

Affordable and Easy-to-use

- TI TITAN chipset and TI voice engine
- 2 programmable keys, Auto provision
- 1xLAN, 1xdual-color LED, Phone label
- Power over Ethernet, Wall-mountable



The SIP-T18P is the simply IP phone that equipped with TI TITAN chipset, comprehensive telephony features, PoE, auto provision, and interoperable with the leading IP-PBX and soft switch. It is economical and can be desktop or wall mounted.

Designed for working environment needing a basic feature IP phone, the SIP-T18P is a very cost-effective choice for small business, SOHO and Hotel, college apartment, supermarket, and warehouse etc.

Phone Features

1 VoIP account, Hotline
Call waiting, Call transfer, Call forward
Call hold, Mute, Redial, DND
3-way conferencing, Speed dial
Direct IP call without SIP proxy
Volume control, Ringtone selection
Tone scheme, System log export
Integrated Voice Response System

IP PBX System Integration

Music on hold
Call park, Call pickup
Dial plan, Dial now
Voicemail
Message Waiting Indication (MWI)
Distinctive ringtone

Voice Features

Wideband codec: G.722
Narrowband codec: G.711μ/A, G.726,
G.729AB, G.723.1
VAD, CNG, PLC, AJB, AGC

Network Features

SIP v1 (RFC2543), v2 (RFC3261)
NAT Traversal: STUN mode
DTMF: In-band, out-of band (RFC2833) and SIP INFO
Proxy mode and peer-to-peer SIP link mode
IP Assignment: Static/DHCP
TFTP/DHCP client
Telnet/HTTP server
DNS client

Management

Built-in HTTP web server
Configuration: browser/phone/auto-provision/IVR
Auto provision via TFTP/FTP/HTTP/PnP
Auto provision for firmware, configuration, ringtone etc

Security

QoS: IEEE 802.1p/q tagging (VLAN), Layer 3ToS
Digest authentication using MD5/MD5-sess
Secure configuration file via AES encryption
Admin/user configuration mode

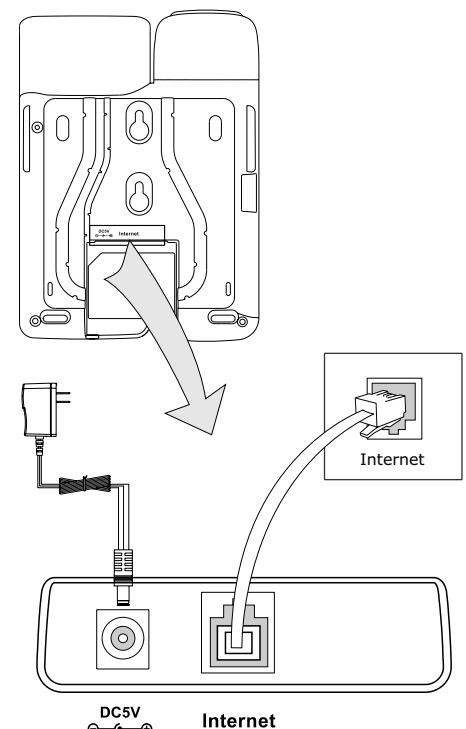
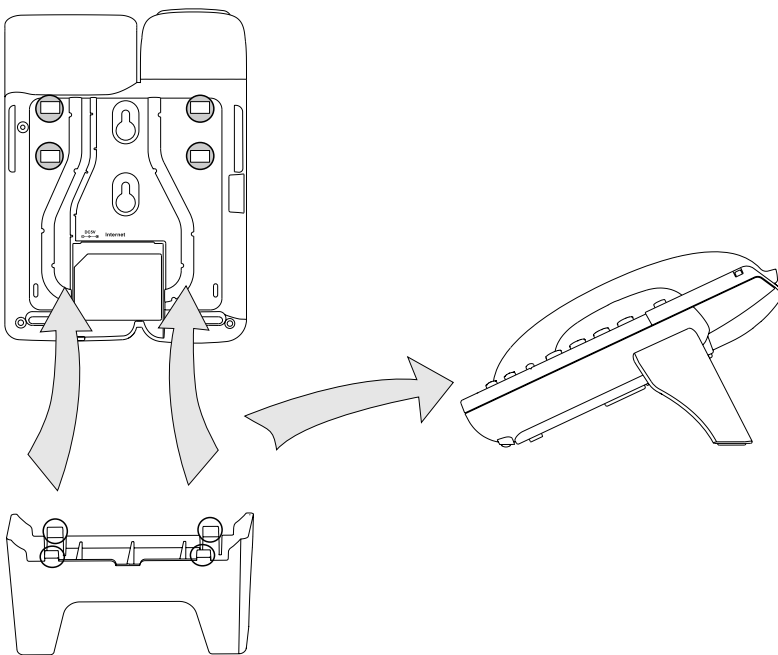
Physical Features

