwidth and communication speed, the front panel is equipped with a 2-inch LCD display, with encoder, you can quickly realize the system function settings; panel can real-time display unit communication status, volume signal, transmission channel signal strength; system supports three role definitions, support 999 wireless unit system capacity, compatible with hand-held, lavalier, seat mic a variety of units, 5ms ultra-low latency, support for four units to speak at the same time two speaking modes, FIFO, self-locking mode.

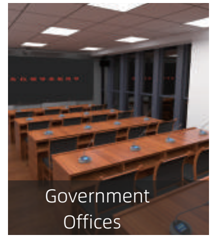
Product Features:

- Numerical Communication
- Digital Encryption
- Stable and Non-Crosstalk
- Ultra-Long Distance Transmission
- Ultra-Low Latency
- Automatic Frequency Cutting


International Conferences


Multifunctional Hall


Large Conference Room

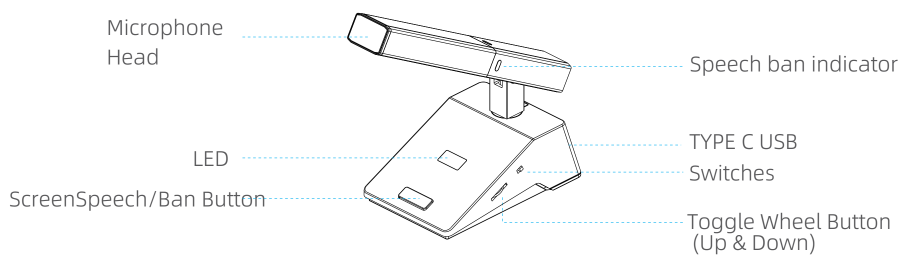

Government Offices

(Widely used in the National People's Congress at all levels, government agencies, international conferences, group board of directors, high star hotel conference room multi-function hall, lecture hall and other places.)

Model	VH02
Multiple Speech Modes	FIFO Mode, SelfLock Mode
Software Settable Roles	Chairman, VIP Column, General Column. Maximum configuration of 999 terminals
Support Transmitter	Handheld Terminal, Seated Microphone Terminal, Fanny Pack Terminal
Communication Mode	Voice Channel: Digital Communication, Control Channel: Digital Communication
Modulation Mode	Voice Channel: Pi/4 DQPSK, Control Channel: GFSK
Transmission Frequency Band	668MHz ~ 698MHz
Bandwidth	30MHz
Maximum Offset	±45KHz
Transmission Distance	100m
Receiving Sensitivity	Offset equal to 25KHz,S/N>60dB at 5dBv Input
Combined S/N	>96dB
Combined T.H.D	<0.2%@1kHz
Frequency Response	30Hz~18kHz
Host Power Supply	DC12V/12W
Size / Weight	480*200*45mm/1200g
Support Unit Capacity	999 Wireless Unit System Capacity
USB Interface	Support
Power Supply Interface	12V2A Power Input Interface
Communication Frequency Band	Communication Frequency Band of the Fourth Generation Digital U-band Wireless

All Digital Wireless Conference System (Seat-Mike Type Terminal)

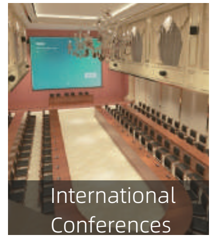
Model: NAJA VH204

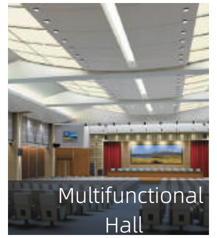


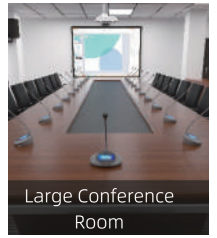
Product built-in rechargeable lithium battery, provide charging interface, can continue to speak for 10 hours or continuous 24 hours of standby work, and the use of unique encryption technology to ensure that the meeting information security; equipped with LCD display, real-time display call volume, call status, circuit status, with independent gain adjustment, speech timing function, support three role definition: the chairman \VIP \ listed.

