Asperient

Enterprise HD IP Phone

Yealink SIP-T23G features intuitive user interface and enhanced functionality which make it easy for people to interact and maximize productivity. Yealink HD technology enables rich, clear, life-like voice communications, outsourced management options, flexible deployment and third-party communications applications. As a cost effective IP solution, it helps users to streamline business processes, delivery a powerful, security and consistent communication experience for small and large offices environment.

http://asperient.com/phones

HD Audio
Yealink HD Voice refers to the combination of software and hardware design as well as the implementation of wideband technology to maximize 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-T23G 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-T23G 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.

Features

- Yealink HD Voice
- Dual-port Gigabit Ethernet
- PoE support
- 132x64-pixel graphical LCD
- Up to 3 SIP accounts
- Supports Dual-color LEDs
- Headset support
- Wall mountable
- IPv6 support
- Paperless design
- Simple, flexible and secure provisioning options

Specification

Audio Features
- HD voice: HD handset, HD speaker
- Wideband codec: G.722
- Narrowband codec: G.711(A/μ), G.729AB,G.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
- 3 SIP accounts
- Call hold, mute, DND
- One-touch speed dial, hotline
- Call forward, call waiting, call transfer
- Group listening, SMS
- 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, off hook hot line
- XML Browser, Action URL/URI
- Integrated Screenshots
- RTCP-XR

Directory
- Local 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
- Hot-desking, emergency call
- Message Waiting Indicator (MWI)
- Voice mail, call park, call pickup
- Intercom, paging, music on hold
- Call completion, call recording

Display and Indicator
• 132x64-pixel graphical LCD with backlight
• LED for call and message waiting indication
• Dual-color (red or 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
• 3 line keys with LED
• 5 features keys: message, headset, redial, tran,hands-free speaker-phone
• 6 navigation keys
• Volume control keys

Interface
• Dual-port Gigabit Ethernet
• 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): 0.72-1.56W
• Power consumption (PoE): 0.96-1.98W
• Dimension(W*D*H*T):209mm*188mm*150mm*41mm
• Operating humidity: 10~95%
• Operating temperature: -10~50°C

Management
• Configuration: browser/phone/auto-provision
• Auto provision via FTP/TFTP/HTTP/HTTPS
for mass deploy
• Auto-provision with PnP
• Zero-sp-touch TR-069
• Phone lock for personal privacy protection
• Reset to factory, reboot
• Package tracing export, system log

Network and Security

- SIP v1 (RFC2543), v2 (RFC3261)
- 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