TI TITAN chipset
20 keys including 2 programmable keys
4 feature keys (Transfer/Hold/Mute/Redial)
Phone label
1xdual-color LED
1xRJ45 10/100M Ethernet port
Desktop with footstand (detachable),
Wall-mountable
Power adapter: AC 100~240V input
and DC 5V/1.2A Output
Power over Ethernet
(IEEE 802.3af class 1) optional
Operating humidity: 10~95%
Storage temperature: up to 60°C

Package Features

Qty/CTN: 10 PCS
N.W/CTN: xx KG
G.W/CTN: xx KG
Measurement: 0.054CMB
Carton Meas: 520x470x220MM

Certifications



Cost-effective Enterprise IP Phone with 2 Lines

- TI TITAN chipset and TI voice engine
- 2 VoIP accounts, 2x15 characters LCD
- HD Voice: HD Codec, HD speaker, HD handset
- IPV6, BLF/BLA
- 2xLAN, PoE, Headset, Wall-mountable



Yealink expands its lineup of IP phones with a new entry-level product SIP-T20P. It is equipped with TI TITAN chipset and 2x15 characters LCD, offers 2 VoIP accounts, high-definition voice, broad range of voice codecs, security protection for privacy, rich features including, headset, PoE, PnP Auto-provision, and seamlessly work with the leading IP-PBX and soft switch.

It allows users to make calls in a simple, convenient and reliable manner and fully meet the requirement in which the basic business features are required. What is more, SIP-T20P is easy to install and inexpensive to start up for corporate office and residential users.

Phone Features

2 VoIP accounts, Hotline, Emergency call
Call hold, Call waiting, Call forward, Call return
Call transfer (blind/semi-attended/attended)
Caller ID display, Redial, Mute, DND
Auto-answer, 3-way conferencing
Speed dial, Voicemail
Message Waiting Indication (MWI) LED
Tone scheme, Volume control
Direct IP call without SIP proxy
Ring tone selection/import/delete
Phonebook (300 entries), Black list
Call history: dialed/received/missed/forwarded
Menu-driven user interface
Localized language and input method
Soft keys programmable
Headset (Call center mode)
Multicast IP Paging

IP PBX System Integration

Busy lamp field (BLF), BLF list
Bridged line appearance (BLA)
DND&Forward synchronization
Intercom, Paging, Music on hold
Call park, Call pickup
Call recording, Call completion
Group listening, Group pickup
Anonymous call, Anonymous call rejection
Network conference
Distinctive ringtone
Dial Plan, Dial-now

Codecs and Voice Features

Wideband codec: G.722
Narrowband codec: G.711 μ /A, G.723.1
G.726, G.729AB
VAD, CNG, AEC, PLC, AJB, AGC
Full-duplex speakerphone with AEC

Network Features

SIP v1 (RFC2543), v2 (RFC3261)
IPV6
DNS SRV (RFC3263)
Redundant server support
NAT Traversal: STUN mode
DTMF: In-Band, RFC2833, SIP Info
Proxy mode and peer-to-peer SIP link mode
IP Assignment: Static/DHCP/PPPoE
Bridge/router mode for PC port
TFTP/DHCP/PPPoE client
Telnet/HTTP/HTTPS server
DNS client
NAT/DHCP server

Management

Auto-provision via FTP/TFTP/HTTP/HTTPS
Auto-provision with PnP
SNMP V1/2
TR069 optional
Configuration: browser/phone/auto-provision
Factory configuration customized
Trace package and system log export
Action URL & Active URI

Security

802.1x, VLAN QoS (802.1pq), LLDP
Transport Layer Security (TLS)
HTTPS (server/client)
SRTP (RFC3711)
Digest authentication using MD5/MD5-sess
Secure configuration file via AES encryption
Phone lock for personal privacy protection
Admin/VAR/User 3-level configuration mode

