FS-FHSS Wireless Simultaneous Interpretation System

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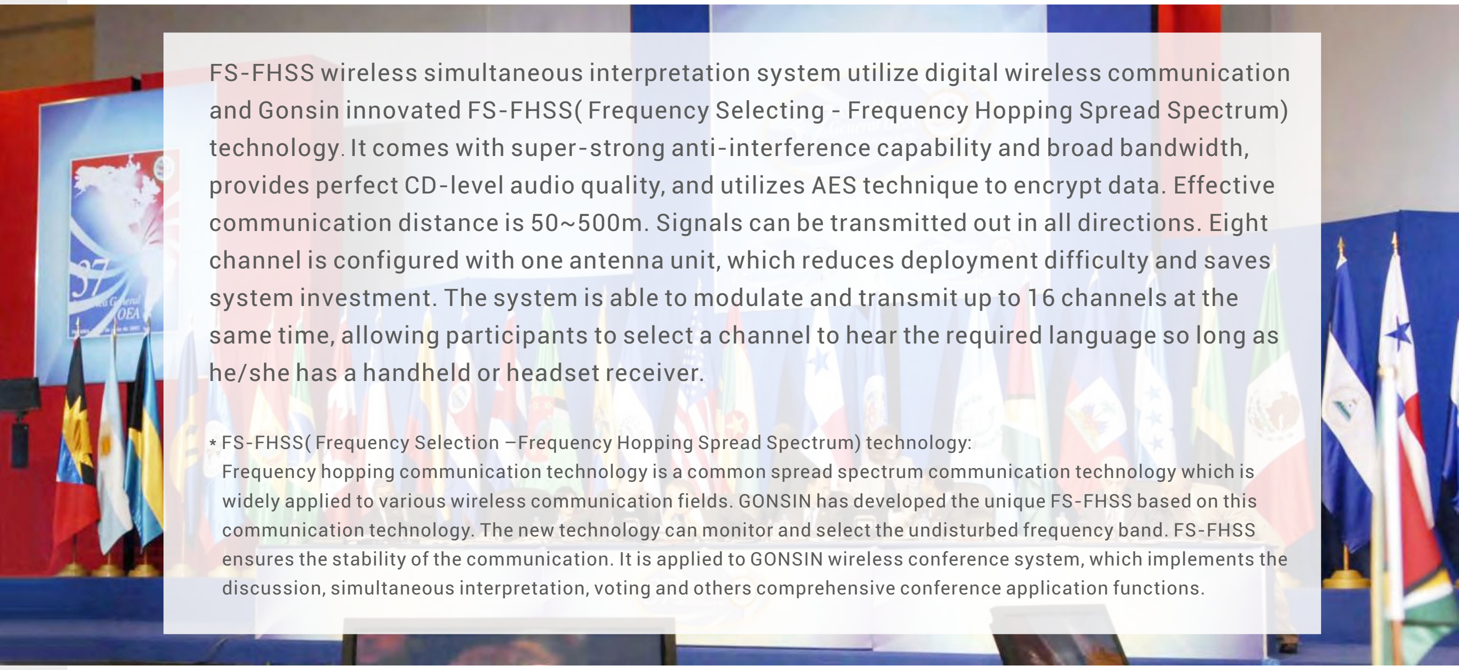

FS-FHSS wireless simultaneous interpretation system utilize digital wireless communication and Gonsin innovated FS-FHSS(Frequency Selecting - Frequency Hopping Spread Spectrum) technology. It comes with super-strong anti-interference capability and broad bandwidth, provides perfect CD-level audio quality, and utilizes AES technique to encrypt data. Effective communication distance is 50~500m. Signals can be transmitted out in all directions. Eight channel is configured with one antenna unit, which reduces deployment difficulty and saves system investment. The system is able to modulate and transmit up to 16 channels at the same time, allowing participants to select a channel to hear the required language so long as he/she has a handheld or headset receiver.

* FS-FHSS(Frequency Selection –Frequency Hopping Spread Spectrum) technology:
Frequency hopping communication technology is a common spread spectrum communication technology which is widely applied to various wireless communication fields. GONSIN has developed the unique FS-FHSS based on this communication technology. The new technology can monitor and select the undisturbed frequency band. FS-FHSS ensures the stability of the communication. It is applied to GONSIN wireless conference system, which implements the discussion, simultaneous interpretation, voting and others comprehensive conference application functions.

