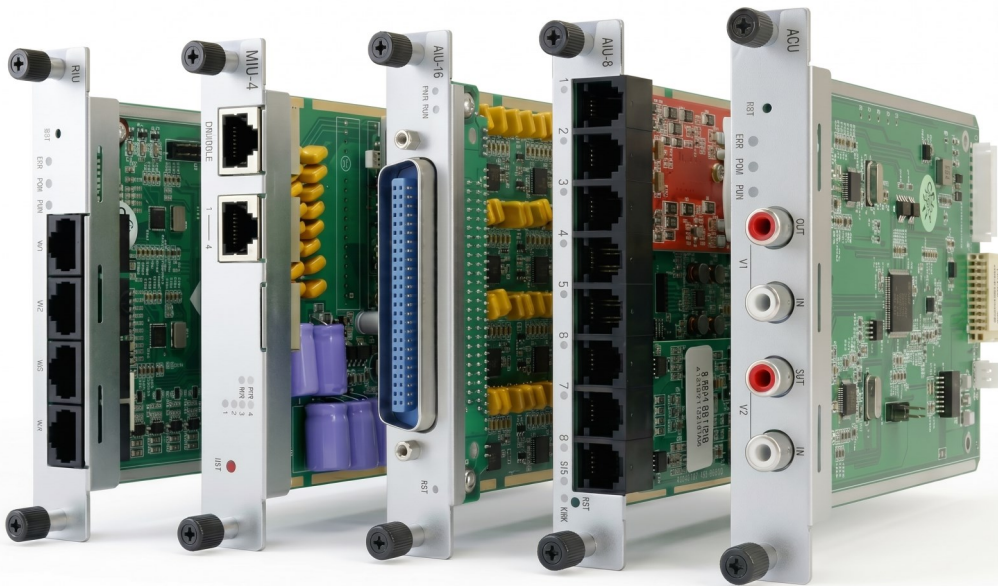




UCP Service Board Datasheet



Version: v1.0



Release History

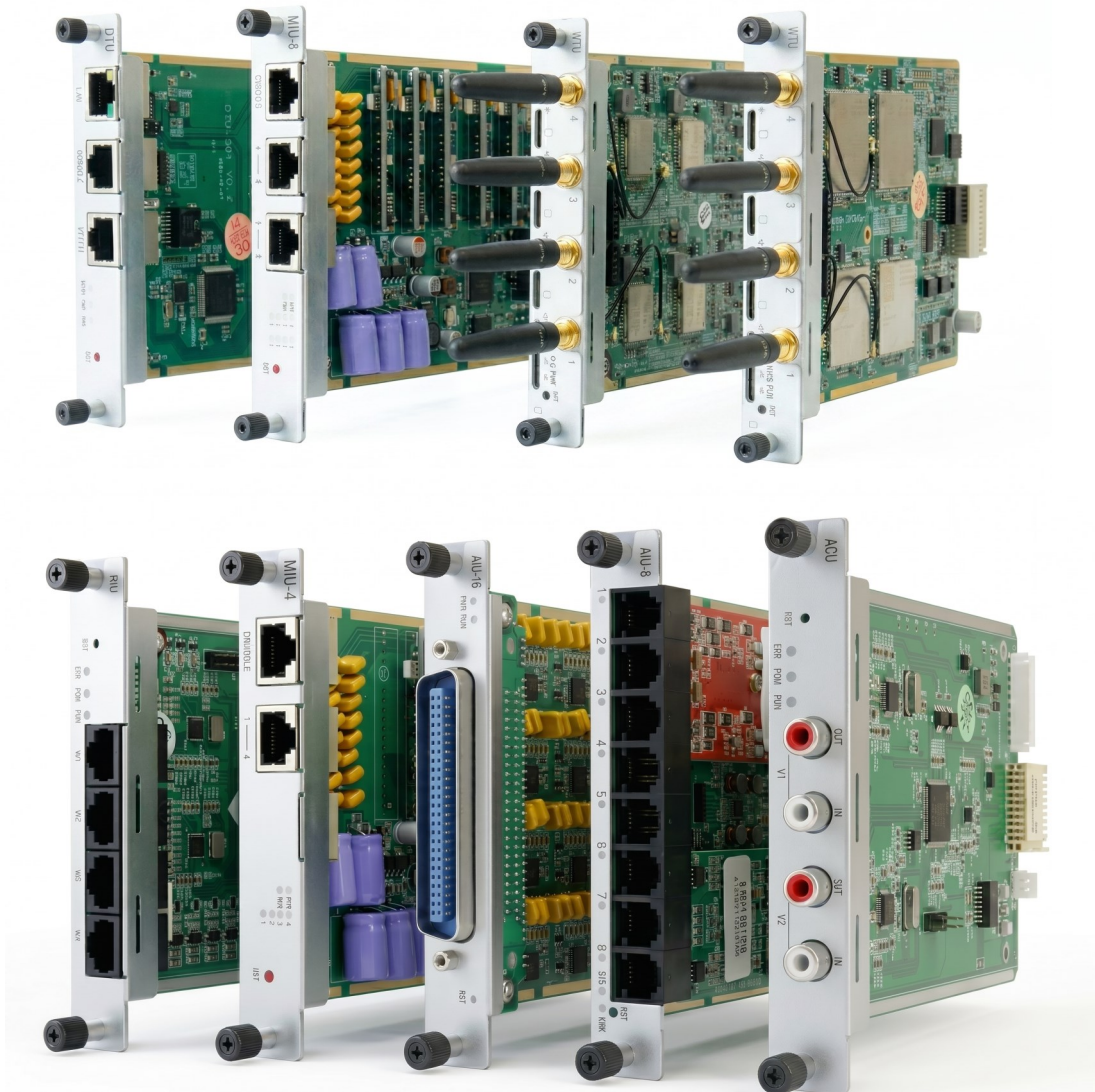
Version number	Release date	Description
1.0	2026.5.13	Released in English for the first time