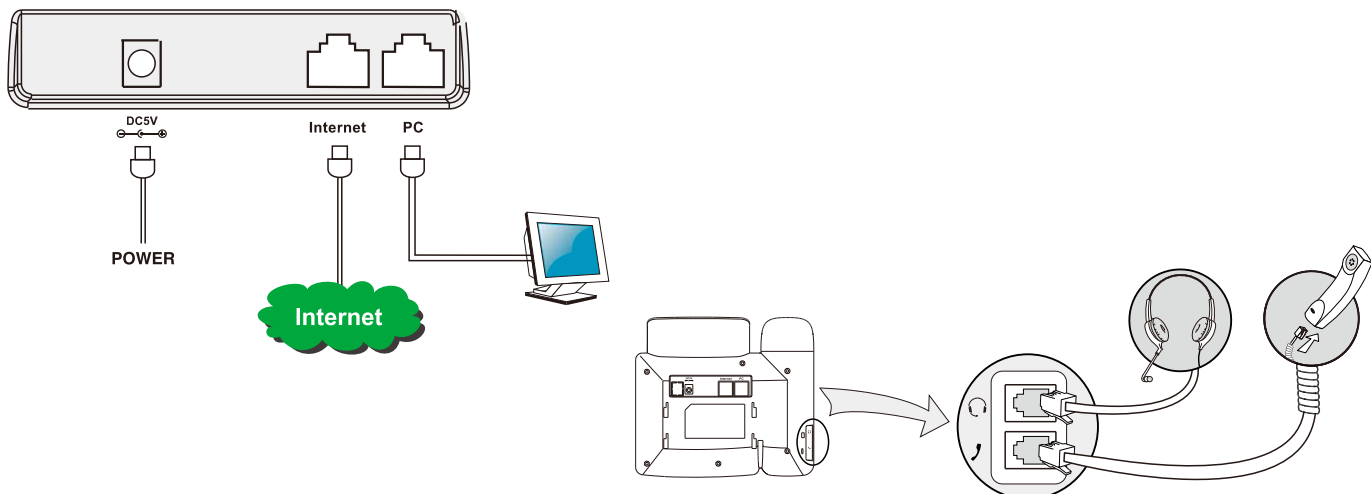
Physical Features

TI TITAN chipset
3-line LCD with an icon line and
2x15 characters lines
31 keys including 9 function keys
4 LEDs: 1xpower, 2xline, 1xmessage
1xRJ9 handset port
1xRJ9 headset port
2xRJ45 10/100M Ethernet ports
Wall-mountable
Power adapter: AC 100~240V input and
DC 5V/1.2A Output
Power over Ethernet (IEEE 802.3af)
Power consumption: 1.4~2.6W
Net weight: 0.77kg
Dimension: 185 x 200 x 90mm
Operating humidity: 10~95%
Storage temperature: up to 60°C

Package Features

Qty/CTN: 10 PCS
N.W/CTN: 11.950KG
G.W/CTN: 13.300KG
Measurement: 0.083CMB
Carton Meas: 565x550x235MM

Certifications



Professional IP phone with 3 Lines & HD voice

- TI TITAN chipset and TI voice engine
- 3 VoIP accounts, 132x64 graphic LCD
- HD Voice: HD Codec, HD speaker, HD handset
- IPV6, BLF/BLA, XML Browser, Hot-desking
- 2xLAN, PoE, Headset, Wall-mountable



Yealink SIP-T22P features intuitive user interface and enhanced functionality which make it easy for people to interact and maximize productivity. With TI TITAN chipset and TI leading VoIP voice engine, it enables enhanced high-definition audio, 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.

Phone Features

3 VoIP accounts, Hotline, Emergency call
Call hold, Call waiting, Call forward, Call return
Call transfer (blind/semi-attended/attended)
Caller ID display, Redial, Mute, DND
Auto-answer, 3-way conferencing
Speed dial, SMS, Voicemail
Message Waiting Indication (MWI) LED
Tone scheme, Volume control
Direct IP call without SIP proxy
Ring tone selection/import/delete
Phonebook (300 entries), Black list
Call history: dialed/received/missed/forwarded
Menu-driven user interface
Localized language and input method
Soft keys programmable

Advanced Features

XML phonebook search/import
LDAP phonebook
XML Browser
PUSH XML
Action URL & Active URI
Headset (Call center mode)
Multicast IP Paging

