

## Affordable IP Phone with Three Lines & HD voice

The SIP-T40P is a feature-rich SIP phone for enhancing daily business processes and operations. With a revolutionary new design that supports flexible and secure provisioning (the phone utilizes industry-standard encryption protocols for users to provision and perform software upgrades both in-house and remotely), this advanced IP phone is intuitively designed with ease of use in mind. With programmable keys, the SIP-T40P boasts extensive productivity-enhancing features such as Power over Ethernet (PoE) support, superb high definition (HD) sound quality and a rich visual experience.



Optima  
HD Voice



Paperless

### Key Features and Benefits

#### Revolutionary New Design

Yealink's SIP Phones continue to evolve alongside the company. From the initial strategizing phase to the development stages, the robust T4 Series, including the SIP-T40P, ensures an optimum user experience first and foremost. Even in the smallest of details, such as the solution's new paper label-free design, simple-to-use foot stand, non-slip rubber feet and ergonomic recessed buttons, ease of use is at the forefront of the product.

#### HD Audio

Yealink's Optima HD Voice refers functionality involves to the combination of software and hardware design as well as the implementation of wideband technology and advanced acoustic elements such as duplex, echo cancellation and adaptive jitter buffer to in order to maximizes the product's acoustic performance. This creates an amazing unparalleled face-to-face, live experience.

#### Enhanced Call Management

The SIP-T40P supports vast productivity-enhancing feature such as SCA, BLF List, call forward, call transfer, three-way conference. Furthermore, this SIP phone provides support for the Yealink YHS32. What's more, with EHS36 the user can conveniently control his or her phone through wireless headset.

#### Efficient Installation and Provisioning

Integrated IEEE 802.3af (PoE) allows for easy deployment with centralized powering and backup. The SIP-T40P supports FTP, TFTP, HTTP, and HTTPS protocols for file provisioning and is configured by default to use Trivial File Transfer Protocol (TFTP). Additionally, the phone supports AES encrypted XML configuration files.

#### Highly secure transport and interoperability

The SIP-T40P uses SIP over Transport Layer Security (TLS/SSL) in order to arm service providers the latest technology for enhanced network security. The entire T40P phone range is certified compatible with 3CX, Asterisk and BroadSoft Broadworks, supporting excellent compatibility with leading soft switch suppliers to ensure the most seamless deployment and installation possible.

- > Revolutionary new design
- > Yealink Optima HD voice
- > 132x64-pixel graphical LCD with backlight
- > Up to three SIP accounts
- > Paper label-free design
- > PoE support
- > Headset, electronic hook switch (EHS) support
- > Integrated stand with two adjustable angles
- > Wall mountable
- > Simple, flexible and secure provisioning options

## Audio Features

- > HD voice: HD handset, HD speaker
- > Wideband codec: G.722
- > Narrowband codec: G.711(A/μ), G.729AB, G.726
- > 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 VoIP accounts
- > One-touch speed dial, redial
- > Call forward, call waiting
- > Call transfer, call hold
- > Call return, group listening, SMS
- > Mute, auto answer, DND
- > 3-way conference call
- > Direct IP call without SIP proxy
- > Ring tone selection/import/delete
- > Hotline, emergency call
- > Set date time manually or automatically
- > Dial Plan
- > XML Browser
- > Action URL/URI
- > RTCP-XR (RFC3611) ,VQ-RTCPXR (RFC6035)

## 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 Appearance(BLA)
- > Anonymous call, anonymous call rejection
- > Hot-desking
- > Message Waiting Indicator (MWI)
- > Voice mail
- > Call park, call pickup
- > Intercom, paging
- > Music on hold
- > Call recording

## Display and Indicator

- > 2.3" 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, mute, redial, hands-free speakerphone
- > 4 context-sensitive "soft" keys
- > 6 navigation keys
- > 2 volume control keys
- > Mute key
- > Headset key
- > Hands-free speakerphone key

## Interface

- > 2xRJ45 10/100M Ethernet ports
- > 1xRJ9 (4P4C) handset port
- > 1xRJ9 (4P4C) headset port
- > 1XRJ12 (6P6C) EHS port
- > Power over Ethernet (IEEE 802.3af), Class 2

## Other Physical Features

- > Stand with 2 adjustable angles
- > Wall mountable
- > External universal AC adapter (optional): AC 100~240V input and DC 5V/600mA output
- > Power consumption (PSU): 1.05-3.23W
- > Power consumption (PoE): 1.7-3.2W
- > Dimension(W\*D\*H\*T): 212mm\*189mm\*175mm\*54mm
- > Operating humidity: 10~95%
- > Operating temperature: -10~50°C

## Package Features

- > Qty/CTN: 5 PCS
- > N.W/CTN: 5.7 kg
- > G.W/CTN: 6.4 kg
- > Giftbox size: 246mm\*223mm\*120mm
- > Carton Meas: 620mm\*256mm\*233mm

## Management

- > Configuration: browser/phone/auto-provision
- > Auto provision via FTP/TFTP/HTTP/HTTPS for mass deploy
- > Auto-provision with PnP
- > BroadSoft device management
- > 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)
- > 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