Product Features:

- Numerical Communication
- Digital Encryption
- Stable and Non-Crosstalk
- Ultra-Long Distance Transmission
- Ultra-Low Latency
- Quick Match


International Conferences


Multifunctional Hall


Large Conference Room


Government Offices

(Widely used in the National People's Congress at all levels, government agencies, international conferences, group board of directors, high star hotel conference room multi-function hall, lecture hall and other places.)

Model	NAJA VH204
Electricity Supply	USB 5V + Built-in Lithium Battery
Endurance (Hours)	> 20h(Lithium Battery)
Weight	860g
Size	150mm x 98mm x 250mm(MAX)
Type	Back-pole Type Capacitor Mac (Operating Voltage 1.1~10V)
Diaphragm	3um Mylar Film, Gold Plated
Directionality	Supercentric Type ＞13dB (135° pointing)
Frequency Response Range	100Hz ~ 2KHz
Sensitivity	-30.0±3dB，31.6mV/Pa (0dB=1V/Pa@1KHz, RL=2.2kΩ，Vs=2.7V DC)
Output Impedance	< 2.2KΩ
Maximum Sound Pressure Level	130dB （1% T.H.D. @ 1KHz, 0dB SPL=2x10^-5 Pa)
Equivalent Noise Level	25dB, (A-weighted)
Communication Mode	U-band Radio Digital Communications
Modulation Method	Pi/4 DQPSK
Transmission Frequency Band	668MHz ~ 698MHz
Bandwidth	30MHz
RF Output	<18dBm
Adopted Frequency	48KHz，24KHz Optional
Transmission Distance	90 meters (Actual range with RF signal absorption, reflection, interference related)
Frequency Response	<2dB（20Hz~20KHz）
Signal-to-noise ratio S/N	>97dB
Distortion Degree T.H.D	< 0.03%（@1KHz）
Antenna	600MHz Built-in,
Encryption	Digital Encryption, Depending on software version

All Digital Wireless Conference System (Handheld Terminal)

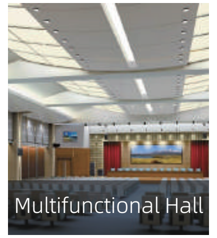
Model: NAJA VH203

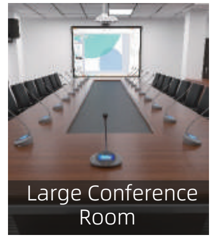
Product Features:

- Digital Communications
- Digital Encryption
- Stable and Non-Crosstalk
- Ultra-Long Distance Transmission (80 m)
- Ultra Low Latency (4 ms)
- Hybrid

The product adopts hybrid circuit technology, can use two 1.5V ordinary batteries or two 3.7V, 14500 rechargeable lithium batteries, can continue to speak for 10 hours or continuous 24 hours of standby work, and the use of unique encryption technology to ensure that the meeting information security; equipped with LCD display, can real-time display call volume, call status, with independent gain adjustment, speech timing function, support two Role definition: VIP \ listed.


International Conferences


Multifunctional Hall


Large Conference Room


Government Offices

(Widely used in the National People's Congress at all levels, government agencies, international conferences, group board of directors, high star hotel conference room multi-function hall, lecture hall and other places.)

