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

TLS&SRTP | OpenVPN | HD audio | Opus | Gigabit Ethernet | Paperless design | Address book

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)

SIP-R20

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