Features

- Gonsin innovated FS-FHSS (Frequency Selecting-Frequency Hopping Spread Spectrum) technology
- Highly flexible for conference places, suits various conference places including outdoor/ indoor/ temporary/fixed conference place
- AES for data encryption, preventing malicious interference and listening
- Super-strong anti-interference capability, being immune to any light source, wireless communication equipment, and signal jammer
- Adjustable transmission power, the effective communication distance could reach from 50m to 500m.
- Unlimited system capacity within the effective transmission scope, suitable for small/medium/large/super-large conferences
- All-direction transmission, allowing participants to move within the effective transmission scope
- Max number of simultaneous interpretation channels in one system: 8; in support of cascading, max number of simultaneous interpretation channels: 32
- FS-FHSS Simultaneous Interpreter System can be used with the GONSIN 10000N system to extend the conference discussion function.

Wireless Simultaneous Interpretation Receiver TC-FSJ04B/08B/12B/16B/32B

Basic functions

- GONSIN innovated FS-FHSS technology
- CD-level sound quality
- HOST ID setting by manually or automatic upgrading, support Auto-tracking by Host ID
- Support up to 32 channels(up to 1 floor channel+31 interpretation channels); Five models optional:4/8/12/16/32 channels
- LED display, human-oriented human-machine interaction
- Lithium battery/AAA battery
- Real-time battery level detection and display, low battery alarm
- Grouping (encryption) function; max number of groups: 255; no interference between the groups
- Dynamic display of such information as signal strength, battery capacity, and selected channel on the OLED screen
- Signal received from all directions
- Support unified shut-down function, reduce workload greatly
- Lightweight receiver; in combination with a headphone to allow users to hear an interpreter
- Clear voice along different independent channels
- Volume and audio channel is adjustable by key pressing
- Low power consumption, allowing battery to consecutively run for 16 hours
- Automatic turn-off function when no signal is receiving for 10 minutes, unified server shutdown function;
- Earphone detection, receivers will auto shut-down if without earphone after 2 minutes
- Number of receivers within the signal coverage area is unlimited;
- Allow users to move within the signal coverage area
- Earphone (single- or double-side) and headphone for selection
- Centralized charging by the charging case, 60 receivers can be charged by one charing case.

Technical Parameters

Modulation mode	FS-FHSS technology
Frequency range	2.4-2.5GHz
Encryption method	AES
System Max capacity	Unlimited receivers within signal distance
Grouping function	up to 255 groups
Sensitivity	-121dBm
Earphone jack	3.5mm jack stereo
OLED screen	35×10mm
Battery display	real-time display, low power alarm
Battery	lithium battery/ 2 AA batteries
Battery service time	Lithium Battery:20 hours NANFU AA(LR6):16 hours (Different AA batteries have different results)
Package	portable aluminum case
Dimension(L×W×H)	136×50×21mm
Weight	136g (with a lithium batterty) 94g (without battery)
Working temperature	0~45°C
Storage temperature	-20~50°C

Wireless AP WAP-30CI

Basic functions

- FS-FHSS frequency selecting-and-hopping technology, built-in 2.4G wireless frequency sweeping software, with spectrum monitoring and screening functions; support intelligent hopping of carrier frequency, and can automatically select a clean frequency point when the system is working
- Adopts full digital wireless communication technology, FSK binary digital frequency modulation/demodulation technology, which can resist interference, multipath fading, and is not affected by light and infrared rays, so it can be used stably in the WLAN environment
- Built-in conference transceiver embedded software. Adopt GSSC scalable audio codec algorithm, Huffman encoding and 128-bit AES encryption technology to ensure the privacy of communication, avoid eavesdropping and malicious interference
- Built-in high gain antenna, stable wireless signal, coverage range 50~500 meters(adjustable)
- Support wall/stand/ceiling/desktop installation, adaptable to all types of venues

Technical Parameters

Color	White
Network Interface	1 RJ45 interface with private power supply protocol
Power Interface	1 DC interface, supports 24V DC power supply
Simultaneous Interpretation Channels	8 (support expansion)
Dimension	260mm×260mm×68mm
Weight	1.2Kg