Model	NAJA VH203
Power Supply	14500 Lithium battery (3.7V) *2 or No. 5 battery (1.5V)*2 [Hybrid power technology]
Endurance (Hours)	> 10h(Li-ion battery) / 3.5h(AA No.5 battery)
Operating Temperature Range	-18℃ ~ 50℃ (Battery characteristics may limit this range)
Weight	150g, No Battery
Size	L=245mm; R=38mm
Material	Cast Aluminum + Engineering Plastic
Storage Temperature Range	-29℃ ~ 74℃ Cast Aluminium + Engineering Plastic
Type	Moving Coil Microphone
Sensitivity	-41±2dB RL=0.68KΩ Vs=1.5V(1KHz 0dB=1V/Pa)
Frequency Response Range	20Hz ~ 16kHz
Operating Voltage Range	1.0V-10V
Maximum Sound Pressure	115dB S.P.L
Output Impedance	Max. 0.68KΩ1KHz (RL=0.68KΩ)
Communication Mode	U-Band Radio Digital Communications
Modulation Method	Pi/4 DQPSK
Transmission Frequency Band	668MHz ~ 698MHz
Bandwidth	30MHz
RF Output	<18dBm
Adopted Frequency	48KHz, 24KHz Optional
Transmission Distance	80 m (Actual range with RF signal absorption, reflection, interference related)
Frequency Response	<2dB (20Hz~20KHz)
Distortion Degree T.H.D	< 0.03% (@1KHz)
Antenna	600MHz, Integrated Single-Band Spiral Type (Built-In)
Encryption	Digital Encryption, Depending on software version

