Introduction to video conferencing products

PRODUCT INTRODUCTION

iFreecomm Technology Co.,Ltd.

Trademark statement

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Other trademarks, product names, service names, and company names appearing in this manual and the products described in this manual are owned by their respective owners.

catalogue

Company Profile 1
1.1 Development History 2
1.2 Company Qualifications 4
2 Product Introduction 6
2.1 Video conferencing terminal product 6
2.1.1 ONE-M8L Yishi Video Intelligent Collaboration Terminal 6
2.1.2 ONE-M3Box-S Ultra HD Video Conference Terminal 9
2.1.3 ONE-M3Mini-S Ultra HD Video Conference Terminal 12
2.1.4 ONE-M1S Ultra HD Video Conference Terminal 16
2.2 Terminal supporting products 20
2.2.1 CP50S One-click Smart Screen Casting Device 20
2.2.2 PM50X conference controller 22
2.2.3 AM50S omnidirectional digital array microphone 24
2.2.4 AM30WS Wireless Omnidirectional Digital Microphone 26
2.3 Camera products 28
2.3.1 VENUS31S Ultra HD Camera 28
2.4 Audio Products 31
2.4.1 JA-AM129S non-inductive sound reinforcement ceiling array microphone 31
2.4.2 JS-ST5E conference fusion host 33
2.4.3 JS-ST5E matching microphone 36

3 Case 41

Disclaimers 43

Company Profile

IFREECOMM was founded in Shenzhen, China, in 2008. It is a national-level "little giant" enterprise specializing in precision, fine, unique, and innovative technologies, and a military-civilian integration enterprise. Its core team comes from Tsinghua University and Huawei, and has been dedicated to the research of multimedia communication technology, audio and video technology, and products since the 1990s. It is one of the first pioneers in audio and video multimedia in China.

IFREECOMM has been dedicated to the technology and product development in the field of "security and confidentiality + video communication" for many years, accumulating profound technology and experience in areas such as audio and video encoding and decoding, intelligent processing, network adaptability, AI development and application, multi-service integration, and localization in China. We are committed to providing customers with excellent and innovative all-media, fully connected products and solutions.

Relying on years of accumulation in the audio and video field, IFREECOMM has collaborated with top scientists in the Chinese industry, achieving breakthroughs in the field of ultra-low-latency remote video transmission and becoming a technology leader in this area. Leveraging this technology, applications have been implemented in military unmanned combat and robotics, gaining recognition from numerous customers and Chinese national entities.

1.1 Development History

Time	Important events and development milestones
2025	We have launched the fourth-generation ultra-high-definition video conferencing terminal and the second-generation fully adaptable MCU based on H.323/SIP. Leveraging our technological prowess, we have successfully been selected as a recognized unit of the Guangdong Provincial Doctoral Innovation Station.
2024	The launch of the "Lingmu" Xunying ultra-low-latency codec based on ultra-low-latency remote video transmission technology, as well as the "Lingmu" panoramic intelligent perception module, has been applied in military unmanned combat and robotics, and has been recognized by numerous customers and Chinese national units.
2023	The company has been selected as a "Little Giant" enterprise in China's national-level specialized, refined, unique, and innovative enterprises.
2022	We have developed ultra-low-latency remote video transmission technology, achieving an end-to-end transmission delay of less than 40 milliseconds. This technology was successfully selected for the 2022 "Huiyan Action" by the Comprehensive Planning Bureau of the Equipment Development Department of the Central Military Commission of China, marking the beginning of military-civilian industry cooperation in the market.
2021	We are pleased to officially introduce the IFREECOMM brand's audio conference series products.
2020	Launched the UCLink cloud video conferencing comprehensive solution based on the SFU framework, catering to various industry needs and SDK customization requirements. Additionally, became a member of the Information Technology Innovation Alliance and launched the first information technology innovation product based on the Feiteng CPU.
2018	We officially launched our self-developed domestic "Yi" video communication operating system (iFOS) and introduced the world's first ultra-high-capacity, fully adaptable ultra-high-definition MCU product. In the same year, we incorporated artificial intelligence technology and launched our third-generation ultra-high-definition video conferencing product based on H.323/SIP.
2017	We have released an interactive educational recording and broadcasting solution, officially entering the field of smart education.
2016	We will organize the "Mou Ding Zhong Hua" national tour exhibition and establish the "Jiefibang" channel cooperation platform.
2015	The company completed its shareholding system reform and officially changed its name to "iFreecomm Technology Co.,Ltd.." In the same year, it ventured into the cloud video business and established a strategic partnership with "Guizhou on the Cloud" in China. It also launched the MCV5000, a multifunctional integrated high-definition video conferencing terminal.
2014	The company launched its second-generation hardware video conferencing system based on H.323/SIP, and set a new strategic direction, transitioning from being a product provider to a "multimedia converged communication solution provider".
2013	The launch of the industry's first Rongzhi UCM full-service integration system solution marks the company's expansion towards the direction of "mobile multimedia converged communication".
2011	Launched the "Longshi" integrated product series based on H.323/SIP and multimedia communication integration solutions, which have been successfully applied to Baosteel Group in Shanghai, China.
2008	iFreecomm Technology Co.,Ltd.. was officially established. In the same year, it launched the first generation of hardware video conferencing system based on H.323/SIP.

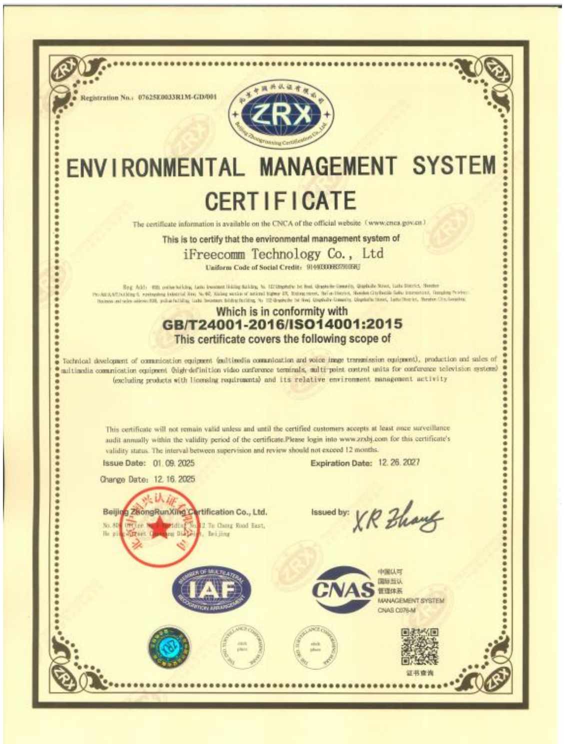
Time	Important events and development milestones
2005	The predecessor of the company was established and began to engage in the research and development of multimedia communication products.

1.2 Company Qualifications

Awards

- 188 proprietary intellectual property rights
- 172 core patents
- 68 copyrights
- 49 industry honorary titles

Quality management



Environmental

Industry honorary titles

Invention Patent

2 Product Introduction

2.1 Video conferencing terminal product

2.1.1 ONE-M8L Yishi Video Intelligent Collaboration Terminal

Overview

ONE-M8S is a new generation of H.265 ultra-high-definition video conferencing terminal from Jieshi Feitong. With its powerful dual 4K ultra-clear picture quality, it brings you an extraordinary ultra-high-definition conference experience. The split design supports multiple specifications of cameras and audio equipment, making it the best choice for the construction of large and medium-sized video conferencing rooms, meeting the needs of various scenarios in the conference room. The standard H.323/SIP communication protocol perfectly solves the needs of various remote audio and video communication interactive applications.

Product Features

- Ultimate Ultra HD Experience: With support for H.265 4K@30fps, combined with beauty enhancement technology and intelligent front-end and back-end video image processing, it delivers unparalleled audio and video quality, featuring vivid and smooth visuals with realistic colors. While saving 50% of bandwidth, it presents you with a stunning and clear world;
- Heshì: Without the involvement of an MCU, the local terminal transmits images inputted by multiple cameras to the remote end in a multi-screen format, satisfying the need for multi-viewing angles while saving bandwidth for customers;
- Multiple microphone access methods: Supports three audio input solutions: digital array microphone, USB microphone, and analog microphone, adapting to different conference scenarios and meeting more audio needs;
- Local recording: Supports recording to USB storage disks, with plug-and-play functionality, ensuring the security and confidentiality of meeting records;
- Superior network adaptability: In the case of severe network packet loss, the V-Link proprietary technology can be activated. Packet loss within 20% has no impact on the system, and meetings can still proceed normally with 30% packet loss, effectively solving the problems of network jitter and packet loss;
- Built-in MCU: The optional built-in MCU multi-point control unit function supports 1+8 networking, meeting the simple networking and rapid meeting needs of small and medium-sized enterprises, and enhancing the communication and collaboration efficiency of small meetings;
- Localization and Security: Supports H.460 firewall traversal technology to address secure connections between public and private networks. Supports H.235, SRTP protocols, AES-128 encryption algorithms, and China's domestically developed controllable chips, providing comprehensive communication security;