IP PBX System Integration

Busy lamp field (BLF), BLF list
Bridged line appearance (BLA)
DND&Forward synchronization
Intercom, Paging, Music on hold
Call park, Call pickup
Call recording, Call completion
Group listening, Group pickup
Anonymous call, Anonymous call rejection
Network conference
Distinctive ringtone
Dial Plan, Dial-now

Codecs and Voice Features

Wideband codec: G.722
Narrowband codec: G.711 μ /A, G.723.1
G.726, G.729AB
VAD, CNG, AEC, PLC, AJB, AGC
Full-duplex speakerphone with AEC

Network Features

SIP v1 (RFC2543), v2 (RFC3261)
IPV6
DNS SRV (RFC3263)
Redundant server support
NAT Traversal: STUN mode
DTMF: In-Band, RFC2833, SIP Info
Proxy mode and peer-to-peer SIP link mode
IP Assignment: Static/DHCP/PPPoE
Bridge/router mode for PC port
TFTP/DHCP/PPPoE client
Telnet/HTTP/HTTPS server
DNS client, NAT/DHCP server
Logout

Management

Auto-provision via FTP/TFTP/HTTP/HTTPS
Auto-provision with PnP
SNMP V1/2 , TR069 optional
Configuration: browser/phone/auto-provision
Factory configuration customized
Trace package and system log export

Security

802.1x, VLAN QoS (802.1pq)
Transport Layer Security (TLS)
HTTPS (server/client), SRTP (RFC3711)
Digest authentication using MD5/MD5-sess
Secure configuration file via AES encryption
Phone lock for personal privacy protection
Admin/VAR/User 3-level configuration mode

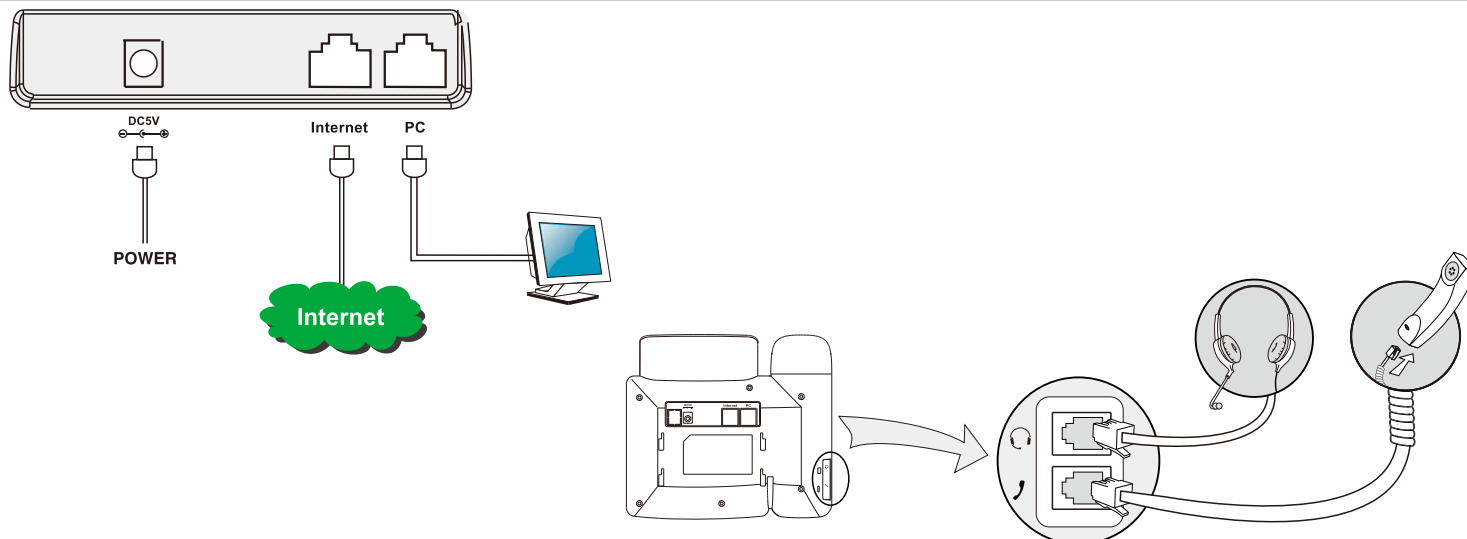
Physical Features

TI TITAN chipset
132x64 graphic LCD
32 keys including 4 soft keys
5 LEDs: 1xpower, 3xline, 1xmessage
1xRJ9 handset port
1xRJ9 headset port
2xRJ45 10/100M Ethernet ports
Wall-mountable
Power adapter: AC 100~240V input and
DC 5V/1.2A Output
Power over Ethernet (IEEE 802.3af)
Power consumption: 1.4-2.6W
Net weight: 0.77kg
Dimension: 185 x 200 x 90mm
Operating humidity: 10~95%
Storage temperature: up to 60°C