Audio Extender DCS-AE10

Basic functions

- It can realize the conversion of D/A.
- The number of channels of a single audio extender is 10, and it can be cascaded up to 32 channels.
- Support the independent or unified setting of DA/AD mode for each channel, and the mode status of each channel can be viewed through the OLED panel.
- Through the DA mode, the digital audio of the GONSIN10000N system can be output to the FS-FHSS wireless simultaneous interpretation and IR simultaneous interpretation system to achieve system expansion.
- Through AD mode, FS-FHSS wireless simultaneous interpretation, IR simultaneous interpretation system, and other analog audio signals can be input into the GONSIN10000N system to achieve unified system management.
- The AD mode can be used to convert analog audio into digital audio to meet the cascading use of the analog audio discussion system and the intelligent ASR system, to expand the voice transcription function.
- 2 audio extenders cooperate to realize 10-channel audio long-distance transmission
- Support host/slave working mode.
- In the host working mode, it can manage digital interpreter terminals, and can access to the digital interpreter terminal to meet the requirements of digital translation room management with FS-FHSS wireless simultaneous interpretation system and IR simultaneous interpretation system.
- In the slave working mode, it needs to be used with the GONSIN10000N server, which manage the digital translation room. Applied to occasions where GONSIN10000N server is used imultaneously with other simultaneous interpretation systems.
- The electrical level of each channel can be adjusted independently or uniformly through the panel knob.

Technical Parameters

Audio frequency response	20Hz~20kHz
SNR	>96 dBA
Channel isolation	>85 dB
Total harmonic distortion	<0.05%
Audio interface	11 groups of lotus interfaces
Display	OLED (61mm×37mm)
Working temperature	-0~45°C
Storage temperature	-20~50°C
Dimensions (L×W×H)	485×325×90mm
Installation	desktop or use 19 inches cabinet
Weight	8kg

Congress Server

GONSIN10000N

Basic Functions

- Designed with black tempered glass panel with navigational knob, OLED screen displays system status, number of speakers, speaking modes and other information, providing multi-language menu.
- Support clock and date synchronized display. Support automatic/manual assignment of terminal ID. Support remote diagnosis and remote upgrade.
- Adopts full-digital audio technology, supports 48KHz audio sampling frequency, audio frequency response up to 20Hz 20kHz, supports automatic mixing, volume adjustment, equalization adjustment, sensitivity adjustment and other functions.
- Based on LINUX operating system platform development, support wired conference terminal and wireless conference terminal access to the same conference server. The default maximum number of simultaneous openings is 5 (wired)/4 (wireless). Support multi-chairman application and unlimited number of chairmen/vice-chairmen.
- Provide 2 network interfaces with PoE power supply capability. Support connection to devices such as wireless access points, switches, routers, and computers. Support SSID wireless network management, managing 128 wireless access points and 650 wireless terminals. Support one-click shutdown function to uniformly power off all wireless terminal devices.
- Provide a terminal interface with 4 channels and 24V power supply capability, support three connection modes: hand-in-hand, star and hybrid. Support dual-chain connection (specified model), which can provide two sets of mutually independent network communication system, and any one equipment failure or line failure of the whole system will not affect the work of the system.
- Provide 4-channel packet audio output interface. The system has a terminal grouping function, which can be used as a large system or split into up to 4 independent systems to meet the application requirements of space merging/splitting.
- Support external audio signal input, which can be connected to alarm audio, broadcast audio or third-party conference system, simultaneous interpretation system, public address, etc.
- Provide 1 channel USB interface on the front panel. Insert U disk (FAT32 format) to support manual/automatic audio recording, and display the recording status. When used with acquisition card and software, it can realize the function of recording and video.
- Provide connection interface for simultaneous interpretation equipment, which can be connected to the interpreter terminal and wireless simultaneous interpretation access point. The audio frequency response of the simultaneous interpretation channel can reach 20Hz-20kHz, and it can realize 6-channel (wireless conference terminal)/32-channel (wired conference terminal)/32-channel (wireless simultaneous interpretation receiver) simultaneous interpretation function (including original sound channel).
- Provide 1 channel AES67 network audio interface (specified model). The system complies with AES67 network audio standard, is compatible with Dante protocol, and supports channel management through Dante Controller software.
- Support multiple speaking modes: FIFO, Auto, Operator, VOX, PTT, Request, Time-limited (requires software integration)
- FIFO (First In First Out): When the number of activated microphones exceeds the system's limit, pressing the speaking key will activate the microphone and deactivate the earliest activated microphone.
- Auto: When the number of activated microphones exceeds the system's limit, pressing the speaking key will restrict microphone activation. Microphones can only be activated after previously activated microphones are deactivated.
- "Operator: When the number of activated microphones exceeds the system's limit, pressing the speaking key will put the microphone into a waiting state. After the activated microphone turns off, the applied