Technical Specifications

Protocol and standard	Multimedia framework protocol	ITU-T H.323, IETF SIP V2
	Video codec protocol	H.265, H.264 High Profile, H.264, H.263+, H.263
	Audio codec protocol	G.711a/u, G.722, G.722.1, G.722.1C, G.726, G.729, AAC-LD, Opus, SILK
	Dual-stream protocol	ITU-T H.239, BFCP
	Network transmission protocol	TCP/IP, DHCP, SSH, HTTP, HTTP SwithSSL/TLS, RTP, RTCP, RFC3261, RFC3264, RFC2190, RFC3407, RFC2833, RFC4585 (RTP/AVPF), SNTP, ARP
	Other agreements and standards	H.221, H.224, H.225, H.235, H.241, H.245, H.281, H.350, H.460, T.140, DTMF
Video features	Mainstream image resolution	4KP30, with a minimum bandwidth of 2048Kpbs 1080p, with a minimum bandwidth of 512Kpbs 720p, minimum bandwidth 384Kbps
	Demo resolution	800x600, 1024x768, 1280x1024, 1280x720, 1920x1080, 3840x2160
	Dual-flow capability	Supports up to dual-channel 4K P30 encoding and decoding
	Other image characteristics	Supports display modes such as PIP and POP, with image effect enhancement and pre- and post-processing of images
Audio characteristics	3A treatment	Automatic Echo Cancellation (AEC), Automatic Gain Control (AGC), Automatic Noise Suppression (ANS)
	Other audio characteristics	Support lip sync and local sound amplification
Stability and safety	bandwidth	64Kbps~8Mbps
	Network adaptability	V-Link boasts superb error correction and packet loss resistance, achieving 30% packet loss resistance for video and 70% for audio
	security features	H.235AES media encryption, SRTP media encryption, conference access password, conference control password, HTTPS, whitelist
interface	Audio input interface	2 x XLR inputs, 2 x RCA inputs, 1 x MIC ARRAY (digital microphone), 1 x HDMI audio input, 1 x USB
	Audio output interface	4 x RCA linear outputs, 1 x HDMI audio output, 1 x USB
	Video input interface	3 x HDMI (supporting up to 4K60), 2 x NET VIDEO
	Video output interface	3 x HDMI (supporting up to 4K60)
	serial port	2 x RS232, supports infrared transparent transmission

	network interface	2 x LAN:10M/100M/1000M Base-T; 2 x RJ45 10M/100M/1000M Base-T (POE powered); 1 x Wi-Fi: 802.11b/g/n/ax, supports STA and AP modes;
	USB port	2 x USB3.0 (supports USB flash drives, USB microphones, etc.)
	Digital microphone cascade	support
Business characteristics	Dual-screen dual display, dual-screen different display, triple-screen triple display	
	Combined view: dual-screen, triple-screen, quad-screen	
	Wi-Fi wireless access	
	Support banners and subtitles	
	Support recording on third-party platforms with RTMP streaming	
	Support USB local recording	
	Support wired digital microphone six-level cascade access (optional AM50S)	
	Support USB wireless screen projector (optional CP50S)	
	Supports conference control via tablet (optional PM50X)	
	Supports optional built-in MCU functionality (with optional License authorization), supporting up to 8 concurrent H.265 1080P30 accesses	
	Support third-party integration and development of API interfaces	
	Maintenance function	Built-in Web management, supports configuration export and backup
		Support OLED display of IP and registration number on the front panel of the terminal
		Support automatic hibernation and wake-up mechanisms for terminals
		Event log recording and exporting
		Supports IP network diagnosis, audio diagnosis, and factory setting recovery
size	446mm*282mm*54mm (without foot pads and interfaces)	
power supply parameters	AC voltage	100～230V 50/60Hz
	power	≤80W
runtime environment	temperature	0°C~40°C (working state), -40°C~70°C (non-working state)
	relative humidity	10% to 80% (in working state), 0% to 95% (in non-working state)
	altitude	≤5000 meters above sea level

2.1.2 ONE-M3Box-S Ultra HD Video Conference Terminal

Overview

ONE-M3Box is a new generation of H.265 ultra-high-definition video conferencing terminal from Jieshi Feitong. With its powerful dual 4K ultra-clear picture quality, it brings you an extraordinary ultra-high-definition conference experience. The split design supports multiple specifications of cameras and audio equipment, meeting the needs of various scenarios in the conference room. The standard H.323/SIP communication protocol, as well as wireless BYOM conferencing, perfectly solves the needs of various remote audio and video communication and interactive communication applications.

Product Features

- Ultimate Ultra HD Experience: With support for H.265 4K@30fps, combined with beauty enhancement technology and intelligent front-end and back-end video image processing, it delivers unparalleled audio and video quality, featuring vivid and smooth visuals with realistic colors. While saving 50% of bandwidth, it presents you with a stunning and clear world;
- Wireless BYOM Conference: Supporting wireless BYOM functionality, it allows for cloud conferencing using AV devices (cameras, microphones, speakers) connected to the conference terminal via a computer, while also supporting wireless screen mirroring. Seamless switching between BYOM meetings and terminal meetings is more convenient;
- Multiple audio access methods: Supports three audio input solutions: digital array microphone, USB microphone, and analog microphone, adapting to different conference scenarios and meeting more audio needs;
- Local recording: Supports recording to USB storage drives, with plug-and-play functionality, ensuring the security and confidentiality of meeting records;
- Superior network adaptability: In cases of severe network packet loss, the V-Link proprietary technology can be activated. Packet loss within 20% has no impact on the system, and meetings can still proceed normally with 30% packet loss, effectively addressing issues of network jitter and packet loss;
- Built-in MCU: Optional built-in MCU multi-point control unit function, supporting 1+8 networking, meets the simple networking and rapid meeting needs of small and medium-sized enterprises, and improves the communication and collaboration efficiency of small meetings;
- Security and confidentiality: It supports H.460 firewall traversal technology to ensure secure connections between public and private networks. It also supports H.235, SRTP protocols, AES-128 encryption algorithms, and China's self-controlled chips, providing comprehensive protection for communication security;

Technical Specifications

Protocol and	Multimedia	ITU-T H.323, IETF SIP V2

standard	framework protocol	
	Video codec protocol	H.265, H.264 High Profile, H.264
	Audio codec protocol	G.711a/u, G.722, G.722.1, G.722.1C, G.726, G.729, AAC-LD, Opus, SILK
	Dual-stream protocol	ITU-T H.239, BFCP
	Network transmission protocol	TCP/IP, DHCP, SSH, HTTP, HTTP SwithSSL/TLS, RTP, RTCP, RFC3261, RFC3264, RFC2190, RFC3407, RFC2833, RFC4585 (RTP/AVPF), SNTP, ARP
	Other agreements and standards	H.221, H.224, H.225, H.235, H.241, H.245, H.281, H.350, H.460, T.140, DTMF
Video features	Mainstream image resolution	4KP30, with a minimum bandwidth of 2048Kpbs 1080p, minimum bandwidth 512Kpbs 720p, minimum bandwidth 384Kbps
	Demo resolution	800x600, 1024x768, 1280x1024, 1280x720, 1920x1080, 3840x2160
	Dual-flow capability	Dual-channel 4KP30 encoding and decoding
	Other image characteristics	Supports display modes such as PIP and POP, with image effect enhancement and pre- and post-processing of images
Audio characteristics	3A treatment	Automatic Echo Cancellation (AEC), Automatic Gain Control (AGC), Automatic Noise Suppression (ANS)
	Other audio characteristics	Support lip sync and local sound amplification
Stability and security	bandwidth	64Kbps~8Mbps
	Network adaptability	V-Link boasts superb error correction and packet loss resistance, with 20% packet loss resistance for video and 30% packet loss resistance for audio
	security features	H.235 AES media encryption, SRTP media encryption, conference access password, conference control password, HTTPS, whitelist
interface	Audio input interface	1 x 3.5mm linear input, 1 x XLR input, 1 x MIC ARRAY (digital microphone), 2 x HDMI audio inputs, 1 x USB
	Audio output interface	1 x 3.5mm linear output, 2 x HDMI audio outputs, 1 x USB
	Video input interface	2 x HDMI (supporting up to 4K60), 1 x USB (supporting up to 1080P30fps), 1 x NET VIDEO
	Video output interface	2 x HDMI (supporting up to 4K60)
	serial port	1 x RS232, supports infrared transparent transmission
	network interface	1 x LAN:10M/100M/1000M Base-T; 1 x MIC ARRAY/TOUCH: 10M/100M/1000M Base-T, POE powered; 1 x Wi-Fi: 802.11b/g/n/ax, supports STA and AP modes;
	USB port	2 x USB3.0 (supports USB flash drives, USB microphones, etc.)