Package Features

Qty/CTN: 10 PCS
N.W/CTN: 12.020KG
G.W/CTN: 14.100KG
Measurement: 0.083CMB
Carton Meas: 565x550x235MM

Certifications



Advanced IP Phone with 3 lines & HD voice

- TI TITAN chipset and TI voice engine
- 3 VoIP accounts, 132x64 graphic LCD
- HD Voice: HD Codec, HD speaker, HD handset
- IPV6, BLF, XML Browser, Hot-desking, OpenVPN
- 2xLAN, PoE, Headset, Expansion module



Yealink SIP-T26P is an advanced IP phone which designed for maximum productivity and efficiency in the everyday business environment. It is equipped with the TI TITAN chipset, offers high definition voice quality through TI voice engine, HD handset, HD speaker and HD codec (G.722). Built-in 10 DSS keys for programmed as the IP-PBX features like BLF/BLA, intercom, call pickup, XML browser, Hot desking, etc. Six navigation keys and four soft keys help you to use the phone easily. Ten dedicated functional keys provide you with direct access to the functions for audio and call control. Moreover, SIP-T26P has the rich external interfaces including 2xLAN, PoE, headset and expansion module ports, supports 802.1x, Open VPN, etc security standards.

Phone Features

3 VoIP accounts, Hotline, Emergency call
Call hold, Call waiting, Call forward, Call return
Call transfer (blind/semi-attended/attended)
Caller ID display, Redial, Mute, DND
Auto-answer, 3-way conferencing
Speed dial, SMS, Voicemail
Message Waiting Indication (MWI) LED
Tone scheme, Volume control
Direct IP call without SIP proxy
Ring tone selection/import/delete
Phonebook (300 entries), Black list
Call history: dialed/received/missed/forwarded
Menu-driven user interface
Localized language and input method
Soft keys programmable
Supports up to 6 expansion modules(EXP38 and EXP39)
Supports Wireless Headset Adapter(EHS36)

Advanced Features

XML phonebook search/import
LDAP phonebook
XML Browser, Hot-desking
PUSH XML
Action URL & Active URI
Headset (Call center mode)
Multicast IP Paging

IP PBX System Integration

Busy lamp field (BLF), BLF list
Bridged line appearance (BLA)
DND&Forward synchronization
Intercom, Paging, Music on hold
Call park, Call pickup
Call recording, Call completion
Group listening, Group pickup
Anonymous call, Anonymous call rejection
Network conference
Distinctive ringtone
Dial Plan, Dial-now

Codecs and Voice Features

Wideband codec: G.722
Narrowband codec: G.711 μ /A, G.723.1
G.726, G.729AB
VAD, CNG, AEC, PLC, AJB, AGC
Full-duplex speakerphone with AEC

Network Features

SIP v1 (RFC2543), v2 (RFC3261)
IPv6
DNS SRV (RFC3263)
Redundant server support
NAT Traversal: STUN mode
DTMF: In-Band, RFC2833, SIP Info
Proxy mode and peer-to-peer SIP link mode
IP Assignment: Static/DHCP/PPPoE
Bridge/router mode for PC port
TFTP/DHCP/PPPoE client
Telnet/HTTP/HTTPS server
DNS client, NAT/DHCP server
Logout

Management

Auto-provision via FTP/TFTP/HTTP/HTTPS
Auto-provision with PnP
SNMP V1/2, TR069 optional
Configuration: browser/phone/auto-provision
Factory configuration customized
Trace package and system log export

Security

Open VPN, 802.1x, VLAN QoS (802.1pq)
Transport Layer Security (TLS)
HTTPS (server/client), SRTP (RFC3711)
Digest authentication using MD5/MD5-sess
Secure configuration file via AES encryption
Phone lock for personal privacy protection
Admin/VAR/User 3-level configuration mode