UCP Service Boards

Overview

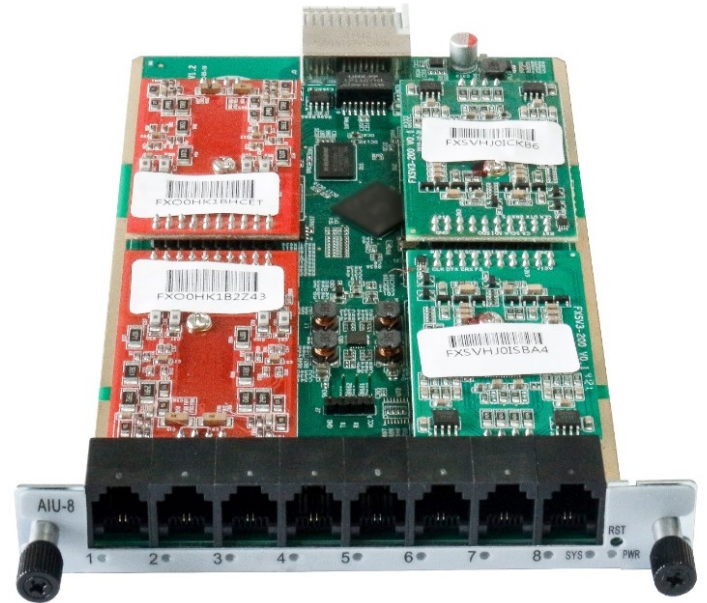
The VoIP voice module is one of the core components of the UCP unified communications platform, responsible for voice access, processing, and transmission. It integrates a variety of voice and interface units, including AIU-8, AIU-16, RIU, ACU, DTU, WTU, MIU, SEU, and RSU. These units provide analog and digital voice access, audio access, wireless voice connectivity, and data security, jointly forming a flexible, efficient, and scalable voice communication architecture that delivers stable and reliable voice service support for the UCP platform.



AIU-8 Analog Gateway Module

The AIU-8 series analog gateway modules, including AIU-80, AIU-8S, and AIU-404S, provide VoIP and PSTN connectivity. Each module offers 8 FXO ports, 8 FXS ports, or 4 FXO plus 4 FXS ports.

The AIU-8 series analog gateway modules deliver high-quality HD voice services and support codecs including G.711A/U, G.723.1, G.729A, G.722, iLBC, OPUS, AMR, and AMR-WB. The AIU-8 series is fully compatible with VoIP platforms such as Asterisk, 3CX, FreePBX, FreeSWITCH, and VOS, helping users reduce telecom and communication costs.