<table>
<tr><td></td><td>Digital microphone cascading</td><td>support</td></tr>
<tr><td rowspan="15">Business characteristics</td><td colspan="2">Dual-screen dual display, dual-screen different display</td></tr>
<tr><td colspan="2">Wi-Fi wireless access</td></tr>
<tr><td colspan="2">Support banners and subtitles</td></tr>
<tr><td colspan="2">Support recording on third-party platforms with RTMP streaming</td></tr>
<tr><td colspan="2">Support USB local recording</td></tr>
<tr><td colspan="2">Support wired digital microphone six-level cascade access (optional AM50S)</td></tr>
<tr><td colspan="2">Support USB wireless screen projector (optional CP50S)</td></tr>
<tr><td colspan="2">Support conference control tablet control (optional PM50X)</td></tr>
<tr><td colspan="2">Supports optional built-in MCU functionality (with optional License authorization), supporting up to 8 concurrent H.265 1080P30 accesses</td></tr>
<tr><td colspan="2">Support third-party integration and development of API interfaces</td></tr>
<tr><td rowspan="5">Maintenance function</td><td>Built-in Web management, supports configuration export and backup</td></tr>
<tr><td>Support OLED display IP on the front panel of the terminal</td></tr>
<tr><td>Support automatic hibernation and wake-up mechanisms for terminals</td></tr>
<tr><td>Event log recording and exporting</td></tr>
<tr><td>Supports IP network diagnosis, audio diagnosis, and factory settings restoration</td></tr>
<tr><td>size</td><td colspan="2">430mm*200mm*45mm (without foot pads and connectors)</td></tr>
<tr><td rowspan="2">power supply parameters</td><td>AC voltage</td><td>100～230V 50/60Hz</td></tr>
<tr><td>power</td><td>≤65W</td></tr>
<tr><td rowspan="3">runtime environment</td><td>temperature</td><td>0°C~40°C (working state), -40°C~70°C (non-working state)</td></tr>
<tr><td>relative humidity</td><td>10% to 80% (in working state), 0% to 95% (in non-working state)</td></tr>
<tr><td>altitude</td><td>≤5000 meters above sea level</td></tr>
</table>

2.1.3 ONE-M3Mini-S Ultra HD Video Conference Terminal

Overview

ONE-M3Mini is a new generation of integrated ultra-high-definition video conferencing terminal from Jieshi Feitong. It is equipped with a built-in 4K60fps ultra-high-definition camera, array microphone, and ultra-high-definition codec, providing users with an ultra-high-definition experience and minimalist intelligent control, suitable for rapid deployment in small and medium-sized conference rooms. With standard H.323/SIP communication protocols and wireless BYOM conferencing, it perfectly meets various remote audio and video communication and interactive communication application needs.

Product Features

- Integrated design: Integrated fanless design, featuring a built-in 4K60fps camera, array microphone, and ultra-high-definition codec, with convenient connectivity and simple installation;
- Ultimate Ultra HD Experience: With support for H.265 4K30fps encoding and decoding, combined with beauty enhancement technology and intelligent front-end and back-end video image processing, it provides unparalleled audio and video quality, featuring vivid and smooth playback with realistic colors. While saving 50% of bandwidth, it presents you with a stunning and clear world;
- Wireless BYOM Conference: Supports wireless BYOM functionality, enabling mainstream cloud conferencing directly using high-quality audio and video devices (cameras, microphones, speakers) on computers, and supports wireless screen mirroring. Seamless switching between BYOM conferences and terminal conferences is more convenient;
- Multiple microphone access methods: Supports three audio input solutions: digital array microphone, USB microphone, and analog microphone, adapting to different conference scenarios and meeting more audio needs;
- Local recording: Supports recording to USB storage drives, with plug-and-play functionality, ensuring the security and confidentiality of meeting records;
- Built-in MCU: Optional built-in MCU multi-point control unit function, supporting 1+8 networking, meets the simple networking and rapid meeting needs of small and medium-sized enterprises, and improves the communication and collaboration efficiency of small meetings;
- Superior network adaptability: In cases of severe network packet loss, the V-Link proprietary technology can be activated. Packet loss within 30% has no impact on the system, and meetings can still proceed normally with 40% packet loss, effectively addressing issues of network jitter and packet loss;
- Security and confidentiality: Supports H.460 firewall traversal technology to address secure connections between

public and private networks. Supports H.235, SRTP protocols, AES-256 encryption algorithms, and China's self-reliant and controllable chips, providing comprehensive protection for communication security;

Technical Specifications

Protocols and standards	Multimedia framework protocol	ITU-T H.323, IETF SIP V2
	Video codec protocol	H.265, H.264 High Profile, H.264, H.263, H.263+
	Audio codec protocol	G.711a/u, G.722, G.722.1, G.722.1C, G.726, G.729, AAC-LD, Opus, SILK
	Dual-stream protocol	ITU-T H.239, BFCP
	Network transmission protocol	TCP/IP, DHCP, SSH, HTTP, HTTP SwithSSL/TLS, RTP, RTCP, RFC3261, RFC3264, RFC2190, RFC3407, RFC2833, RFC4585 (RTP/AVPF), SNTP, ARP
	Other protocols and standards	H.221, H.224, H.225, H.235, H.241, H.245, H.281, H.350, H.460, T.140, DTMF
Video features	Mainstream image resolution	4KP30, with a minimum bandwidth of 2048Kpbs 1080p, minimum bandwidth 512Kpbs 720p, with a minimum bandwidth of 384Kbps
	Demo resolution	800x600, 1024x768, 1280x1024, 1280x720, 1920x1080, 3840x2160
	Dual-flow capability	Supports up to dual-channel 4K30fps encoding and decoding
	Other image characteristics	Supports display modes such as PIP and POP, image effect enhancement, and image pre- and post-processing
Built-in camera	sensor	1/2.5 inch, CMOS, effective pixels: 8.51 million
	scanning method	line by line
	Lens focal length	12×,f=4.4mm～52.8mm,F1.8~F2.6
	resolution	Supports up to 4K60fps
	Horizontal Field of View	80.8°-7.5°
	Vertical Field of View	49.9°~4.3
	Horizontal rotation range	±170°
	Vertical rotation range	-30°~+30°
	Horizontal and vertical flipping	support
	minimum illumination	5lux
	Recommended	Greater than 300 lux

	illuminance	
Audio characteristics	built-in microphone	Equipped with 4 digital array microphones, it has an effective pickup range of 6 meters
	3A treatment	Automatic Echo Cancellation (AEC), Automatic Gain Control (AGC), AI Noise Suppression (ANS)
	Other audio characteristics	Support lip sync and local sound amplification
Stability and security	bandwidth	64Kbps~8Mbps
	Network adaptability	V-Link boasts superb error correction and packet loss resistance, achieving 20% packet loss resistance for video and 30% for audio
	security features	H.235AES media encryption, SRTP media encryption, conference access password, conference control password, HTTPS, blacklist
interface	Audio input interface	Built-in array microphone, 1 x 3.5mm linear input, 1 x miniXLR input, 1 x MIC ARRAY (digital microphone), 1 x HDMI audio input, 1 x USB
	Audio output interface	1 x 3.5mm linear output, 2 x HDMI audio outputs, 1 x USB
	Video input interface	Built-in 4K60fps ultra-high-definition gimbal camera, 1 x HDMI (supporting up to 4K60fps), 1 x USB (supporting up to 1080P30fps), 1 x NET VIDEO
	Video output interface	2 x HDMI (supporting up to 4K60fps)
	network interface	1 x LAN:10M/100M/1000M Base-T; 1 x MIC ARRAY/TOUCH: 10M/100M/1000M Base-T, POE powered; 1 x Wi-Fi: 802.11b/g/n/ax, supports STA and AP modes;
	USB port	3 x USB3.0 (supports USB flash drives, USB microphones, etc.)
	TF card expansion slot	support
	Digital microphone cascading	support
Business characteristics	Dual-screen independent display	
	Wi-Fi wireless access	
	Support banners and subtitles	
	Support recording on third-party platforms for RTMP streaming	
	Support USB local recording	
	Supports optional built-in MCU function (with optional License authorization), and supports up to 8 concurrent accesses of H.265 1080P30	
	Support wired digital microphone three-level cascade access (optional AM50S)	
	Support USB wireless screen projector (optional CP50S)	
	Support conference controller (optional PM50X)	
	Support third-party integration and development of API interfaces	
	Maintenance function	Built-in Web management, supports configuration export and backup
		Support automatic hibernation and wake-up mechanisms for terminals

		Event log recording and exporting
		Supports IP network diagnosis, audio diagnosis, and factory settings recovery
size	258mm(W)×146mm(D)×170mm（H）（without foot pads and interfaces）	
power supply parameters	Working Voltage	DC12V 4.5A
	power	≤54W
runtime environment	temperature	0°C to 40°C (working state), -40°C to 70°C (non-working state)
	relative humidity	10% to 80% (in working state), 0% to 95% (in non-working state)
	altitude	≤5000 meters above sea level

2.1.4 ONE-M1S Ultra HD Video Conference Terminal

Overview

ONE-M1S is an iFreecomm integrated ultra-high-definition video conferencing terminal featuring a four-in-one design, combining a camera, microphone, speaker, and system. It provides users with an ultra-high-definition experience and offers minimalist and intelligent control, making it suitable for rapid deployment in small conference rooms. With standard H.323/SIP communication protocols and wireless BYOM conferencing, it perfectly meets various remote audio and video communication and interactive communication application needs