microphones will be activated automatically"
- VOX: When the microphone detects sound, it will automatically activate. When no sound is detected for a set period, the microphone will automatically deactivate. Sensitivity and automatic deactivation time can be adjusted. Manual activation and deactivation by a key are also supported.
- PTT(Push-to-Talk): Holding down the speaking key on a terminal will activate the microphone. Releasing the speaking key will deactivate the microphone.
- Request: When press the delegate terminal, the microphone enters a request state. Upon approval by the chairman terminal, microphones in the request state will automatically activate. The chairman/vice-chairman terminals can speak without needing approval.
- Time-limited (Requires Software Integration): When the microphone is activated, it enters a countdown state. When the countdown reaches zero, the microphone will automatically deactivate. The speaking countdown time can be adjusted.
- The conference server has a dual-server hot backup function. One conference server can be set to backup mode, when the main server fails, the backup server automatically takes over the work to ensure uninterrupted operation of the system.

- Management software with dual-computer hot backup function. One computer can be set as backup mode, when the main operating computer fails, the backup computer will automatically take over the work to ensure uninterrupted operation of the system.
- Built-in 8 in 4 out video matrix, support the connection of SD cameras. Support external HD matrix, switcher, HD camera, support multi-camera cascade. Support automatic camera tracking function, support panoramic view.
- With the upper computer management software (additional purchase required), it can realize the functions of meeting registration, meeting voting, discussion control and so on.
- With chairman screen tablet (additional purchase required), it can realize meeting registration, meeting voting, discussion control and other functions, to meet the application needs of portable control.
- With the call service management software/tablet (additional purchase required), it can check the service applications submitted by terminal devices and carry out service processing (it needs to be used with terminals with call service function).
- With Automatic Speech Recognition System (need to purchase separately), it can realize system audio role separation, voice transcription, voice translation, subtitle display, minutes output and other functions.
- With voice control system (additional purchase required), it can realize voice control function. Realize equipment control, task execution and other operations through voice speech. Support multiple scene quick call. Support wake-up word and command word customization.
- Support Web page management and client software management. Support access to the central control system, visual media interactive system, intelligent conference management platform, which can realize remote control, status monitoring, fault reminder and other functions.
- The power supply is designed with 100V~240V wide voltage input and built-in linear voltage regulator circuit for over-voltage protection. Built-in power management embedded software can monitor the voltage status of the equipment, reduce the equipment failure caused by voltage fluctuation, and realize all-weather protection.

Technical Specifications

Installation	Desktop or 19' cabinet installation
Device Interface	Balanced audio input interface (XLR) × 1 Balanced audio output interface (XLR) × 1 Unbalanced audio output interface (RCA) × 2 Packet audio output interface (Phoenix terminal) × 4 Control interface (RS-232) × 1 Debugging interface (RS-232) × 1 Camera tracking interface (RS-485) × 1 Terminal interface (RJ45) × 4 Network interface (RJ45) × 2 Recording interface (USB) × 1 Video input interface (RCA) × 8 Video output interface (RCA) × 4 AES67 network audio interface (RJ45) × 1
Audio Frequency Response	20Hz-20KHz
Signal-to-Noise Ratio	>96dBA
Channel Isolation	>85dB
Dynamic Range	>94dB
Total Harmonic Distortion	<0.05%
Output Impedance	>1KΩ
Maximum Audio Output	LINE OUT (unbalanced): +20 dBu BLANCE OUT (balanced): +20 dBu MIXER OUT (balanced): +20 dBu AUDIO OUT 1 to 4 (balanced): +20 dBu
Maximum Power Consumption	300W
Power Supply	AC100-240V, 50/60Hz
Dimensions (L×W×H)	483mm×430mm×90mm
Weight	8.9Kg