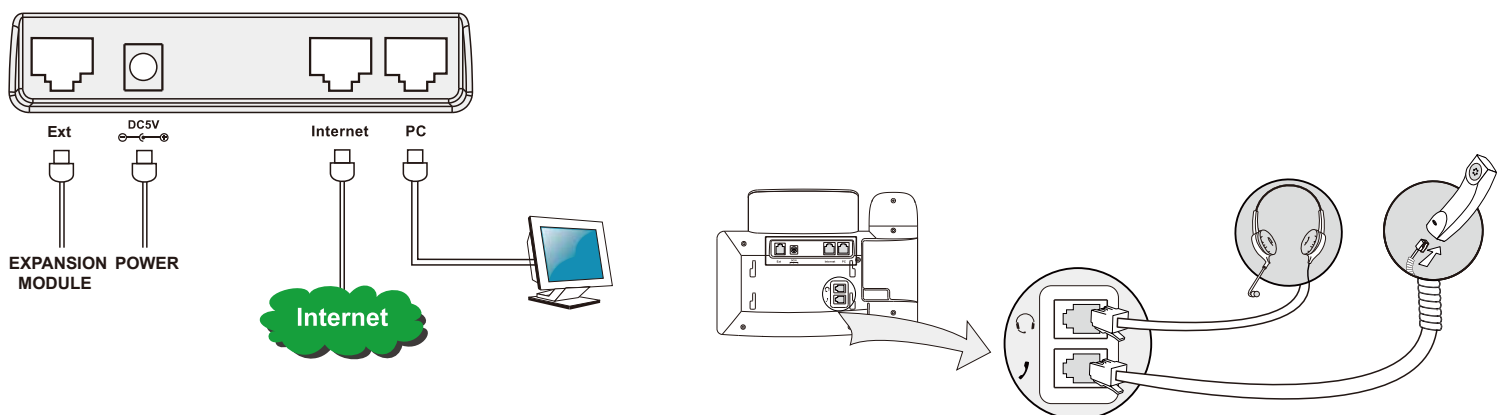
Physical Features

TI TITAN chipset
132x64 graphic LCD
45 keys including 13 programmable keys
1xRJ9 (4P4C) handset port
1xRJ9 (4P4C) headset port
2xRJ45 10/100M Ethernet ports
1XRJ12 (6P6C) EXT port
Power adapter: AC 100~240V input and DC 5V/1.2A output
Power over Ethernet (IEEE 802.3af)
Power consumption: 1.6-2.6W
Net weight: 1.05KG
Dimension: 273x204x42MM
Operating humidity: 10~95%
Storage temperature: up to 60°C

Package Features

Qty/CTN: 5 PCS
N.W/CTN: 8.065KG
G.W/CTN: 8.915KG
Measurement: 0.062CMB
Carton Meas: 580x320x300MM

Certifications



Executive IP Phone with 6 Lines & HD Voice

- TI TITAN chipset and TI voice engine
- 6 VoIP accounts, 320x160 graphic LCD
- HD Voice: HD Codec, HD speaker, HD handset
- IPV6, BLF/BLA, XML Browser, Hot-desking, OpenVPN
- 2xLAN, PoE, Headset, Expansion module, EHS



Yealink SIP-T28P represents the next generation VoIP phone which designed for the business user who needs rich telephony features, friendly UI and super voice quality. It is equipped with the TI TITAN chipset, offers high definition voice quality through TI voice engine, HD handset, HD speaker and HD codec (G.722). By the large, high-resolution graphical display, and together with all the 48 keys, SIP-T28P offers an excellent user experience to configure, make calls, express XML browser, etc. Moreover, to avoid problems with unwanted violations of your audio data, Yealink SIP-T28P supports the security standards TLS, SRTP, HTTPS, 802.1x, Open VPN and AES encryption which are necessary to protect against electronic eavesdropping and data theft.

Phone Features

6 VoIP accounts, Hotline, Emergency call
Call hold, Call waiting, Call forward, Call return
Call transfer (blind/semi-attended/attended)
Caller ID display, Redial, Mute, DND
Auto-answer, 3-way conferencing
Speed dial, SMS, Voicemail
Message Waiting Indication (MWI) LED
Tone scheme, Volume control
Direct IP call without SIP proxy
Ring tone selection/import/delete
Phonebook (300 entries), Black list
Call history: dialed/received/missed/forwarded
Menu-driven user interface
Localized language and input method
Soft keys programmable
Supports up to 6 expansion modules(EXP38 and EXP39)
Supports Wireless Headset Adapter(EHS36)