Product Features

- Integrated design: Integrated fanless design, combining ultra-high-definition camera, conference terminal, microphone, and high-fidelity speaker into one, with convenient wiring and simple installation;
- Ultimate Ultra HD Experience: With support for H.265 4K30fps encoding and decoding, combined with beauty enhancement technology and intelligent front-end and back-end video image processing, it delivers unparalleled audio and video quality, featuring vivid and smooth playback with realistic colors. While saving 50% of bandwidth, it presents you with a stunning and clear world;
- AI automatic participant frame selection: Equipped with built-in AI face recognition and AI human figure recognition algorithms, it can automatically adjust the camera position and focal length according to the number of participants and their position changes, ensuring that each participant is clearly visible in the frame.
- AI automatic tracking: With a built-in sound source localization algorithm, when a participant speaks, the camera will automatically follow the sound and lock onto the speaker, presenting a close-up image in the center of the screen, bringing a "face-to-face" communication experience. If no one speaks within a few seconds, it will automatically return to the panoramic view of the conference venue
- Wireless BYOM Conference: Supporting the wireless BYOM function, after initiating or joining a cloud conference from a personal PC, users can access the conference terminal's camera, microphone, and external display and speakers, and support wireless screen mirroring. The seamless switching between BYOM conference and terminal conference is more convenient;
- Multiple microphone access methods: Supports three audio input solutions: digital array microphone, USB microphone, and analog microphone, adapting to different conference scenarios and meeting more audio needs;
- Built-in MCU: Optional built-in MCU multi-point control unit function, supporting 1+8 networking, meets the simple networking and rapid meeting needs of small and medium-sized enterprises, and improves the communication and collaboration efficiency of small meetings;
- Superior network adaptability: In the event of severe network packet loss, the V-Link proprietary technology can be activated. Packet loss within 20% has no impact on the system, and meetings can still proceed normally with 30% packet loss, effectively addressing issues of network jitter and packet loss;
- Security and confidentiality: Supports H.460 firewall traversal technology to address secure connections between public and private networks. Supports H.235, SRTP protocols, AES-128 encryption algorithms, and China's

self-controlled chips, providing comprehensive protection for communication security;

Technical Specifications

Protocol and standard	Multimedia framework protocol	ITU-T H.323, IETF SIP V2
	Video codec protocol	H.265, H.264 High Profile, H.264, H.263, H.263+
	Audio codec protocol	G.711a/u, G.722, G.722.1, G.722.1C, G.726, G.729, AAC-LD, Opus, SILK
	Dual-stream protocol	ITU-T H.239, BFCP
	Network transmission protocol	TCP/IP, DHCP, SSH, HTTP, HTTPS with SSL/TLS, RTP, RTCP, RFC3261, RFC3264, RFC2190, RFC3407, RFC2833, RFC4585 (RTP/AVPF), SNTP, ARP
	Other agreements and standards	H.221, H.224, H.225, H.235, H.241, H.245, H.281 H.350, H.460, T.140, DTMF
Video features	Mainstream image resolution	4KP30, with a minimum bandwidth of 2048Kpbs 1080p, with a minimum bandwidth of 512Kpbs 720p, minimum bandwidth 384Kbps
	Demo resolution	800x600, 1024x768, 1280x1024, 1280x720, 1920x1080, 3840x2160
	Dual-flow capability	Dual-channel 4K 30fps encoding and decoding
	Other image characteristics	Supports display modes such as PIP and POP, with enhanced image effects and pre- and post-processing of images
built-in camera	sensor	1/2.5 inch, UHD CMOS image sensor, 8.51 million effective pixels
	scanning method	line by line
	Lens focal length	5x electronic zoom, f=4.4mm~52.8mm, F1.8~F2.6
	resolution	Supports up to 4K60fps
	Horizontal Field of View	120°
	Vertical Field of View	110°
	Horizontal rotation range of mechanical gimbal	±15°
	Vertical rotation range of mechanical gimbal	±15°
	signal-to-noise ratio	≥55dB
	Horizontal and vertical flipping	support
	minimum illumination	0.5Lux @ (F1.8, AGC ON)
	Recommended illuminance	Greater than 300 Lux
Audio characteristics	built-in microphone	Built-in digital array microphone (4mic), effective pickup range ≥6 meters, optimal pickup distance ≤3 meters

	3A treatment	Automatic Echo Cancellation (AEC), Automatic Gain Control (AGC), AI Noise Suppression (ANS)
	Other audio characteristics	Support lip sync
	built-in speaker	Power: 8W, frequency: 100Hz-16kHz, maximum sound pressure level (SPL): 96dB@0.5m
Stability and security	bandwidth	64Kbps~8Mbps
	Network adaptability	V-Link boasts exceptional error correction and packet loss resistance, achieving 20% packet loss resistance for video and 30% for audio
	security features	H.235 AES media encryption, SRTP media encryption, conference access password, conference control password, HTTPS, blacklist
interface	Audio input interface	Built-in array microphone, 1 x 3.5mm linear input, 1 x digital microphone input, 1 x HDMI audio input, 1 x USB
	Audio output interface	1 x 3.5mm linear output, 2 x HDMI audio outputs, 1 x USB
	Video input interface	Built-in 4K30fps ultra-high-definition camera, 1 x HDMI (up to 4K60fps), 1 x USB (up to 1080P30fps)
	Video output interface	2 x HDMI (supporting up to 4K60fps)
	network interface	1 x LAN: 10/1000Base-T; 1 x MIC ARRAY/TOUCH: 10/1000Base-T, POE powered; 1 x Wi-Fi : 802.11 b/g/n/ac&BT5.0
	USB port	2 x USB3.0 (supports USB flash drives, USB microphones, etc.); 1 x Type-C (for use in USB peripheral mode)
	Digital microphone cascade	support
Business characteristics	Dual-screen independent display	
	Wi-Fi wireless access	
	Support banners and subtitles	
	Support the selection of participants and close-up shots of speakers	
	Supports optional built-in MCU 1+8 channels (optional License authorization)	
	Support wired digital microphone three-level cascade access (optional)	
	Support USB wireless screen projector (optional)	
	Support for conference controller (optional)	
	Support third-party integration and development of API interfaces	
	Maintenance function	Built-in Web management, supports configuration export and backup
		Support automatic hibernation and wake-up mechanisms for terminals
		Event log recording and exporting
		Supports IP network diagnosis, audio diagnosis, and factory settings recovery

size	601mm (W) × 141mm (D) × 115mm (H) (including stand)	
power supply parameters	Working Voltage	DC12V 3A
	power	≤36W
runtime environment	temperature	0°C ~ 40°C (operating state), -40°C ~ 70°C (non-operating state)
	relative humidity	10% ~ 80% (working state), 0% ~ 95% (non-working state)
	altitude	≤ 2000 meters above sea level

2. 2 Terminal supporting products

2. 2. 1 CP50S One-click Smart Screen Casting Device

Overview

The CP50S wireless screen mirroring device is a 4K wireless video interactive collaboration device specifically designed for the ONE series of video conferencing terminals. It enables ultra-clear screen mirroring and sharing of content from personal computers through wireless Wi-Fi, as well as BYOM (Bring Your Own Meeting) applications, flexibly meeting the needs of various collaboration scenarios.

Product Features

- Plug and Play: Connect the video conferencing device via a USB cable, quickly pair it within 3 seconds, and then connect it to a laptop. Without the need to install drivers or software, you can easily mirror the computer screen to the display with a single click, facilitating smoother and more efficient collaboration;
- Ultra-clear picture quality: Utilizing hardware codec technology, it supports wireless screen mirroring up to 4K resolution, ensuring lossless transmission of images. In BYOM mode, it supports 1080P resolution, presenting conference details smoothly;
- Stable transmission: Equipped with 2T2R dual-antenna technology, it boasts stronger transmission efficiency, ensuring stable transmission within a 15-meter line-of-sight range, with an average latency of less than 150ms, providing lag-free conferencing;
- Universal Interface: USB Type-C & Type-A combo interface, supporting DP Alt mode, compatible with mainstream devices, replaceable male plug to adapt to different interface needs, female socket integrates audio, video, power supply, and other functions, solving all connection problems with one cable;
- Independent Wi-Fi: Equipped with a dedicated Wi-Fi module, it does not require the use of the computer's network, ensuring that screen mirroring and network tasks can proceed simultaneously, thus avoiding any interference from lag or delay in shared content.