Simultaneous Interpretation Audio Converter

U-BOX08A/08B

Simultaneous interpretation audio converter is a GONSIN self-developed product. With this device, GONSIN FS-FHSS simultaneous interpretation system, IR simultaneous interpretation system and wired simultaneous interpretation system can be connected seamlessly, and compatible with other brands' simultaneous interpretation systems, which forms a new large various complete solution of interpretation systems, saves customers'investment in expanding the current system and ensures a stable performance without affecting on product functions and sound quality.

Basic Functions

- High compatibility, realize the interconnection and communication between multiple simultaneous interpretation systems(including other brands) to form a large scale system.
- Easy to extend, supporting output and input of simultaneous interpretation system
- Flexible interface, supporting signal converting between DB25 audio interface and other interfaces
- Connecting the interpreter console to simultaneous interpretation server to realize interconnection among multiple simultaneous interpretation systems, including GONSIN equipments and other brands'.
- 3 channels of DB25 audio interfaces and 8 channels of phoenix terminal interfaces, signal is converted from different interfaces for compatible use
- Support 8 channels of audio output to meet the demand of audio recording, press release, live video broadcast and other applications.
- Support 8 channels of audio input, can be connect to the external line-in audio to realize interconnection among multiple simultaneous interpretation systems
- Support DB25 cable hand-in-hand connection, to realize 16 channels of simultaneous interpretation

Technical Specifications

Power supply	24V, powered by central control unit
Channel number	8
Interfaces	2 DB25 Male interfaces 1 DB25 Female interface 8 groups of phoenix terminals
Dimensions(L x W x H)	200mm×185mm×30mm
Weight	590g
Operating temperature	0～45℃
Storage temperature	-20~50℃

Interpreter Console IC-2032N


Basic Functions

- In compliance with IEC60914 international standard
- The design of physical buttons / knobs is reasonable , which is easy to operate; The buttons / knobs are sensitive with good touch, accurate operation response, long service life and without mechanical button sound
- With fully digital audio technology, built-in high-speed CPU
- Interpreter console, conference terminal, power HUB and conference server are all connected by network cables. Flexible cabling reduces project workload and makes large-scale construction projects easier
- It supports hand-in-hand connection, star connection and mixed connection. The system is highly stable and reliable. The breakdown of one terminal won't influence the other terminals , and the malfunction of one terminal cabling will not affect the whole system, ensuring the system stability
- Automatic restoration function: supports hot plug
- Supports 48kHz audio sampling freq, acoustic frequency response can be up to 20Hz to 20kHz
- Built-in high-performance digital audio processor(DSP), helpful to remove the useless low frequency and avoid low-frequency acoustical shock, which is better to improve the clarity of sound
- Support 32 channels simultaneous interpretation(including the floor channel)
- Support direct interpretation and relay interpretation
- 7-inch TFT LCD screen to display setting information ,input and output channel information
- With channel selection button, long press to quickly increase / decrease the channel, short press to increase / decrease the channel, convenient to select channels
- 7 input channels can be preset, with corresponding shortcut keys and input channel occupation indicator
- 3 output channels can be preset , with corresponding shortcut keys. When all interpreter consoles are closed, the output channel can be automatically switched to the floor channel
- Standard cardioid directional electret microphone with indicator ring (On or Off) . Multiple soft gooseneck microphone stems can be selected (standard: 410mm, optional:310mm/510mm)
- Pluggable microphone stem; During the meeting recess, it can be disassembled for equipment maintenance
- Built-in 2W loudspeaker with volume switch; Can play the floor channel or interpreted channel language
- With the same channel interlock function, when the output channel is preset, the occupied channel will skip