Features

Call Features

- Call waiting
- Blind transfer
- Call transfer
- Call forward on busy
- No-answer call forward
- Unconditional call forward
- Hotline
- Message waiting indicator
- Supports SIP registration to multiple servers
- Call hold
- Do not disturb
- 3-way conference

Voice and Fax

- G.711A/U, G.723.1, G.729A, G.722, iLBC, OPUS, AMR, and AMR-WB
- Comfort noise generation (CNG)
- Supports Modem / POS
- Adaptive dynamic jitter buffer
- Programmable gain control
- DTMF types: SIP INFO / RFC4733 / INBAND
- Supports T.38 and pass-through fax
- Echo cancellation (G.168), up to 128 ms

Software Features

- Ring group function
- Web ACL
- Telnet ACL
- Action URL
- PPPoE
- OpenVPN
- Supports digit map rules
- Supports flexible call routing policies
- Supports caller / callee number manipulation
- Advanced switching features

FXS

- Interface: RJ11
- Dialing mode: DTMF or pulse
- Pulse dialing: 10 and 20 PPS
- Caller ID: DTMF / FSK standards
- Maximum loop length: 5 km
- Polarity reversal
- Programmable call progress tones

FXO

- Interface: RJ11
- Dialing mode: DTMF or pulse
- Caller ID: DTMF / FSK standards
- Polarity reversal
- Answer delay
- Busy tone detection
- No-current detection

VoIP

- SIP v2.0 (UDP/TCP), RFC3261
- SDP, RTP (RFC2833), RFC3262, RFC3263, RFC3264, RFC3265, RFC3515, RFC2976, RFC3311
- RTP / RTCP, RFC2198, RFC1889
- IPv4 and IPv6
- Outbound proxy
- RFC2806 TEL URI
- RFC3581 NAT, rport
- VLAN 802.1P / 802.1Q
- RFC4028 Session Timer
- DNS SRV / A query / NAPTR query
- NAT traversal: STUN, static / dynamic NAT

Maintenance

- SNMP v1 / v2 / v3
- TR-369
- TR-069
- Auto provisioning
- Web page configuration
- Data backup / restore
- Firmware upgrade via web page
- CDR statistics
- IPOE
- Supports Debug, Info, Error, Warning and Notice system log levels
- Ping / Traceroute test
- Network packet capture
- NTP time synchronization / daylight saving time
- Cloud-based management

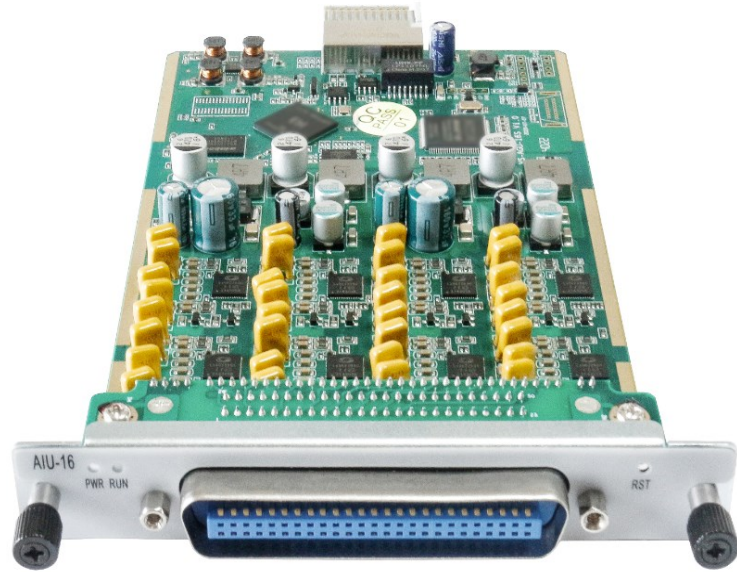
AIU-8 module product details

Port Type	RJ45
Protocol	Standard SIP, TCP/IP
Dimensions	160 × 100 mm
Weight	189 g
Maximum Power Consumption	12 W
Operating Temperature	0°C to 50°C
Operating Humidity	10% to 90%
Storage Temperature	-20°C to 70°C

AIU-16 Analog Gateway module

The AIU-16 gateway module supports PSTN connectivity to VoIP devices and provides 16 FXS channels. It delivers excellent HD voice services through codecs such as G.711A/U, G.723.1, G.729A, G.722, iLBC, OPUS, AMR, and AMR-WB.

The AIU-16 gateway module is compatible with Asterisk, 3CX, FreePBX, FreeSWITCH, and VOS VoIP platforms, helping users reduce telecom and communication costs.