Technical Specifications

interface	Type-C*1 (or Type-C to USB A) for power supply, audio and video transmission, pairing, and upgrading
USB Type-C video capture resolution	3840x2160P30, 1920x1080P60, 1920x1080P50, 1920x1080P30, 1366x768P60, 1280x800P60, 1280x720P60, 1280x720P50, 1024x768P60
USB Type-A video capture resolution	1920x1080P30, 1366x768P60, 1280x800P60, 1280x720P60, 1280x720P50, 1024x768P60

Screen video encoding format	H.265, H.264
Screen audio encoding format	44.1KHz/48KHz/16bit PCM, stereo
Return stream resolution and frame rate	Up to 1920x1080, 30 frames per second
Returned code stream video format	MJPEG
Returned code stream audio format	8~48KHz, mono/stereo
Supported operating systems	Windows 7/8.1/10/11 32-bit and 64-bit; Mac OS X 10.10/10.11/10.12/10.13 and above
USB control protocol	UVC/UAC standard protocol
Wireless connection method	Insert the button for automatic connection, automatic operation, and one-click sharing
Wireless transmission protocol	IEEE802.11ac/802.11n
Wireless transmission distance	Up to 15 meters of line-of-sight range (actual distance may be affected by environmental interference)
Wireless transmission frequency band	5G and 2.4G
Wireless encryption protocol	WPA2-PSK
connection time	The time from inserting the button to being able to cast the screen is less than 13 seconds
transmission delay	The average delay is less than 150ms
antenna	2T2R, with built-in on-board antenna
power supply	USB5V/1A
power consumption	Average power consumption: 3.0W (when using USB Type-C for 4K screen mirroring) / 1.8W (when using USB Type-A for 1080P screen mirroring)
size	62mm x 62mm x 16mm (excluding the adapter cable)
temperature range	Operating temperature: +5°C to +40°C; Storage: -20°C to +60°C
humidity range	Storage: 0% to 90% relative humidity; Operation: 0% to 90% relative humidity, no condensation

2. 2. 2 PM50X conference controller

Overview

The PM50X conference controller is designed for the flat touch interface of the ONE series video conferencing terminals. The user interface is simple and intuitive, allowing you to see what you get. It supports touch and drag functions. During use, you can easily control the conference by simply tapping and touching, providing you with an ultimate conference experience.

Product Features

- Android operating system, easy to get started, 10.1-inch large capacitive screen, multi-touch, convenient and smooth operation
- Equipped with a multi-angle adjustable stand, you can clearly see the screen in different sitting positions.
- Supports power supply via USB and PoE interfaces, with plug-and-play functionality
- Simple and user-friendly, it will bring you a refreshing conference experience.

Technical Specifications

hardware	operating system	Android9.0
	processor	Six-core, 4 x Cortex A73 + 2 x Cortex A53
	RAM (Random Access Memory)	4GB
	storage	16GB
display	screen size	10.1-inch
	screen resolution	1920x1200
	brightness	300cd/㎡(typ.)
	viewing angle	170°
	Screen Type	IPS
	Pointing and selection device	10-point multi-touch
function	terminal configuration	
	functions such as mute/unmute, volume adjustment, camera PTZ control, preset position recall, dual-stream sharing, and call/end conference site.	

	Tencent Meeting Rooms Controller APP (optional)	
interface	USB port	1 x USB2.0 Type-C (supports power supply and data transmission)
	network interface	1xRJ45:10/100Base-T(POE) Wi-Fi: IEEE802.11a/b/g/n/ac,WPA,WPA2
physical properties	power supply range	DC5V/PoE(802.3af)
	input current	3.0A
	connection method	Wired connection
	ambient temperature	Temperature: 0°C~40°C (working state), -40°C~70°C (non-working state) Relative humidity: 10% to 80% (working state), 0% to 95% (non-working state)
	Overall dimensions	242mm(W)×162mm(D)×29mm(H)
	Net weight	800g

2. 2. 3 AM50S omnidirectional digital array microphone

Overview

AM50S is a brand-new generation of omnidirectional digital array microphone from Jieshi Feitong, equipped with an audio processing unit and AI audio algorithm capabilities. It can effectively eliminate various noises, murmurs, echoes, howls, etc. in the conference room, bringing a high-quality voice call experience. The microphone can be flexibly cascaded and expanded according to the size of the space, ensuring that the sound pickup range evenly covers the entire space, meeting the needs of different venues and bringing users a brand-new high-fidelity stereo sound quality experience.

Product Features

- Comprehensive sound quality experience: Equipped with a ring array consisting of six high signal-to-noise ratio microphones, it boasts a 360-degree sound pickup range, enabling comprehensive high-fidelity audio communication and providing you with a zero-distance video communication experience;
- AI audio algorithm: Equipped with a blind beamforming algorithm, it automatically locates the sound source and adjusts the direction of sound pickup to ensure balanced sound pickup; equipped with an audio 3A processing algorithm, it can effectively suppress environmental noise, static, and speech reverberation, effectively enhancing sound quality;
- POE Cascading: Intelligent hand-in-hand conference audio acquisition method, with an effective pickup distance of 6 meters and a clear pickup distance of 3 meters for a single unit. It supports up to 6 microphones in PoE cascading, providing distributed pickup and uniform coverage for medium to large conference rooms;
- Built-in audio processing unit: Equipped with a powerful built-in audio processing unit, eliminating the need for external audio processors and reducing equipment wiring; featuring ultra-low signal transmission delay, ensuring seamless remote calls without any latency;
- Multiple interface outputs: Equipped with standard USB2.0 and Aux audio interfaces, the device is plug-and-play, meeting dual-mode applications for both digital and analog audio.
- Impressive high-definition audio: With a sampling rate of 32KHz, full-frequency voice, and support for dual-channel stereo sound, it provides users with a stunning high-fidelity stereo sound quality, presenting clear audio details and allowing you to feel like you're having a conversation at zero distance.

Technical Specifications

Featur	microphone array	Built-in six omnidirectional microphones form a circular array

es	signal-to-noise ratio	65dB(A)
	frequency response	50Hz-16kHz
	sensitivity	-38dBFS
	sampling rate	32K sampling, high-definition wideband audio
	cascade quantity	Up to 6 units (POE), supported only in POE scenarios
	Audio algorithm	Echo cancellation, noise suppression, and gain control algorithms
	Control method	Mute button
	Sound pickup method	omnidirectional
	Sound pickup range	360° omnidirectional sound pickup
	Pickup distance	Effective pickup range of 6 meters, with optimal pickup at 3 meters
interface	Network port interface (compatible with ONE series terminals)	1-way Up: upstream networking port
		1 Down: Downlink networking port
	USB port	1-way USB: USB audio interface (Type-C), UAC1.0 protocol for audio data communication, software upgrade and parameter configuration
	audio interface	Channel 1 Aux1: 3.5mm audio input and output interface
		1-channel Aux2: 3.5mm audio input and output interface
physical properties	power supply method	PoE cascade 48V power supply supports IEEE802.3at standard/USB 5V 500mA (single unit)
	connection method	Wired connection
	ambient temperature	0℃~45℃
	bare machine size	Φ170mm×H40mm
	weight	370g

2. 2. 4 AM30WS Wireless Omnidirectional Digital Microphone

Overview

AM30WS is a brand-new wireless omnidirectional digital microphone from Jieshi Feitong. It adopts advanced audio pickup technology, boasts superior anti-interference capabilities and excellent sound pickup performance, with a 360-degree sound pickup range of up to 6 meters. It features lossless broadband audio transmission, meeting the needs of various conference scenarios such as small conference rooms and personal offices.

Product Features

- Multiple connection methods: The host is equipped with a 5.8G wireless adapter, ensuring stable transmission and no interference. It supports both wireless connection and Type-C wired connection, offering plug-and-play functionality for easy connection.
- Array Microphone: Built-in high signal-to-noise ratio microphones form a circular array, providing a 360-degree sound pickup range, enabling comprehensive high-fidelity audio communication, and offering you a zero-distance video communication experience.
- Large-capacity battery: Equipped with a 5000 mAh rechargeable lithium battery pack, it offers a continuous usage time of 10 hours and an extended standby time of 120 days.
- Built-in high-performance DSP chip: 3A audio algorithm, automatic echo cancellation, intelligent noise suppression, automatic gain control, reverberation reduction, and full-duplex communication
- Human-machine interaction: Equipped with 5 buttons, it provides various interactive settings for the machine; and features 5 colored LED indicators, which display various states of the machine.
- Voice prompt: Provides voice prompts for various states of the machine. The prompt language can be selected between Chinese, English, and ringtone through button presses, and the prompt sound can be turned on or off by pressing a combination of buttons.