automatically. Channel output can be set to interlock mode and free mode

- The number of interpreter console in the same room is not limited
- Through software setting, multiple interpreter consoles can be set up in a single language channel. When the interpreter console in the same language channel turns on the microphone, the microphone of the previous interpreter terminal will be automatically turned off
- MUTE key to enable a brief muting of the microphone for cough cut
- Speaking Speed remind function, the interpreter can press the “slow” key to remind the speaker to slow down the speaking speed (The microphone’s indicator ring or screen will flash)
- Left and right headphone output jack (3.5 mm×2), connect the headset to listen to the floor or interpreted channel language, with left and right selection buttons; headphone volume is adjustable
- Left and right microphone input jack (3.5 mm×2), can be connected to external microphone, with left and right selection buttons, to meet different usage scenarios
- Left and right recording output jack (3.5 mm×2), can be connected to recording device, with left and right selection buttons, the output audio depends on the channel listened by the interpreter console
- With message function, text message can be sent to interpreter terminal through management software
- With call service function, interpreters can call the administrator to provide help through the interpreter console
- Setting the interpreter console information can be locked by password, and the interpreter can use the setting function through password and IC card
- No need independent system connection, it can be mixed connected with all conference terminals
- Highly resistant to mobile phone interference
- Every interpreter comes with an unique serial number and the conference system supports automatically or manually assigning ID to the corresponding equipment
- Interpretation timing function (timing unit: hour/minute)

Technical Specifications

Installation Way	Desktop
Connection Way	Network Cable
Display Specifications	7-inch TFT color display
Frequency Range	20Hz~20kHz
Sensitivity	-46 dBV/Pa
Equivalent Noise Level	20dBA(SPL)
Max. SPL	125dB(THD<3%)
THD	<0.05%
SNR	≥96dB
Microphone Type	Cardioid directional electret
Directivity	0°/180°: >20 dB (1 kHz)
MIC Stem Length	410mm (standard); 310mm/510mm (optional)
Input Impedance	2KΩ
Headphone Output Jack	3.5 mm jack×2
Microphone Input Jack	3.5 mm jack×2
Headphone Load	>16Ω
Headphone Volume	10mW
Built-in Loudspeaker	2W/8Ω
Channel Number	32 channel (including the floor channel)
Power supply	24V
Max. Power Consumption	7W
Working Temperature	0~45°C
Storage Temperature	–20~50°C

Interpreter Console Management System V2.1.0

Basic Functions

- ♦ Support interpreter console online search and numbering
- ♦ Numbering the interpreter console corresponding with interpretation room; the number of interpreter consoles in the same interpretation room is not limited
- ♦ Support name setting for the interpreted channel and corresponding language
- ♦ Support interpreter console locking mode, including password locking and IC card locking
- ♦ Support interpreter console to set SOLW function permissions
- ♦ Support interpreter console call service function(on / off)
- ♦ Support interpreter console playback function, turn on (5/10/15/30 seconds)/turn off setting
- ♦ Support the setting of the output channel after the microphone is turned off; can choose mute or monitor output
- ♦ Support interpreter console date and time synchronization
- ♦ Support 32 effective channels

Charging Case

GX-60/60B/60C

Features

- ♦ Intelligent charging management
- ♦ Independent LED indicators for charging status
- ♦ Portable aluminum storage case
- ♦ Anti-misplug design for charging port

Basic Functions

- ♦ Aluminum storage flight case for charging & storage
- ♦ Accommodates up to 60 units
- ♦ Independent management for each charing port, it will automaticlly charge the unit with low battery, it will stop charging if the unit is fully charged to ensure the security
- ♦ Independent LED indicators for charging status
- ♦ Charging time: 2.5 hours
- ♦ Anti-misplug design for charging port
- ♦ GX-60 is suitable for charging IR simultaneous interpretation system receivers
- ♦ GX-60B is suitable for charging TC-FSJ04/08/12/16 wireless simultaneous interpretation receivers, BJ-W5 voting units and BJ-W5I voting and interpretation units
- ♦ GX-60C is suitable for charging TC-FSJ04B/08B/12B/16B wireless simultaneous interpretation receivers.