Features

Call Features

- Call waiting
- Blind transfer
- Call transfer
- Call forward on busy
- No-answer call forward
- Unconditional call forward
- Hotline
- Message waiting indicator
- Supports SIP registration to multiple servers
- Call hold
- Do not disturb
- 3-way conference
- Speed dial

FXS

- Interface: RJ11
- Dialing mode: DTMF or pulse
- Pulse dialing: 10 and 20 PPS
- Caller ID: DTMF / FSK standards
- Maximum loop length: 5 km
- Polarity reversal
- Programmable call progress tones

Software Features

- Ring group function
- Web ACL
- Telnet ACL
- Action URL
- PPPoE
- OpenVPN
- Supports digit map rules
- Supports flexible call routing policies
- Supports caller / callee number manipulation
- Advanced switching features

Voice and Fax

- G.711A/U, G.723.1, G.729A, G.722, iLBC, OPUS, AMR, and AMR-WB
- Comfort noise generation (CNG)
- Supports Modem / POS
- Adaptive dynamic jitter buffer
- Programmable gain control
- DTMF types: SIP INFO / RFC4733 / INBAND
- Supports T.38 and pass-through fax
- Echo cancellation (G.168), up to 128 ms

VoIP

- SIP v2.0 (UDP/TCP), RFC3261
- SDP, RTP (RFC2833), RFC3262, RFC3263, RFC3264, RFC3265, RFC3515, RFC2976, RFC3311
- RTP / RTCP, RFC2198, RFC1889
- IPv4 and IPv6
- Outbound proxy
- RFC2806 TEL URI
- RFC3581 NAT, rport
- VLAN 802.1P / 802.1Q
- RFC4028 Session Timer
- DNS SRV / A query / NAPTR query
- NAT traversal: STUN, static / dynamic NAT

Maintenance

- SNMP v1 / v2 / v3
- TR-369
- TR-069
- Auto provisioning
- Web page configuration
- Data backup / restore
- Firmware upgrade via web page
- CDR statistics
- IPOE
- Supports Debug, Info, Error, Warning, and Notice system log levels
- Ping / Traceroute test
- Network packet capture
- NTP time synchronization / daylight saving time
- Cloud-based management

AIU-16 Module Specifications

Port Type	DB232
Protocol	Standard SIP, TCP/IP
Dimensions	160 × 100 mm
Weight	189 g
Maximum Power Consumption	20 W
Operating Temperature	0°C to 50°C
Operating Humidity	10% to 90%
Storage Temperature	-20°C to 70°C

MIU Gateway

The MIU magneto gateway is a powerful voice access device. It is available in two models, MIU-4 and MIU-8, providing 4 and 8 magneto channels respectively. Both models use the standard SIP protocol and can interconnect with standard SIP softswitch systems to provide VoIP / MoIP solutions for magneto telephone users. The device is easy to deploy, highly secure, and resistant to interference, and is widely used in the military and other specialized command and dispatch systems.



Features

- Supports 4 / 8 magneto telephone interfaces
- Supports WEB management configuration
- Supports multiple network protocols and can integrate multiple interface types with the unified communications platform
- Can connect to enterprise IP telephony systems and various unified communication systems to improve communication efficiency
- Can automatically detect and trigger calls on different magneto telephone interfaces for rapid call setup
- Supports automatic off-hook when the remote party is called and automatic hang-up when the remote party hangs up
- The local magneto ring can call a preconfigured phone for communication; hanging up on the local side requires pressing the control button
- The magneto telephone interface uses transformer coupling for strong anti-interference capability
- Adapts to different network environments for stable transmission
- Uses adaptive codec technology to ensure call quality
- Supports SIP v1 (RFC2543) and v2 (RFC3261)
- Provides electrical Ethernet and optical interfaces, 10 / 100BASE-T full / half duplex, auto-adaptive, up to 100 Mbps bandwidth
- Supports DHCP, DNS / DDNS, and NAT networking
- Supports voice activity detection (VAD) and comfort noise generation (CNG)
- Rich logging functions
- DTMF detection
- Supports volume adjustment, gain adjustment, call hold, call waiting, call forward, and caller ID
- Supports SSH remote operation
- Supports configuration file backup and upload
- Compatible with Asterisk, Elastix, 3CX, FreeSWITCH SIP Server, and VOS VoIP platforms

