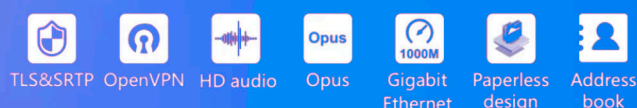




SIP-R20

- ✓ 2.3-inch LCD display with backlight
- ✓ 2 SIP lines, 13 programmable keys
- ✓ Optimized noise reduction, high fidelity sound quality
- ✓ 2.4G&5.8G WiFi module is optional
- ✓ One-stop remote address book management
- ✓ Simple appearance combines elegance and functionality



Explore the SIP-R20 Series - Unified Communication

SIP-R20 series phone, simple design, intuitive operation, redefine enterprise IP communication, designed for small and medium-sized enterprises to create efficient and intelligent communication experience.

Brightening characteristic

Efficient communication: Dual SIP lines, efficient call management.

Clear display: 2.3-inch backlit LCD screen for a striking and clear visual experience.

Simple operation: 13 programmable keys, simplify the operation process, improve work efficiency.

Smooth collaboration: Support for 5-party conference calls, so that team collaboration is more smooth.

Powerful: Cover all advanced call functions to meet diverse office needs.

Security: OpenVPN, TLS & SRTP encryption technology is integrated to ensure call security, so that every conversation is at ease.

Convenient management: The advanced LDAP technology is integrated to achieve integrated remote address book management, simplifying contact access and synchronization, and improving communication efficiency.

Seamless connection: DSS expansion disk, speed dial, real-time monitoring of user status.

Flexible configuration: PoE power supply, 2.4G/5.8G Wi-Fi, free desktop space, flexible deployment.

Advanced audio experience

- Hd sound quality: Advanced noise reduction technology to face to face

Phone function

- 2 SIP lines
- Multi-language selection (Chinese/English/Turkish/Korean/Russian/Traditional Chinese/French/Italian/German/Portuguese/Polish/Thai/Hindi/Arabic/Japanese/Spanish/Vietnamese)
- Caller ID with display name
- Custom DSS key, BLF monitoring, hotline, Speed dial
- Handle/hands-free/headphone mode
- Call hold, mute, blind transfer, inquiry transfer
- Do not disturb, call forwarding, call waiting
- Multicast, intercom, text message, keypad lock, emergency call
- Direct IP call, redial, callback, auto answer
- Five-party conference, line sharing, dialing rules
- 10 types of built-in ringtones / upload custom ringtones
- Manually or automatically synchronize network time
- Action URL & Action UR

Audio characteristic

- High-fidelity sound quality: HD handle, HD hands-free
- Broadband codec: G.722, Opus
- Narrowband codec: G.711(A/μ), G.723.1, G.729AB, G.726, iLBC
- DTMF: In-band transmission (In-band), Out-of-band transmission (RFC 2833), SIP INFO
- Full-duplex hands-free
- Acoustic Echo Cancellation (AEC)
- Voice Activity Detection (VAD)
- Comfort noise generation (CNG)
- Packet loss compensation (PLC)
- Background noise detection (BNE)
- Automatic gain control (AGC)
- Dynamic adaptive RTP jitter buffer up to 300ms

Address book

- Blacklist
- XML/LDAP address book
- Contact smart search
- Contact export/import
- Local contacts (1000 entries can be stored)
- Call history: dialed/received/missed/ forwarded calls(1200 records in total)
- RTCP-XR, VQ-RTCPXR
- OpenVPN, IEEE802.1X

IP PBX Features

- BLF, voice message
- Anonymous call, anonymous call rejection
- Call park, call pickup
- Server-based recording function

Configuration and maintenance

- Web configuration, Phone configuration
- Automatic deployment: Support FTP/ TFTP/HTTP/HTTPS/SIP PnP/TR-069
- BIN configuration
- Contact configuration export/import
- CSTA remote control
- Network packet capture and diagnosis
- Custom/full restore factory settings
- System firmware upgrade

Network characteristics

- Support IPv4 / IPv6 / IPv4&IPv6
- IP address allocation mode: static IP / DHCP
- Virtual Local Area Network: VLAN
- Data link layer protocol: CDP / LLDP
- SIP v1 (RFC2543), v2 (RFC3261)
- Support redundant servers
- Support ICE, STUN (NAT session traversal)
- Support UDP / TCP / TLS transmission
- Support SRTP encryption protocol
- SIP trust server (anti-attack mechanism)
- HTTP/HTTPS
- TR-069 network management protocol
- SNTP automatically synchronizes network dates and times
- QoS: QoS: 802.1p/Q tagging (VLAN), Layer3/ToS/DSCP
- HTTPS certificate management
- Support configuration file encryption
- Support information verification mechanism
- RTCP-XR, VQ-RTCPXR
- OpenVPN, IEEE802.1X
- WiFi(2.4G&5G) optional

Phone interface

- 1 x RJ-9 (4P4C) handle interface
- 1 x RJ-9 (4P4C) headset interface
- 2 x RJ-45 10/100M adaptive Ethernet port
- DC power interface
- USB2.0 interface

Product specification

- Phone main screen: 2.3 inches (132x64) LCD display with backlight
- Information indicator: LED red light
- Phone keyboard:
 - 2 SIP line keys (with LED lights, programmable)
 - 4 customizable soft keys
 - 12 standard telephone number keys
 - 6 function keys: headset, voice message, mute, Call forwarding, redialing, hands-free
 - 5 navigation keys
 - 2 volume up/down adjustment keys
- Color: business gray
- Installation method
 - Desktop
 - Wall-mounted
- Desktop size: 187mm*184mm*147mm
- Phone power consumption: 2~3W
- PoE: IEEE 802.3af/IEEE 802.3at
- Working environment humidity: 10~95%
- Working temperature:
 - 10~50° C (+14~122° F)

Contact Us

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