Technical Specifications

Power input	AC100V~240V, 50/60Hz
Capacity	60 units
Charging time	2.5 hours
Charging indicator	LED indicator
Dimension (L*W*H)	680mm×430mm×245mm
Weight	11 Kg
Operating temperature	0~45℃
Storage temperature	-20~50℃

Power Hub

HUB-P150S

Basic Functions

- ♦ Adopt a black metal chassis design, with working status indicator, supporting hidden installation or cabinet installation, supporting to install 2 devices side by side (requires installation accessories).
- ♦ Provide 7-way network interfaces with 24V power supply capability to connect conference terminal, power HUB, conference server (need to connect according to the interface instruction). Supports three connection modes: ring hand-held, star-type and hybrid.
- ♦ Adopt distributed power supply management and support synchronized switching with the conference server. Built-in auto-detection mechanism, can automatically detect voltage, current and temperature status, support overload, high temperature automatic protection.
- ♦ Power supply adopts 100V-240V wide voltage input design, built-in linear voltage regulator circuit for over-voltage protection. Built-in power management embedded software can monitor the equipment voltage status, reduce the equipment failure caused by voltage luctuation, and realize all-weather protection.
- ♦ More than 16 channels need to be equipped with a hub to power the interpreter console.

Technical Specifications

MTBF	⩾ 10000 hours
Installation	Hidden installation or 19" cabinet installation (installation accessories required)
Device Interface	Terminal interface (RJ45) × 4 Cascade interface (RJ45) × 2 Server interface (RJ45) × 1
Max. number of mounts per unit	65 conference terminals (monitoring off) / 5 desktop paperless display terminals (The number of carriers varies according to the terminal model and line length)
Maximum Power Consumption	150W
Power Supply	AC100~240V, 50/60Hz
Dimension (L×W×H)	238mm×165mm×45mm
Weight	2kg

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Accessories

Monaural Headphone TC-D1

Color	Black ring with embedding metal
Type	Monaural mono headphone
Interface	3.5mm
Impedance	40Ω±10%
Frequency reponse range	7 Hz - 24KHz
Voltage	0.4V
Input power	8-50mW
Distortion rate	<5%
Sensitivity	L102dB±3dB

Stereo Headphone TC-D3

Color	Black
Type	Stereo headphone
Interface	3.5mm
Impedance	32Ω±15%
Frequency reponse range	20-20KHz
Voltage	0.8V
Input power	30-50mW
Distortion rate	<5%

Interpreter Headset TC-D4

♦ Adopt ergonomic and lightweight design to reduce the pressure on the head when wearing the headset, making it comfortable to wear

♦ Support manual telescopic head beam to meet most human head shapes

♦ It has a high-definition noise-cancelling microphone, which can effectively prevent the interference of the surrounding environment, accurately pick up the sound, and record clearly

♦ Adopt 40MM classic speaker to drive the sound unit, so that the sound of the earphone is clear and lossless, bringing the original sound experience

♦ Adopt strong anti-violence material, resistant to pulling and folding

♦ Using long plastic straw, 150-degree angle adjustment, convenient for recording and pickup