Multimodal Conference System

System Introduction

Management Model

Quantity Limitation Mode\First-in-first-out Mode

Speech Queuing Mode\Voice Activated Mode

Application Speech Mode \ Free Speech Mode

Multimodal Output Method

The system unit Is Provided with a 6-core Shielded Cable

with a T-handle Connection DANTE Digital output interface

Provide XLR analog output interface

Terminal Management

With Speaker ID Setting and Speech Time Setting

Single or Mixed Output and Partition Output Audio Function

Centralized Control and Management of All Conference Functions in the Host

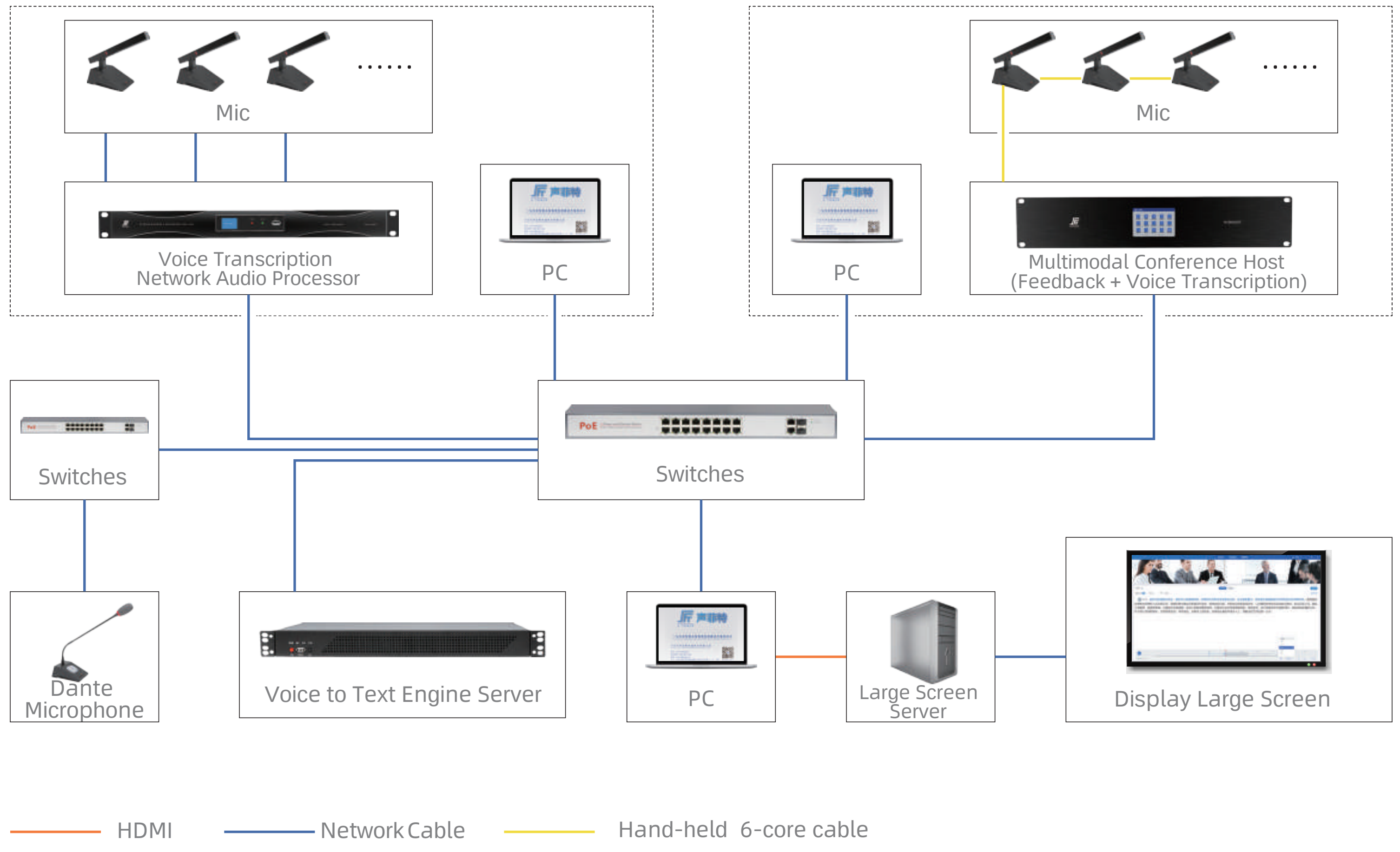
Function Control

Input channels have noise gate\gain\phase\mute\delay and 31-segment parametric equalization

The output channels have high and low pass filters\delay\phase\mute\voltage limiter and 10-segment parametric equalization, etc.

System can customize the conference unit number

Voice Transcription Conference System



System Introduction

Intelligent Voice Transcription

Mandarin Speech Recognition and Transcription Accuracy of over 95% on average

Recognition Speed <200ms

Support Hotword Management/Sensitive Word Management/ Tone Word Filtering

System Network Architecture

Connected Using Pure Network Architecture

Space-Separated Deployment

Centralized Control of Multiple Meeting Rooms

Function Management

Audio Playback Audio Word Comparison Support

DSP Audio Processing / Auto Mixing Console / Comprehensive Matrix Mixing Function

Feedback Cancellation (AFC)/Echo Cancellation (AEC)/Noise Cancellation (ANC) Module 2 Concurrent Real-Time

Transcription Audio Streams with 1 Offline Transcription Voice Transcription Interface Function