Technical Specifications

name	parameter
Built-in silicon microphone	6-channel microphone array
sensitivity	Built-in microphone: -36dBFS
signal-to-noise	65dB

ratio	
frequency response range	100Hz - 16kHz
transmit power	5.8G:+7dBm
Echo cancellation	Industry-leading echo cancellation technology, capable of achieving an echo cancellation duration of up to 256ms
Noise cancellation	>20dB
AI Noise Reduction	Reduce steady-state noise in the absence of human voice
sampling rate	16K
Sound pickup range	360° omnidirectional sound pickup
Pickup radius	Effective pickup range is 6 meters, with the best pickup range being 3 meters
transmission method	Wired Type-C / Wireless 5.8G
Speaker power	5.2W
Control method	Button control
Button Function	Power on/off (check battery level), volume up, volume down, speaker mute, microphone mute, multiple configurations
status display	Color LED indicator light
interface	Type-c
Built-in battery	5000mAH rechargeable lithium battery pack
Usage time	10 hours
Full of time	approximately 5 hours
standby time	No less than 6 months
ambient temperature	-40℃～50℃ (Storage temperature) -10℃～40℃ (Operating temperature)
ambient humidity	0～95% RH
Length × Width × Height	230mm×220mm×48mm
weight	800g

2. 3 Camera products

2. 3. 1 VENUS31S Ultra HD Camera

Overview

VENUS31S is a newly launched 12x optical zoom 4K camera from Jieshi Feitong. Its sensor measures 1/2.8 inches and boasts a horizontal viewing angle of 72.6°. It offers comprehensive functionality, superior performance, and a wealth of interfaces. Advanced ISP processing technology and algorithms ensure vivid and lifelike image effects, uniform picture brightness, strong light and color gradations, high clarity, and excellent color reproduction. This device is an ultra-high-definition camera suitable for various fields such as conferencing, education, and healthcare.

Product Features

- 4K Ultra HD: Equipped with a brand-new generation of high-quality UHD CMOS sensor with a 1/2.8-inch sensor size and a maximum of 8.42 million pixels, it delivers superior images with 4K ultra-high resolution. With an optical zoom of up to 12x, it presents clear and realistic ultra-high-definition videos, vividly capturing expressions and movements of characters, and providing images of superb clarity and resolution quality;
- Rich and comprehensive interfaces: Supports HDMI, USB3.0, network, etc., capable of simultaneously outputting 4K video;
- Autofocus technology: Advanced autofocus algorithms enable the lens to achieve fast, accurate, and stable autofocus;
- Low noise and high signal-to-noise ratio: The low-noise CMOS effectively ensures an ultra-high signal-to-noise ratio for the image. Advanced 2D and 3D noise reduction technologies are employed to further minimize noise while maintaining image clarity;
- Intelligent tracking: Equipped with AI algorithms, it achieves automatic human-form following.
- Multiple preset positions: Supports up to 255 preset positions (10 for remote control settings), including presets for pan, tilt, and zoom. Even if the camera is powered off, the preset data will be saved;
- Multiple network protocols: Supports various network media protocols such as RTSP, RTMP/RTMPS, Onvif, Multicast, GB/T28181, etc;

Technical Specifications

Main parameters	
sensor	1/2.8 inch, UHCMOS, effective pixels: 8.42 million
Optical magnification	12x wide-angle
Lens focal length	f=4.14～47.08mm
lens aperture	F1.8~F2.8
horizontal viewing angle	7.4°~72.6°(±2%)
Vertical perspective	4.16°~40.83°(±2%)
diagonal	8.4°~80.1°(±2%)
shutter	1/25 to 1/10,000 seconds
white balance	Automatic, Shutter Priority
backlight compensation	support
Digital noise reduction	2D/3D digital noise reduction
signal-to-noise ratio	≥50dB
Image style	Standard, clear, beautiful, bright
Image flipping	Horizontal mirror, vertical mirror
AI Intelligence	Support human tracking
Rotation range	Horizontal: -165° to +165° Vertical: -30° to +90°
rotational speed	Horizontal: 1.7°～ 86.7°/S Vertical: 1.7°～ 52.2°/S
PoE power supply	support
TOF	Support (optional)
Pre-set digits	255
preset position accuracy	0.1°
Professional video output	HDMI: 4K P30, 4K P25, 1080P60, 1080P50, 1080P30, 1080P25, 720P60, 720P50
Input/output interface	
HDMI output	1 HDMI port, version: 1.4B
network interface	1-way RJ45: 10M/100M/adaptive Ethernet, supports PoE (802.3af)
audio interface	1-channel Line In, 3.5mm audio interface
USB port	1 USB3.0 port, Type C socket
communication interface	1-channel RS232 In: 8-pin mini DIN, maximum distance of 30 meters, VISCA/PELCO-D protocol; 1-channel RS232 Out: 8-pin mini DIN, maximum distance of 30 meters, VISCA/PELCO-D protocol; 1-channel RS422: 4-pin Phoenix connector, maximum distance of 1200 meters, VISCA/PELCO-D protocol;
power interface	JEITA type (DC IN 12V)
network characteristics	

Video encoding standard	H.265/H.264
Video stream	First stream, second stream
Route 1	3840x2160, 1920x1080, 1280x720, etc
Route Two	704x576, 640x480, 640x360, 576x480, 352x288, 320x240, 320x180, etc
video bitrate	First channel: 100bps-40960kbps, Second channel: 100bps-4096kbps
Bit rate control	Variable bit rate, fixed bit rate
frame rate	5fps-30fps
Audio compression standard	AAC, G.726, G.711a, G.711u
Audio bit rate	32kbps, 40kbps, 48kbps, 64kbps, 96kbps, 128kbps
Supported Protocols	TCP/IP, HTTP, RTSP, RTMP/RTMPS, Onvif, DHCP, Multicast, GB/T28181, NDI®|HX2 (optional, requires separate authorization), etc
USB characteristics	
support system	Windows 7, Windows 8, Windows 10, Windows 11, macOS, Linux versions 2.4.6 and above, Kylin, UWC, and Android versions that require UVC-related drivers
color space	YUY2/ MJPEG
video format	MJPEG: 3840x2160, 2560x1440, 1920x1080, 1600x896, 1280x720, 1024x768, 1024x576, 720x576, 960x540, 848x480, 800x600, 800x448, 640x480, 640x360, 480x272, 325x288, 320x240, etc YUY2: 720x576, 848x480, 800x600, 800x448, 640x480, 640x360, 480x272, 325x288, 320x240, etc MJPEG frame rate: 30, 25, 20, 15, 10, 5 frames at 3840x2160 and 2560x1440, and 60, 50, 30, 25, 20, 15, 10, 5 frames at other resolutions YUY2 frame rates: 25, 20, 15, 10, 5 frames
USB video	UVC1.0, UVC1.1, UVC1.5
UVC PTZ control	support
General specifications	
input voltage	DC12V/PoE(802.3af)
input current	800mA (maximum)
power consumption	<10W
Storage temperature	-40~60℃
Operating Temperature	0~50℃
Operating Humidity	0～95% RH
size	224mm x 132mm x 162mm
body weight	1500g
body color	iron gray

2. 4 Audio Products

2. 4. 1 JA-AM129S non-inductive sound reinforcement ceiling array microphone

Overview

JA-AM129S is a non-sensory sound reinforcement ceiling array microphone launched by IFREECOMM. Equipped with a built-in 129-element omnidirectional microphone array, it features the latest audio AI algorithms, top-tier core hardware support, cutting-edge AI intelligent noise reduction technology, and innovative single-unit minimalist sound reinforcement design, redefining the standards for intelligent audio equipment in meetings. It also boasts functions such as AI noise reduction, AI reverberation suppression, directional sound pickup, high-definition sound quality, and multi-unit cascading, meeting the core needs of remote meetings and local non-sensory sound reinforcement in various high-end large conference rooms, lecture halls, multi-function halls, interactive meeting spaces, and smart classrooms.

Product Features

- 129 Microphone Array: A 129-unit ceiling-mounted omnidirectional microphone array, capable of capturing sound up to a distance of 20 meters, allowing you to speak freely wherever you go, achieving true seamless sound reinforcement;
- Unique AI audio algorithm: By utilizing a vast amount of labeled training data, it learns the mapping relationship between noise and clean signals. The algorithm then outputs a four-dimensional noise reduction engine, which precisely identifies sounds, removes various noises, enhances speech signals, and provides a pure audio experience;
- AI dynamic sound field optimization: Real-time analysis of spatial acoustic characteristics, 129-channel beamforming, precise sound capture, and improved sound quality;
- High-volume high-fidelity: Acoustic feedback gain ≥15dB, providing louder volume without the risk of feedback howling, ensuring effortless speech delivery;
- Multiple independent audio zones: Supports customizable setup for directional sound pickup, supports 37 sound pickup zones and 17 sound amplification area modes, and supports cascading multiple units to adapt to different scenarios;
- Multiple interfaces: A wide range of analog and digital (Dante, USB) audio interfaces are available, which can be used independently and can be connected to wireless microphones, etc., without the need for additional audio processors;

- High performance and low latency: 48khz high sampling, with latency as low as 18ms, and cascaded microsecond-level latency, ensuring more fidelity and clarity in sound;
- Flexible control: Provides multiple external control methods such as infrared, RS485, and network, meeting different needs and making the system more flexible;