Advanced Features

XML phonebook search/import
LDAP phonebook
XML Browser, Hot-desking, PUSH XML
Action URL & Active URI
Headset (Call center mode)
Multicast IP Paging

IP PBX System Integration

Busy lamp field (BLF), BLF list
Bridged line appearance (BLA)
DND&Forward synchronization
Intercom, Paging, Music on hold
Call park, Call pickup
Call recording, Call completion
Group listening, Group pickup
Anonymous call, Anonymous call rejection
Network conference
Distinctive ringtone
Dial Plan, Dial-now

Codecs and Voice Features

Wideband codec: G.722
Narrowband codec: G.711 μ /A, G.723.1
G.726, G.729AB
VAD, CNG, AEC, PLC, AJB, AGC
Full-duplex speakerphone with AEC

Network Features

SIP v1 (RFC2543), v2 (RFC3261)
IPv6
DNS SRV (RFC3263)
Redundant server support
NAT Traversal: STUN mode
DTMF: In-Band, RFC2833, SIP Info
Proxy mode and peer-to-peer SIP link mode
IP Assignment: Static/DHCP/PPPoE
Bridge/router mode for PC port
TFTP/DHCP/PPPoE client
Telnet/HTTP/HTTPS server
DNS client, NAT/DHCP server
Logout

Management

Auto-provision via FTP/TFTP/HTTP/HTTPS
Auto-provision with PnP
SNMP V1/2, TR069 optional
Configuration: browser/phone/auto-provision
Factory configuration customized
Trace package and system log export

Security

Open VPN, 802.1x, VLAN QoS (802.1pq), LLDP
Transport Layer Security (TLS)
HTTPS (server/client), SRTP (RFC3711)
Digest authentication using MD5/MD5-sess
Secure configuration file via AES encryption
Phone lock for personal privacy protection
Admin/VAR/User 3-level configuration mode

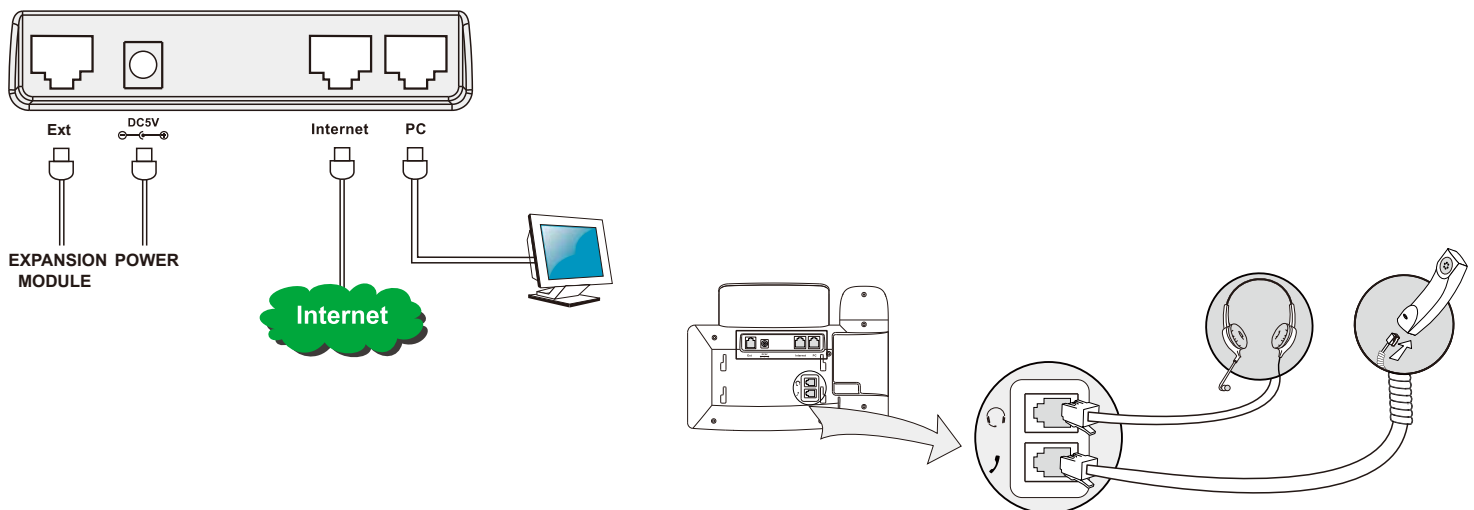
Physical Features

TI TITAN chipset
320x160 graphic LCD with 4-level grayscales
48 keys including 16 programmable keys
1xRJ9 (4P4C) handset port
1xRJ9 (4P4C) headset port
2xRJ45 10/100M Ethernet ports
1XRJ12 (6P6C) EXT port
Power adapter: AC 100~240V input and DC 5V/1.2A output
Power over Ethernet (IEEE 802.3af)
Power consumption: 1.6-2.6W
Net weight: 1.05KG
Dimension: 273x204x42MM
Operating humidity: 10~95%
Storage temperature: up to 60°C