System Features

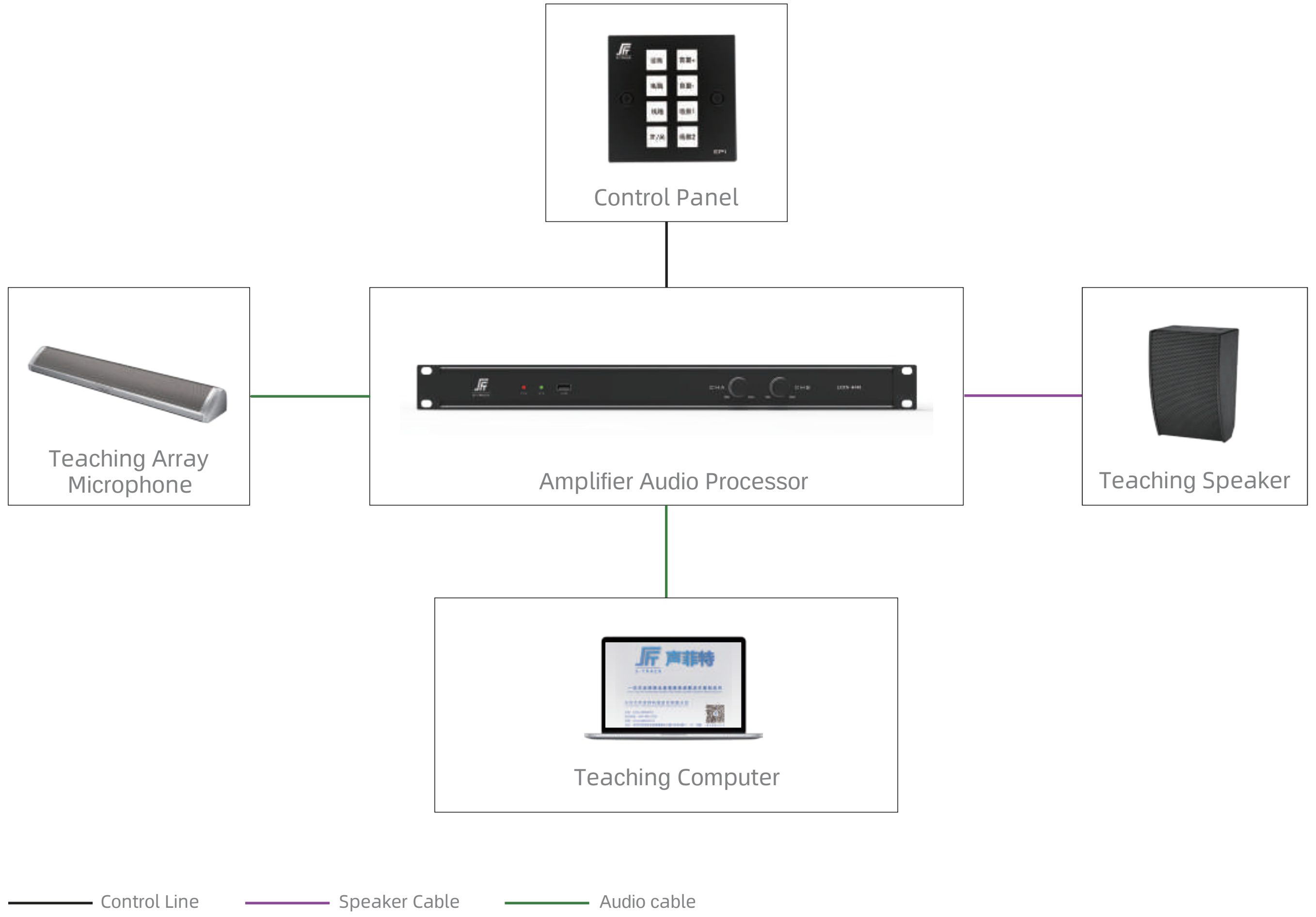
Humanized Interactive Interface, Enhancing Meeting Efficiency

in All Aspects Real-time Subtitle Casting, with 2 Casting Modes

Adopt Offline Server to Avoid Leakage of Conference Information

Support Multi Role Partition Transcription Seat / Support Dante Microphone Signal Input

Teaching Amplification System



System Introduction

Ultimate Voice

The use of the array of suspended microphone pickup, support 5-8 meters long-distance pickup, and equipped with a control panel, can realize the system volume, mode switching and other operations, easy to use, a key operation.

Professional Algorithms

Support automatic echo cancellation, auto feedback suppression, auto gain and auto noise cancellation, no bottom noise, no whistling, with natural, clear sound, back row students listening to the class without obstacles.