Technical Specifications

module	function	parameter
Algorithm specification	Audio algorithm	AFC, ANS, AEC, AGC, ARR
	Acoustic feedback gain	≥18dB
	Noise reduction amplitude	≥30dB
	Echo cancellation amplitude	≥90dB
	Echo cancellation length	≥1s
	Reverberation suppression	≥18dB
	maximum gain	≥30dB
	AI Noise Reduction	support
	auto-mixing	support
	beamforming	support
	sound source localization	support
Microphone specifications	number of microphones	129
	sensitivity	-32±2dB
	signal-to-noise ratio	70dB
	frequency response	75-20KHz
Hardware Specifications	sampling rate	48K
	frequency response	20Hz~20KHz,±0.5dB
	signal-to-noise ratio	100dB
	distortion	≦0.1%
	Audio input interface	4-way
	Audio output interface	4-way
	Input impedance (balanced)	20KΩ
	Output impedance (balanced)	200Ω
	Maximum input level (balanced)	4dBu
	Maximum output level (balanced)	10dBu
	USB	Type-C firmware upgrade interface, USB2.0 audio transmission
	Ethernet port	3 network ports: 1 LAN port, supporting Dante protocol and PoE power supply; 2 cascade network ports.
	Dante audio	Supports 2 input channels and 2 output channels
	cascade function	Supports cascading of 20 units
	RS485 central control interface	support
	input power	DC 12V~48V/POE power supply
	power	25W
	Installation method	Embedded into the ceiling (must be fixed with rope hoisting), rope hoisting, and boom installation
Other Specifications	Size (length x width x height)	600×600×42.8mm
	Net weight of equipment	6.5kg
	Operating Temperature	0℃~40℃
	Storage temperature	-20℃~60℃
	color	pearl white

2. 4. 2 JS-ST5E conference fusion host

Overview

JS-ST5E is a multifunctional conference integration host launched by IFREECOMM, integrating various device functions such as wired & wireless conference discussion systems, UHF receivers, audio processors, power amplifiers, and more. The device is designed to be lightweight, and with simple connection, it can achieve complete conference functions. It is plug-and-play, eliminating the need for complex debugging processes. It is widely applicable to various scenarios such as conference rooms, multimedia classrooms, corporate training centers, broadcasting systems, and more. It supports temporary meetings and outdoor activities, allowing for quick setup of conference systems.

Product Features

- The all-aluminum alloy one-piece chassis is equipped with a 2.0-inch LCD display screen, featuring both Chinese and English display interfaces. The user-friendly human-machine interface enhances debugging efficiency;
- Equipped with high-performance industrial-grade FPGA and pure Chinese embedded processor, it can implement custom digital logic and operate at a faster speed;
- Support the mixed use of wired conference units, wireless conference units, and handheld microphones, allowing for flexible combinations to meet the needs of various conference scenarios;
- Operating within the frequency band of 630-690MHz, it features 5 sets of UHF channels to choose from, supports the simultaneous use of 5 systems, and easily avoids various types of interference, preventing the occurrence of cross-frequency interference;
- It features seven conference modes: first-in first-out, last-in first-out, request to speak, queue to speak, free mode, chairman mode, and voice control mode;
- The number of wired speakers can be set between 1 and 20, with no limit on the chairman. The number of wireless speakers can be set between 1 and 4. The wired unit's free discussion mode is not limited and can be fully enabled;
- Utilizing high-fidelity lossless audio transmission technology, with a 48KHz audio sampling frequency and a frequency response ranging from 20Hz to 20KHz;
- Equipped with a built-in DSP digital audio module, it features adaptive feedback suppression, digital equalizer, and automatic gain control, effectively minimizing interference, distortion, and crosstalk to the greatest extent possible;
- Equipped with a built-in perpetual calendar and a speaking timing function, the display screen can dynamically show the date, day of the week, and time. The system can automatically synchronize the date, day of the week, and time with the computer;
- Using ID addressing, customizable unit numbers can be assigned to ensure unique ID numbers, effectively avoiding ID duplication and conflicts;
- Built-in camera tracking function, supporting protocols such as PELCO-D, PELCO-P, and SONY, with the

ability to adjust, save, and recall preset positions;

- Built-in audio processor and digital power amplifier;
- Supports bidirectional USB sound card, USB audio input and output for PC, as well as U-disk recording function;
- Support web page control function, allowing users to set and adjust various conference functions through browser access;
- It features a commissioning-free function, supports preset scene selection, and allows quick selection and setting of all functions;
- Equipped with Wi-Fi/Bluetooth/UHF antenna interfaces, it supports network function control and wireless audio transmission;
- Supporting hot plugging and automatic repair functions for lines, enhancing the reliability and stability of the system;
- It features a factory reset function, allowing for one-click reset to the factory default state to prevent parameter tampering;

Technical Specifications

Technical Specifications	audio processing	Noise floor (A-weighted): -79dBu
		Input impedance (balanced type): 20KΩ
		Output impedance (balanced): 100Ω
		System latency: <3ms
		Dynamic range: 92dB
		Total harmonic distortion: < 0.05%
		Signal-to-noise ratio: 90dBA
		Maximum input level: +18dBu, balanced
		Frequency response: 20Hz~20KHz
		Maximum output level: +18dBu, balanced
		Equivalent Input Noise (EIN) (20-20kHz, A-weighted): ≤-110dBU
		Phantom power (per IO card input): 48V
		Channel isolation, 1kHz: 100dB
	Wireless performance	Wireless frequency range: UHF: 630-690MHz, RF: 433Mhz, Wi-Fi
		Wireless coverage: RF effective communication distance is ≥50 meters (in an unobstructed environment), and WIFI effective communication distance is ≥30 meters (in an unobstructed environment) The effective communication distance of BT is: ≥10 meters (in an unobstructed environment)
		Wireless security: AES (Advanced Encryption Standard), Frequency Hopping Spread Spectrum (FHSS) technology
	Meeting management and control	Control interface: 2.0-inch display screen, supporting both Chinese and English interfaces
		Maximum simultaneous speaking capacity: Supports simultaneous speaking by 4 (wireless) / 20 (wired) units
		First-in-first-out, last-in-first-out, request to speak, queue to speak, free

		mode, chairman mode, voice control mode
		Meeting minutes: Support high-definition audio recording
		Meeting management and control: mobile APP and computer software
	Wireless performance	Wireless frequency range: UHF: 630-690MHz, RF: 433Mhz, supporting up to 256 wireless units; Wi-Fi: 2.400GHz~2.4835GHz, BT: 2.400GHz to 2.4835GHz
interface	Conference unit interface	Equipped with 2 wired conference unit connection interfaces, each supporting up to 28 units
	Audio input interface	2-channel 3-pin Phoenix terminal balanced input, supports 48V phantom power supply
	Audio output interface	2-channel 3-pin Phoenix terminal balanced output
	Power amplifier interface	4-channel power amplifier output with Phoenix terminal sockets (8Ω/100w x4ch), supporting independent volume adjustment knobs
	USB port	1-channel USB sound card interface: Supports bidirectional USB sound card and USB audio input and output for PC; 1-channel USB recording interface: Insert a USB flash drive to record audio from the conference discussion system;
	network	1 x LAN:10/1000Base-T 1 x Wi-Fi:802.11 b/g/n&BT4.0
other	Video Control	1-channel RS-485 Phoenix terminal 1-channel RS-232 Phoenix terminal
	power supply	110～240V 50/60Hz
	power consumption	< 350W

2. 4. 3 JS-ST5E matching microphone

2. 4. 3. 1 JS-ST20 wired conference unit

model	name	Product Category
JS-ST20C	Wired chairman unit	Hand-held microphone
JS-ST20D	Wired representative unit	Handheld microphone

Product Features

- Adopting a wired hand-in-hand design and high-precision phase-locked loop (PLL) frequency synthesis technology, the transmission is more stable;
- Equipped with a 3.5-inch LCD high-brightness touch screen, it supports display in three languages and can show system signals, channels, ID addresses, and speaking times;
- Equipped with a gravity-sensing and touch-sensing chip, it features an elliptical orbit three-color indicator light display. There is no mechanical button sound when switching on or off, ensuring a long service life;
- The aluminum alloy panel is embedded with a touch glass panel with a Mohs hardness of 6, featuring waterproof and dustproof functions to prevent tea and water splashes during meetings from affecting the normal use of the equipment;
- Utilizing nanoscale fingerprint isolation technology, the oleophobic and waterproof layer effectively isolates water, oil, and stains, thus achieving anti-fingerprint and anti-sweat effects;
- The system combines digital DSP control circuit technology with analog audio circuit technology to perform activities detection, noise suppression, and other sound processing tasks;
- It adopts a zinc-aluminum alloy microphone boom, with built-in 2×14mm high-fidelity gold-plated condenser cartridges, supporting dual-cartridge transverse array technology;
- The unique cavity and integrated sound-picking steel mesh design allow for precise control of sound pickup angles, preventing acoustic feedback and howling;
- The microphone boom features 32,500 high-density sound interference holes with a diameter of 0.4mm. The hole diameter and spacing are precisely calculated, and combined with interference correction technology, the sound pickup details are precisely controlled, effectively enhancing the pickup sensitivity in the vocal frequency range;
- Equipped with professional aluminum foil and waterproof shielded 1.5-meter CAT5 cable, it can effectively avoid and prevent electromagnetic interference in the circuit;
- High-sensitivity design, with built-in automatic gain function, enabling an effective pickup distance of up to 60-100cm;
- High-fidelity lossless audio transmission technology, featuring a 48KHz audio sampling frequency, delivers sensitive sound pickup and clear voice, with a frequency response spanning from 30Hz to 20KHz;
- Utilizing forward error correction and retransmission mechanisms, it boasts exceptional anti-interference capabilities and effectively isolates signal interference from mobile phones, radio waves, Bluetooth, Wi-Fi, and other sources, ensuring no noise is generated during incoming calls;