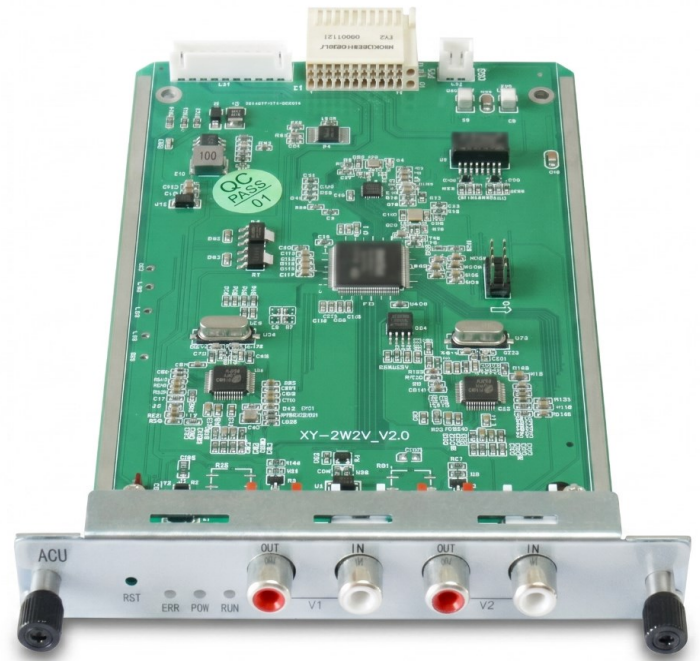
MIU Module Specifications

Protocol	Standard SIP, TCP/IP
Dimensions	160 × 100 mm
Weight	178 g
Maximum Power Consumption	6.5 W
Storage Temperature	-40°C to 125°C
Operating Temperature	0°C to 50°C
Operating Humidity	10% to 90%

ACU Audio Module

The ACU audio module is a powerful voice access device that can easily interface teleconferencing systems with conference-room audio mixer systems. It enables voice pickup and amplification for telephone conferences to be transmitted over telephone lines, meeting video conferencing and teleconferencing needs across industries while ensuring clear voice quality and strong telephone access control capabilities. The ACU audio board supports both PSTN access and SIP line registration, making it an ideal choice for conference-room projects that require high reliability.

The device provides two audio control interfaces. Users can connect external IP phones, computers, central control systems, and other devices to route telephone audio into the conference room, allowing two conference rooms to share one audio access console. The ACU audio board gateway is flexible and simple to deploy, and can be used with straightforward cabling. It supports simultaneous access to PSTN, SIP, and GSM.



Features

System Features

- Strong compatibility
- Echo suppression and noise suppression
- Simultaneous dual audio output
- Unmatched noise elimination technology
- Balanced cost performance and easy system integration
- 1U standardized design saves deployment space
- Patented voice algorithm ensures clear voice quality
- Simple and convenient configuration via WEB GUI
- Multi-line telephony access processing for SIP, PSTN, and GSM

Audio Gateway Uses

- Large conference rooms
- Courtroom conference rooms
- Dispatch command centers
- Military and public security
- Government conference rooms
- Field command conference rooms
- Building PA systems
- Training classrooms
- Command vehicle conference rooms

Strong compatibility

The device supports different telephone line types, including PBX subscriber lines, carrier subscriber lines, and analog user gateway IAD devices. In addition, it supports SIP applications on IP-PBX systems, softswitches, and SIP servers from major vendors, and can be seamlessly integrated with dispatch and command systems, video conferencing, and central control systems.