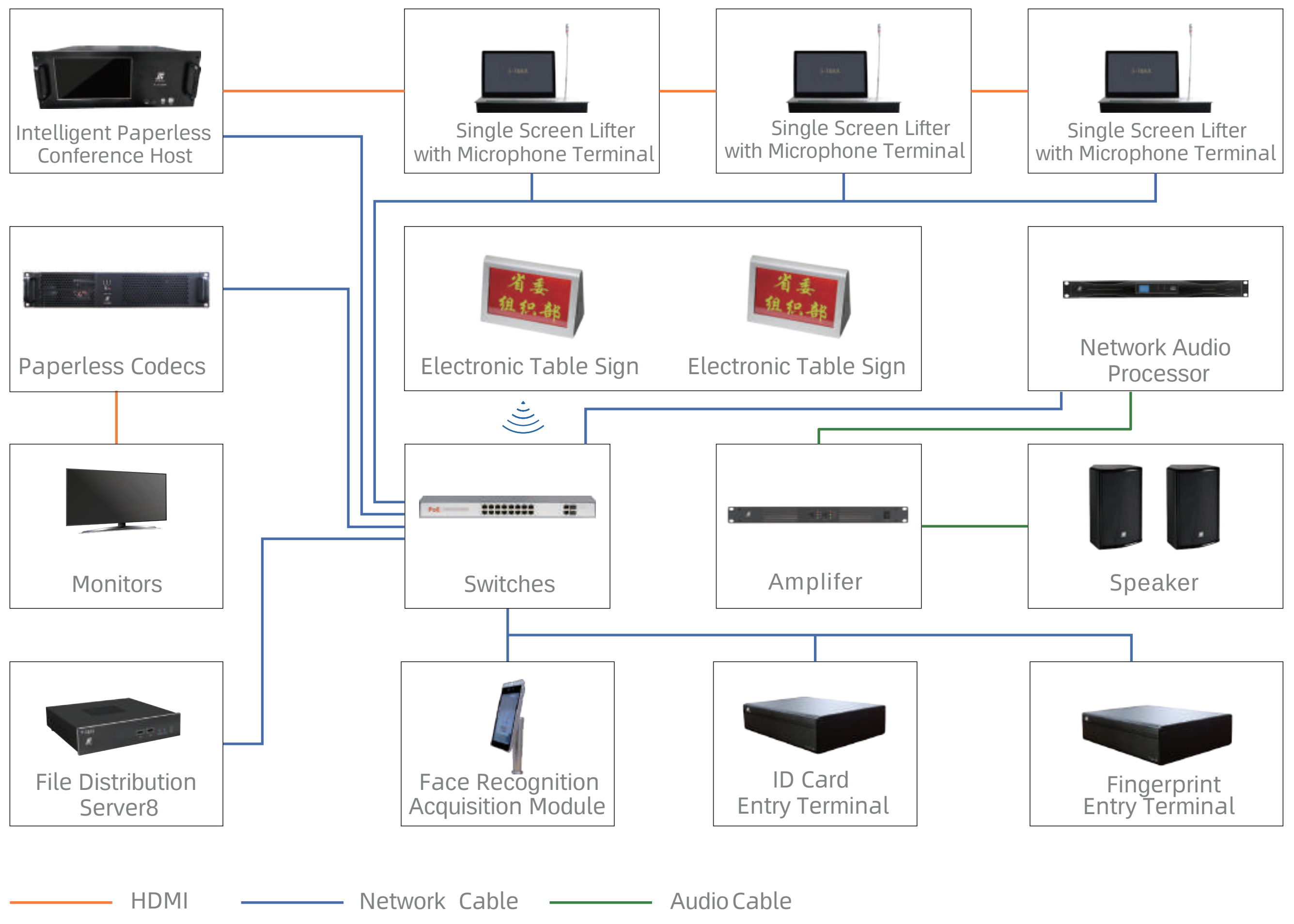
Simple Configuration

The system configuration is streamlined, only the processor, hanging microphone and speakers and other audio-visual equipment can be correctly connected to the selected location, easy and simple to build, strong compatibility, can be docked to the vast majority of recording equipment and network video equipment.

Amplifier + Audio Processing

The audio processor integrates a dual-channel 150W digital amplifier module to achieve two-in-one amplifier and audio processing functions.

Paperless System



System Introduction

Conference Application

Conference voting/sign-in function/issue setting/document distribution sharing/audio/video interaction/content broadcasting/background specialization of nameplate/electronic whiteboard/signal control/centralized control and other special editable and flexible functions.

Extensive Interface

In addition to convenient access to video and audio conferencing functions, the S-TRACK intelligent voice transcription system can also be installed to help smarten the office.

System Application

S-TRACK paperless meeting system is a system for party and government organs, large enterprises and institutions to conduct important meetings.

System Features

Combined with modern communication technology to achieve easy to read, print-free, easy to save documents in the meeting, greatly enhance the efficiency of the meeting, and do to respond to the slogan of environmental protection and low-carbon call.