virtuallandline office

Yealink SIP-T21P

Entry-level IP phone with 2 Lines & HD voice

Yealink's new SIP-T21P takes entry-level IP phones to a level never achieved before. Making full-use of high-quality materials, plus an extra-large 132x64-pixel graphical LCD showing a clear 5-line data display, it offers a smoother user experience, much more visual information at a glance, plus HD Voice characteristics. Dual 10/100 Mbps network ports with integrated PoE are ideal for extended network use. The T21P supports two VoIP account, simple, flexible and secure installation options, plus support for IPv6, Open VPN and a redundancy server. It also operates with SRTP/ HTTPS/ TLS, 802.1x. As a very cost-effective and powerful IP solution, the T21P maximizing productivity in both small and large office environments.

HD IPv6

Key Features and Benefits

HD Audio

Yealink HD Voice refers to the combination of software and hardware design as well as the implementation of wideband technology to maximizes the acoustic performance. Coupled with advanced acoustic clarity technology such as full duplex, echo cancellation, adaptive jitter buffer etc. Provide clearer, more lifelike voice communications.

Enhanced Call Management

The SIP-T21P supports vast productivity-enhancing feature such as XML Browser, call park, call pickup, BLF, call forward, call transfer, 3-way conference. Which make it the natural and obvious efficiency tool for today's busy small and large offices environment.

Efficient Installation and Provisioning

Integrated IEEE 802.3af Power-over-Ethernet allows easy deployment with centralized powering and backup.The SIP-T21P support the FTP, TFTP, HTTP, and HTTPS protocols for file provisioning and are configured by default to use Trivial File Transfer Protocol (TFTP), supports AES encrypted XML configuration file.

Highly secure transport and interoperability

The Communicator uses SIP over Transport Layer Security (TLS/SSL) to provide service providers the latest technology for enhanced network security. The range is certified compatible with 3CX, Asterisk and Broadsoft Broadworks, ensuring excellent compatibility with leading soft switch suppliers.

- Yealink HD Voice
- 132x64-pixel graphical LCD
- Two-port 10/100 Ethernet Switch
- PoE support
- Up to 2 SIP accounts
- Headset support
- Wall mountable
- Simple, flexible and secure provisioning options

SIP-T21P Specifications

Audio Features
- HD voice: HD handset, HD speaker
- Wideband codec: G.722
- Narrowband codec: G.711(A/μ), G.723.1, G.729AB,
- 726, iLBC
- DTMF: In-band, Out-of-band(RFC 2833) and SIP INFO
- Full-duplex hands-free speakerphone with AEC
- VAD, CNG, AEC, PLC, AJB, AGC

Phone Features
- 2 VoIP accounts
- Call hold, mute, DND
- One-touch speed dial, hotline
- Call forward, call waiting, call transfer
- Group listening, SMS, emergency call
- Redial, call return, auto answer
- Local 3-way conferencing
- Direct IP call without SIP proxy
- Ring tone selection/import/delete
- Set date time manually or automatically
- Dial plan
- XML Browser, action URL/URI

Directory
- Loal phonebook up to 1000 entries
- Black list
- XML/LDAP remote phonebook
- Intelligent search method
- Phonebook search/import/export
- Call history: dialed/received/missed/forwarded

IP-PBX Features
- Busy Lamp Field (BLF)
- Bridged Line Apperance(BLA)
- Anonymous call, anonymous call rejection
- Message Waiting Indicator (MWI)
- Voice mail, call park, call pickup
- Intercom, paging, music on hold, emergency call
- Call completion, call recording, hot-deskin

Display and Indicator
- 132x64-pixel graphical LCD
- LED for call and message waiting indication
- One-color (green) illuminated LEDs for line status information
- Intuitive user interface with icons and soft keys
- National language selection
- Caller ID with name, number

Feature keys
- 2 line keys with LED
- 6 features keys: message, headset, redial, tran, mute, hands-free speakerphone
- 6 navigation keys
- Volume control keys

Interface
- 2xRJ45 10/100M Ethernet ports
- Power over Ethernet (IEEE 802.3af), class 2
- 1xRJ9 (4P4C) handset port
- 1xRJ9 (4P4C) headset port

Other Physical Features
- Wall mountable
- External universal AC adapter (optional): AC 100~240V input and DC 5V/600mA output
- Power consumption (PSU): 1.2-1.9W
- Power consumption (PoE): 1.8-2.3W
- Dimension(W*D*H*T): 209mm*188mm*150mm*41mm
- Operating humidity: 10~95%
- Operating temperature: -10~50°C

Package Features
- Qty/CTN: 10 PCS
- N.W/CTN: 10.8 kg
- G.W/CTN: 12.3 kg
- Giftbox size: 215mm*200mm*118mm
- Carton Meas: 615mm*436mm*208mm

Management
- Configuration: browser/phone/auto-provision
- Auto provision via FTP/TFTP/HTTP/HTTPS for mass deploy
- Auto-provision with PnP
- Zero-sp-touch
- Phone lock for personal privacy protection
- Reset to factory, reboot
- Package tracing export, system log

Network and Security
- SIP v1 (RFC2543), v2 (RFC3261)
- Call server redundancy supported
- NAT transverse: STUN mode
- Proxy mode and peer-to-peer SIP link mode
- IP assignment: static/DHCP
- HTTP/HTTPS web server
- Time and date synchronization using SNTP
- UDP/TCP/DNS-SRV(RFC 3263)
- QoS: 802.1p/Q tagging (VLAN), Layer 3 ToS DSCP
- SRTP for voice
- Transport Layer Security (TLS)
- HTTPS certificate manager
- AES encryption for configuration file
- Digest authentication using MD5/MD5-sess
- OpenVPN, IEEE802.1X
- IPv6

Certifications