ACU Audio Module Product Specifications

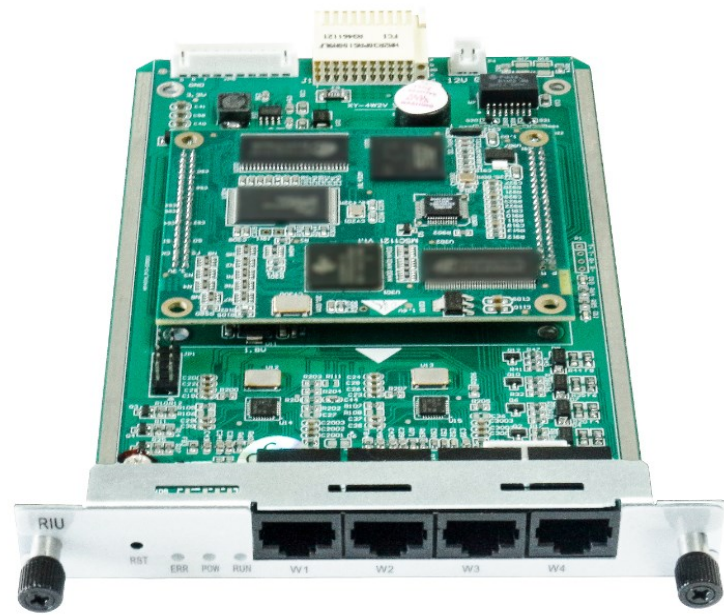
Port Type	AV RCA jack
Port Protocol	Standard SIP, TCP/IP
Dimensions	160*100mm
weight	132g
Capacity	2-channel audio input and output
Voice Codec	G.711、G.729、G.723
Radio Control	PTT、VOX、COR
Distortion	$\leq 1\%$
Signal-to-Noise Ratio	$\geq 70\text{dB}$ (excluding telephone line noise)
Side Tone Control	$\geq 45\text{dB}$
Max Power	2W
Operating Temperature Range	-20°C ~ 60°C
Operating Humidity Range	8% ~ 90% non-condensing
Minimum Ventilation Space	6.4cm

RIU Wireless Trunking Module

The RIU wireless trunking board is a powerful voice access device that enables easy integration between trunked radio systems and telephone systems. Users can conveniently call two-way radios from telephones, and can also place phone calls from two-way radios. The system supports traditional PSTN phone lines and SIP-based VoIP lines, and is easy to deploy and use for true plug-and-play operation.

The wireless trunking board adopts a carrier-grade design with strong networking and audio processing capabilities. Using microcontroller chip technology and electronic switching technology, each control path operates independently and can switch audio signals in and out with high sensitivity, enabling simultaneous access for two radio conversations. The wireless trunking board can be widely used in specialized command and dispatch systems across many industries.

The device provides four radio interfaces using RJ45 connectors and dedicated control cables, and is compatible with mainstream handheld and vehicle-mounted radios from Motorola, Kenwood, and other brands. In addition, drawing on extensive experience in the voice field, the wireless trunking board uses optimized voice algorithms to deliver better call quality than similar products currently on the market.



Features

System Features

- Analog trunking individual calls and group calls
- Patented voice algorithm ensures clear voice quality
- Unmatched noise elimination technology
- Multiple dialing and digit collection rule configurations
- Multi-line telephony access processing for SIP and PSTN
- Standard SIP protocol for interoperability with telephony and dispatch systems
- Adaptive VOX (voice activation) with adjustable sensitivity
- Adjustable input and output volume
- Adjustable line level control
- Voice delay buffer settings to prevent clipped speech and dropped syllables
- Noise suppression, tone detection, and trigger settings
- User-configurable COR and PTT valid signal settings
- Supports WEB-based management
- Strong compatibility with radios from Motorola, Kenwood, and other brands
- Supports SIP INFO floor control
- Supports DTMF dialing
- Supports PTT hotline calling

Strong compatibility

The device supports different telephone line types, including PBX subscriber lines, carrier subscriber lines, and analog user gateway IAD devices. In addition, it supports SIP applications on IP-PBX systems, softswitches, and SIP servers from major vendors, and can be seamlessly integrated with dispatch and command systems, video conferencing, and central control systems.

Radio Gateway Uses

The wireless trunking board can be widely used in command and dispatch systems for public security, armed police, firefighting, the military, railways, civil defense, industrial and mining enterprises, forestry, petroleum, power utilities, and government departments, enabling rapid emergency response and the integration of multiple communication methods.

RIU Wireless Trunking Module Product Details

Port Type	RJ45
Port Protocol	Standard SIP, TCP/IP
Dimensions	160*100mm
Weight	132g
Capacity	4-channel radio or audio input/output
Voice Codec	G.711、G.729、G.723
Radio Control	PTT、VOX、COR
Distortion	$\leq 1\%$
Signal-to-Noise Ratio	$\geq 70\text{dB}$ (excluding telephone line noise)
Side Tone Control	$\geq 45\text{dB}$
Max Power	5W
Operating Temperature Range	-20°C ~ 60°C
Operating Humidity Range	8% ~ 90% non-condensing
Minimum Ventilation Space	6.4cm

DTU-30X Digital Gateway Module

The DTU-30X series includes three models, DTU-301, DTU-302, and DTU-304, which provide 1, 2, and 4 E1 / T1 interfaces respectively and support up to 120 concurrent calls. The digital gateway module is a VoIP trunk voice gateway module designed for carriers and call centers. Used with the UCP series, it connects traditional telephone systems with IP networks and enables seamless interconnection between VoIP PBX systems and PRI / SS7 / R2 networks. Its user-friendly interface and simple operation make it easy for users to configure personalized gateways. Users can also perform secondary development of gateway functions through the API.