- Support the first-in first-out (FIFO) conference mode, which allows for the simultaneous activation of 4 units, with the number of speakers configurable to 1/2/3/4 stations;
- The system allows for customization of conference unit ID numbers, effectively avoiding ID address conflicts and ensuring the uniqueness of system ID addresses;
- Equipped with one TYPE-C interface, it can be used for charging or upgrading programs and maintenance without dismantling the device;
- Equipped with one 3.5mm headphone jack, it can be used for conference monitoring or recording by participants;
- Equipped with a set of volume buttons, allowing for adjustment of the monitoring volume level;

Technical Specifications

connection method	RJ45 wired connection	working power supply	DC24V host power supply
display screen	3.5-inch LCD touch display	Mic core method	2×14mm capacitive array high-fidelity gold-plated microphone cartridge
pointing feature	Heart-shaped directional electret	output impedance	<200Ω
frequency response	20Hz-20KHz	Maximum withstandable sound pressure	136dB(1%T.H.D.1kHz,0dB SPL=2x10Pa)
signal-to-noise ratio	>92dB	equivalent noise	18dB, A-weighted
dynamic range	98dB	Camera tracking	possess
Total Harmonic Distortion	<0.05%	connection method	T-shaped head
Microphone sensitivity	-35dB±2dB	Installation method	desktop-style
(Length LxWidth WxHeight H of the meter pole)	225×34×23mm (excluding the rotating shaft)	Base size (LxWxH)	100×130×55mm

2. 4. 3. 2 JS-ST20 Wireless Conference Unit

model	name	Product Category
JS-ST20CW	Wireless chairman unit	Wireless handheld tabletop microphone, USB charging
JS-ST20DW	Wireless representative unit	Wireless handheld tabletop microphone, USB charging

Product Features

- Adopting Wi-Fi multi-channel + UHF four-channel design, high-precision phase-locked loop (PLL) frequency synthesis technology ensures more stable transmission;
- Equipped with a 3.5-inch LCD high-brightness touch screen, it supports display in three languages and can show system signals, channels, ID addresses, and speaking times;
- Equipped with a gravity-sensing touch chip, it features an elliptical orbit three-color indicator light display, silent mechanical button presses during switching, and a long service life;
- The aluminum alloy panel is embedded with a touch glass panel with a Mohs hardness of 6, featuring waterproof and dustproof functions to prevent tea and water splashes during meetings from affecting the normal use of the equipment;
- Utilizing nanoscale fingerprint isolation technology, the oleophobic and hydrophobic layer effectively isolates water, oil, and stains, thus achieving anti-fingerprint and anti-sweat effects;
- Operating within the frequency band of 610-690MHz, it features 20 sets of Wi-Fi channels to choose from, supports the simultaneous use of 20 systems, and easily avoids various types of interference;
- The system combines digital DSP control circuit technology with analog audio circuit technology to perform activities detection, noise suppression, and other sound processing tasks;
- It adopts a zinc-aluminum alloy microphone boom, with built-in 2×14mm high-fidelity gold-plated condenser cartridges, supporting dual-cartridge horizontal array technology;
- The unique cavity and integrated sound-picking steel mesh design allow for precise control of sound-picking angles, preventing acoustic feedback and howling;
- The microphone boom features 32,500 high-density sound interference holes with a diameter of 0.4mm. The hole diameter and spacing are precisely calculated, and combined with interference correction technology, the sound pickup details are precisely controlled, effectively improving the pickup sensitivity of the vocal frequency range;
- Equipped with professional aluminum foil and water-blocking 1.5-meter CAT5 cable, it can effectively avoid and prevent electromagnetic interference in the circuit;
- High sensitivity design, built-in automatic gain function, effective pickup distance up to 60-100cm;
- High-fidelity lossless audio transmission technology, with a 48KHz audio sampling frequency, offers sensitive sound pickup and clear voice transmission, with a frequency response ranging from 30Hz to 20KHz;
- Utilizing forward error correction and retransmission mechanisms, it boasts exceptional anti-interference capabilities, effectively isolating signal interference from mobile phones, radio waves, Bluetooth, Wi-Fi, and more, ensuring no noise is generated during incoming calls;

- Support the first-in-first-out (FIFO) conference mode, which allows the simultaneous activation of 4 units, with the number of speakers settable to 1/2/3/4 stations;
- The system allows for customization of conference unit ID numbers, effectively avoiding ID address conflicts and ensuring the uniqueness of system ID addresses;
- Equipped with a high-capacity 3000mA/H lithium battery, it features a dual charging capability, supporting both USB charging and charging via a charging case;
- Equipped with one TYPE-C interface, it can be used for charging or upgrading and maintaining the device without disassembly;
- Equipped with one 3.5mm headphone jack for participant monitoring or recording during meetings;
- Equipped with a set of volume buttons, allowing for adjustment of the monitoring volume level;

Technical Specifications

operating frequency	UHF: 630-689.875 MHz, RF: 433 MHz	working power supply	3.7V DC
communication distance	30M±5M (with a 14dB antenna)	Unit power consumption	<1.5W
display screen	3.5-inch LCD display	Standby operating current	85mA±3mA
Microphone type	Heart-shaped directional electret	Operating current for speech	108mA±3mA
frequency response	30Hz~20KHz	equivalent noise	20dBA(SPL)
signal-to-noise ratio	>92dB	Camera tracking	possess
dynamic range	98dB	Maximum Sound Pressure Level	136dB(THD<3%)
Total Harmonic Distortion	<0.05%	Installation method	desktop-style
Microphone sensitivity	-35dB±2dB	Base size (LxWxH)	100×130×55mm
Lithium battery capacity	3000mA	(Length Lx, Width Wx, Height H)	225×34×23mm (excluding the shaft)

2. 4. 3. 3 JS-SU20DS Wireless Handheld Microphone

Product Features

- Adopting a UHF single-channel multi-frequency design and utilizing high-precision phase-locked loop (PLL) frequency synthesis technology, it ensures more stable transmission;
- Using a frequency range of 600-700MHz, with a channel spacing of 250KHz, it provides 100 channels for selection, easily avoiding various types of interference;
- Equipped with a 1.0-inch high-brightness LCD display, it can dynamically display information such as unit battery level, volume, channel, etc;
- Equipped with high-quality mesh head and microphone head modules, and featuring an all-metal body, it can eliminate other external noises and achieve high sound quality reproduction;
- It features independent sensitivity adjustment, with a range of 0-40, allowing for the adjustment of sensitivity for each microphone according to actual needs;
- Adopting a low-power design, it can operate with 2 AA batteries or nickel-hydrogen rechargeable batteries, with the latter providing a continuous usage of over 10 hours;

3 Case

High-definition video conferencing

Remote court hearing system of the Circuit

China Securities Regulatory

High-definition video conferencing

Video conference system of the National

Provincial Video Conference Reception

Remote Video Letter and Visit Reception

Video conference system of the

10+ national-level 100+ provincial 100,000+ multimedia devices online 50+win-win

National-level customers	Provincial customers	Provincial customers	Win-win cooperation unit
Ministry of Emergency Management of China	Traffic Management Bureau of Guizhou Provincial Public Security Department, China	Henan Provincial Public Security Department, China	Huayu Group
Supreme People's Court of China	Bureau of Letters and Visits, Guizhou Province, China	Forestry Department of Hunan Province, China	Unilumin Digital
State Post Bureau of China	Department of Justice, Guizhou Province, China	Publicity Department of the Zhejiang Provincial Committee of the Communist Party of China	Digital China
China's National Meteorological Administration	Market Supervision and Administration Bureau of Hainan Province, China	Market Supervision and Administration Bureau of Hubei Province, China	China Mobile Group
China Securities Regulatory Commission	Discipline Inspection Commission of Anhui Province, China	Hubei Provincial Food and Drug Administration, China	Sichuan Mobile
China Marrow Donor Program	Administration of Radio and Television of Qinghai Province, China	Natural Resources Department of Sichuan Province, China	Xinjiang Telecom
China's State Administration for Market Regulation and Ministry of Water Resources	Civil Air Defense Office of Fujian Province, China	Department of Housing and Urban-Rural Development, Sichuan Province, China	Liaoning Unicom
China Yangtze River Water Resources Commission	Emergency Management Department of Shaanxi Province, China	The Confidential Affairs Bureau of the Party Committee of Xinjiang Uygur Autonomous Region, China	Shandong Unicom
	Emergency Management Department of Heilongjiang Province, China	Department of Civil Affairs, Xinjiang Uygur Autonomous Region, China	Beijing Unicom
	Guangxi Public Security Department, China	Emergency Management Department of Shandong Province, China	Henan Wan'an
			Jingyeda
			Neusoft Hanfeng Medical Technology

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