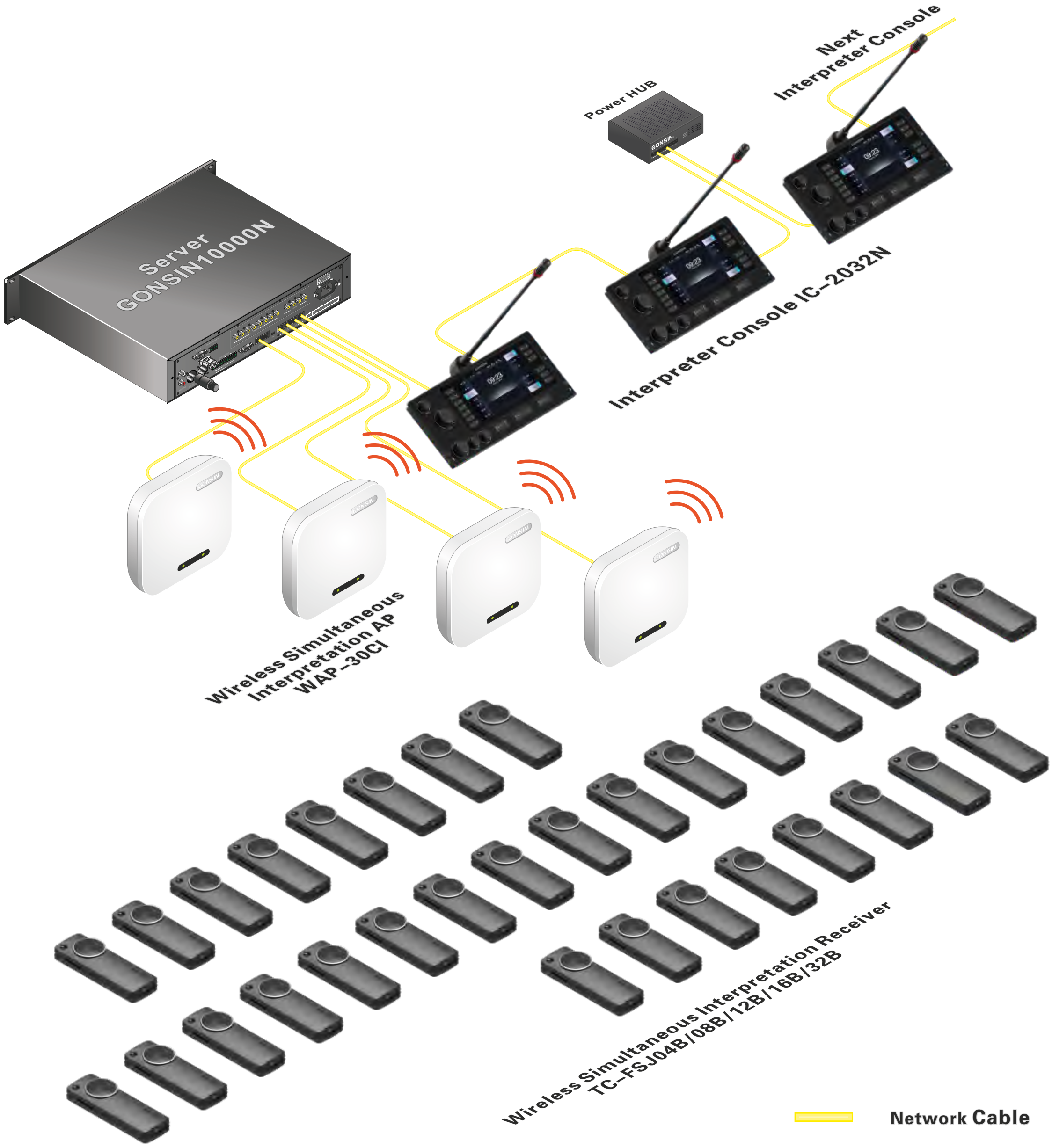
Product Weight	180g
Microphone Specifications	
MIC	Diameter 6.5mm
Impedance	2200Ω
Sensitivity	-58dB±3dB
Directionality	Directivity
S/N ratio	Above 60dB
Frequency range	20Hz-20kHz
Operating voltage	4.5V

Headset Specifications	
Trumpet	Diameter 40mm
Impedance	32Ω
Sensitivity	105±3dB
Rated power	⩾100mW
Frequency response	15Hz-30KHz
Wire length	2.2M
Volume adjustment	Gear tuning
Port	Dual 3.5mm ports

System Components

DESCRIPTION	MODEL	4 channels	8 channels	12 channels	16 channels	32 channels
Server	GONSIN10000N	1 PCS	1 PCS	1 PCS	1 PCS	1 PCS
Wireless SimultaneousInterpretation Access Point	WAP-30CI	1 PCS	1 PCS	2 PCS	2 PCS	4 PCS
Wireless Simultaneous InterpretationReceiver	TC-FSJ04B	√	-	-	-	-
	TC-FSJ08B	√	√	-	-	-
	TC-FSJ12B	√	√	√	-	-
	TC-FSJ16B	√	√	√	√	-
	TC-FSJ32B	√	√	√	√	√
Interpreter Console	IC-2032N	√	√	√	√	√
Power Hub	HUB-P150S	-	-	-	-	√
Stereo Headphone	TC-D3	√	√	√	√	√
Single Earphone	TC-D1	√	√	√	√	√
Interpreter Headset	TC-D4	√	√	√	√	√
Charging Case	GX-60C	√	√	√	√	√

System Configuration



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