Features

System Features

- Provides 1 / 2 / 4 T1 / E1 interfaces, up to 120 concurrent calls
- Signaling: PRI / R2 / SS7
- Supports standard R2 signaling for 24 countries
- Simple and convenient web configuration
- Supported protocols: SIP, TCP, UDP, RTP, SSH, HTTP, HTTPS
- Supports automatic NTP time synchronization
- Supports SSH
- Supports SNMP / TR-069 / TR-369
- Open full-featured API
- Supports port groups
- Supports custom dial plans
- Firmware upgrade via HTTP
- Supports call analysis
- Supports auto provisioning
- Supports automatic channel status display
- Supports backup / upload of configuration files
- Detailed log output
- Supports scheduled reboot
- Supports multiple IPPBX platforms including Asterisk, Issabel, 3CX, FreeSWITCH, VOS, and BroadSoft

Voice and Fax

- G.711A/U, G.723.1, G.729A, iLBC, OPUS, AMR, AMR-WB
- Comfort noise generation (CNG)
- Adaptive (dynamic) jitter buffer
- Programmable gain control
- DTMF types: SIP INFO / RFC4733 / INBAND
- T.38 / pass-through fax

PSTN

- SS7 signaling
- ITU-T, ANSI, ITU-CHINA MTP1 / MTP2 / MTP3, ISUP
- ISDN PRI
- 23B+D (T1), 30B+D (E1), NT or TE, ITU-T Q.921, ITU-T Q.931, Q.Sig
- E1 frame types
- DF, CRC-4, CRC_ITU
- T1 frame types
- Superframe (F12, D3/4), Extended Superframe (F24, ESF)
- Line coding

Network

- Network types: Static IP, DHCP
- IPv4, IPv6, UDP / TCP, DHCP, TFTP, SCP
- HTTP / HTTPS / SSH
- Supports ping and traceroute commands
- Supports web-based network packet capture
- Supports MGT and IP aliases
- Layer 2 QoS and Layer 3 QoS
- Fast DNS SRV switchover
- Loopback network topology hiding

Routing

- Flexible routing configuration
- Supports 512 routing rules
- Supports filtering and manipulation of caller / callee numbers
- Supports adding, modifying, and deleting routes
- E1 / T1 port grouping
- Supports failover switching

VoIP

- SIP v2.0 (UDP/TCP), RFC3261
- SDP, RTP (RFC2833), RFC3262, 3263, 3264, 3265, 3515, 2976, 3311, SIP TLS / SRTP
- RTP / RTCP, RFC2198, 1889
- SIP-T, RFC3372, RFC3204, RFC3398
- TLS / SRTP
- SIP / IMS registration: up to 2000 SIP accounts
- NAT: dynamic NAT, rport

DTU-30X Digital Gateway Module Product Specifications

Parameter	DTU-301	DTU-302	DTU-304
Number of E1 Ports (RJ45)	1	2	4
Dimensions	160*100mm		
Concurrent Calls	30	60	120
Other Ports Provided	2x Fast Ethernet ports, 1x console management port, RST		
Maximum Power	7W	7W	7W
Operating Temperature Range	0C ~ 50°C		
Operating Humidity Range	10% ~ 90% (non-condensing)		
Storage Temperature Range	-20°C ~ 70°C		

WTU Wireless Module

WTU wireless board supports two modes, LTE and GSM, where GSM supports 2G signal and LTE supports 4G signal. In terms of voice encoding, it supports voice codecs including G.711U, G.711A, GSM, G.722, G.726, and G.729, and offers flexible SMS services and HTTP-based APIs. The WTU series wireless gateway modules are fully compatible with VoIP platforms such as Asterisk, 3CX, FreePBX, FreeS-WITCH, and VOS, providing users with more diverse telecom access options while reducing communication costs.



Features

VoIP Features

- SIP v2.0 (UDP/TCP), RFC3261
- Supports SIP and IAX2 protocols
- DTMF modes: RFC2833 / Inband / SIP Info
- Multiple SIP / IAX2 registration modes
- Supported codecs: G.711U, G.711A, GSM, G.722, G.726, G.729
- Includes both SIP client and server modes

Network

- IPv4, UDP / TCP, ARP, DHCP, NTP, DNS, HTTP / HTTPS, SSH
- PPTP VPN
- Ping and traceroute
- Security policies: blacklist, whitelist, security rules

Admin

- WEB management
- SSH maintenance
- Config backup / restore
- Chinese / English UI
- HTTP firmware update
- WEB / SSH password change
- Factory reset

Software Features

- Configure SIP trunks and trunk groups
- Supports Caller ID display and concealment
- Random call intervals
- Per-port call duration limit
- Single outbound call duration limit
- Supports open API protocol
- Supports DISA
- SMSC / SMS / USSD
- Supports PIN verification
- Voice codecs
- Port group management
- Supports SMS remote control
- Supports bulk SMS, email delivery, and automatic retransmission
- Supports SMS forwarding and quick reply
- Send SMS to email
- Outbound proxy
- Supports automatic reboot
- Displays call status
- Mobile number portability (MNP)
- Supports custom scripts and dial plans
- Supports OpenVox cloud platform

WTU Wireless voice module product details		
Product Name	WTU-4(GSM)	WTU-4(LTE)
Module type	GSM	LTE
Number of Channels	4	4
Dimensions (without antenna)	160*100mm	
Weight (without antenna)	164g	164g
Frequency Band(s)	GSM: 850/900/1800/1900MHz	<p>WUT-CE China/India LTE FDD: B1/B3/B5/B8 LTE TDD: B38/B38/B40/B41 WCDMA: B1/B8 TD-SCDMA: B34/B39 CDMA: BC0 GSM: 900/1800MHz</p> <p>WTU-E EMEA/Korea/ Thailand LTE FDD: B1/B3/B5/B7/B8/B20 LTE TDD: B38/B40/B41 WCDMA: B1/B5/B8 GSM: B3/B8</p> <p>WTU-AU Australia/ New Zealand/Taiwan/Brazil LTE FDD: B1/B2/B3/B4/B5/B7/ B8/B28 LTE TDD: B40 WCDMA: B1/B2/B5/B8 GSM: B2/B3/B5/B8</p>
Maximum Power	28W	28W
SIM Card	Hot plugging	
Operating Temperature Range	0°C ~ 40°C	
Operating Humidity Range	10% ~ 90%	
Storage Temperature	-20°C ~ 70°C	

RSU Module

The RSU module is designed to deliver outstanding stability and reliable data protection, making it suitable for application scenarios with demanding requirements for storage continuity and business security. To further enhance system fault tolerance and data redundancy, the RSU must be used together with two SEU units (hard drive expansion boards) and leverages RAID1 data mirroring technology to build a complete and reliable RAID storage system. This solution synchronously writes the same data to two hard drives to achieve real-time mirrored backup. Even if one storage device fails, the system can continue operating normally and ensure that data is not lost. With this highly reliable architecture, the RSU effectively enhances system stability, improves data security, and provides a solid foundation for long-term continuous operation.



Features

System Features

- Uses RAID1 data mirroring technology to improve data security and system reliability.
- Two hard drives are written in real time simultaneously, ensuring that if the primary drive fails, the system can continue running from the mirrored drive.
- Supports redundant storage protection to reduce the risk of data loss caused by single-drive failure.
- Improves disaster recovery capability for critical data without affecting business continuity.
- Suitable for unified communications platform storage scenarios with high stability and reliability requirements.
- Modular design for convenient deployment and maintenance with the RSU storage system.
- Supports long-term stable operation to meet enterprise communications platform requirements for continuous service.

Reliability Benefits

RAID1 achieves dual data protection by writing the same data to two storage media at the same time to form a fully mirrored copy. When one hard drive fails, the other can immediately take over data read and write operations, ensuring continued system availability. This mechanism is especially suitable for communication platform environments that require highly reliable storage and stable operation.

Application Scenarios

The RSU module is mainly used for data storage and protection scenarios on the UCP platform, and is suitable for deployment environments that need to prevent single-drive failures, strengthen data security, and improve business continuity. Used together with two SEU units, it can build a stable and reliable RAID1 storage solution.

RSU Module product details

Dimension	160*100mm
Mode	Supports RAID1
Mode Switching	Select via jumper cap: Position 0: AHCI Clear Mode (used to clear RAID) Position 1: RAID1 Mode (normal operation)