Package Features

Qty/CTN: 5 PCS
N.W/CTN: 8.065KG
G.W/CTN: 8.915KG
Measurement: 0.062CMB
Carton Meas: 580x320x300MM

Certifications



Future-proof IP Phone that combines GigE

- Dual-port Gigabit Ethernet, Power over Ethernet
- 4,3" TFT-LCD, 16.7M colors, Intuitive user interface
- TI Aries chipset and TI voice engine
- HD Voice: HD Codec, HD speaker, HD handset
- 6 VoIP accounts, BLF/BLA, IPv6, Open VPN
- Headset, Support Wireless Headset, LCD Expansion module



SIP-T38G IP Phone is Yealink latest innovation for managers with demanding collaborative communication needs. For managers, it is a future-proofing network investment which supports seamless migration to GigE-based network infrastructure. Dual-port Gigabit Ethernet is designed for flexible deployment options and lower cabling expenses. With its high-resolution TFT color display, The T38G IP Phone offers a brilliant depiction of caller's information. And the user interface was designed for clarity and intuitive operation. Moreover, Equipped with the TI Aries chipset, HD handset, HD speaker and HD codec (G.722), The T38G IP Phone offers an unrivaled, lifelike audio experience.

Additionally, T38G IP Phone includes a host of telephony features to increase efficiency.

Phone Features

6 VoIP accounts, Hotline, Emergency call
Call hold, Call waiting, Call forward, Call return
Call transfer (blind/semi-attended/attended)
Caller ID display, Redial, Mute, DND
Auto-answer, 3-way conferencing
Speed dial, SMS, Voicemail
Message Waiting Indication (MWI) LED
Tone scheme, Volume control
Direct IP call without SIP proxy
Ring tone selection/import/delete
Phonebook (1000 entries), Black list
Call history: dialed/received/missed/forwarded
Menu-driven user interface
Localized language and input method
Soft keys programmable
Supports up to 6 expansion modules (EXP39 and EXP38)
Supports Wireless Headset Adapter(EHS36)

Advanced Features

XML phonebook search/import
LDAP phonebook
XML Idle Screen
Action URL & Active URI
Wallpaper, Screensaver
Color Picture Caller-ID
Theme, Screen Sleep mode

IP PBX System Integration

Busy lamp field (BLF), BLF list, (BLA)
DND & Forward synchronization
Intercom, Paging, Music on hold
Call park, Call pickup
Call recording, Call completion
Group listening, Group pickup
Anonymous call, Anonymous call rejection
Network conference
Distinctive ringtone
Dial Plan, Dial-now

Codecs and Voice Features

Wideband codec: G.722
Narrowband codec: G.711 μ /A, G.723.1
G.726, G.729AB
VAD, CNG, AEC, PLC, AJB, AGC
Full-duplex speakerphone with AEC

Network Features

SIP v1 (RFC2543), v2 (RFC3261)
IPv6
DNS SRV (RFC3263)
Redundant server support
NAT Traversal: STUN mode
DTMF: In-Band, RFC2833, SIP Info
Proxy mode and peer-to-peer SIP link mode
IP Assignment: Static/DHCP/PPPoE
Bridge/router mode for PC port
TFTP/DHCP/PPPoE client
Telnet/HTTP/HTTPS server
DNS client, NAT/DHCP server
Logout

Management

Auto-provision via FTP/TFTP/HTTP/HTTPS
Auto-provision with PnP
SNMP V1/2 optional, TR069 optional
Configuration: browser/phone/auto-provision
Factory configuration customized
Trace package and system log export

Security

Open VPN, 802.1x, VLAN QoS (802.1pq), LLDP
Transport Layer Security (TLS)
HTTPS (server/client), SRTP (RFC3711)
Digest authentication using MD5/MD5-sess
Secure configuration file via AES encryption
Phone lock for personal privacy protection
Admin/VAR/User 3-level configuration mode

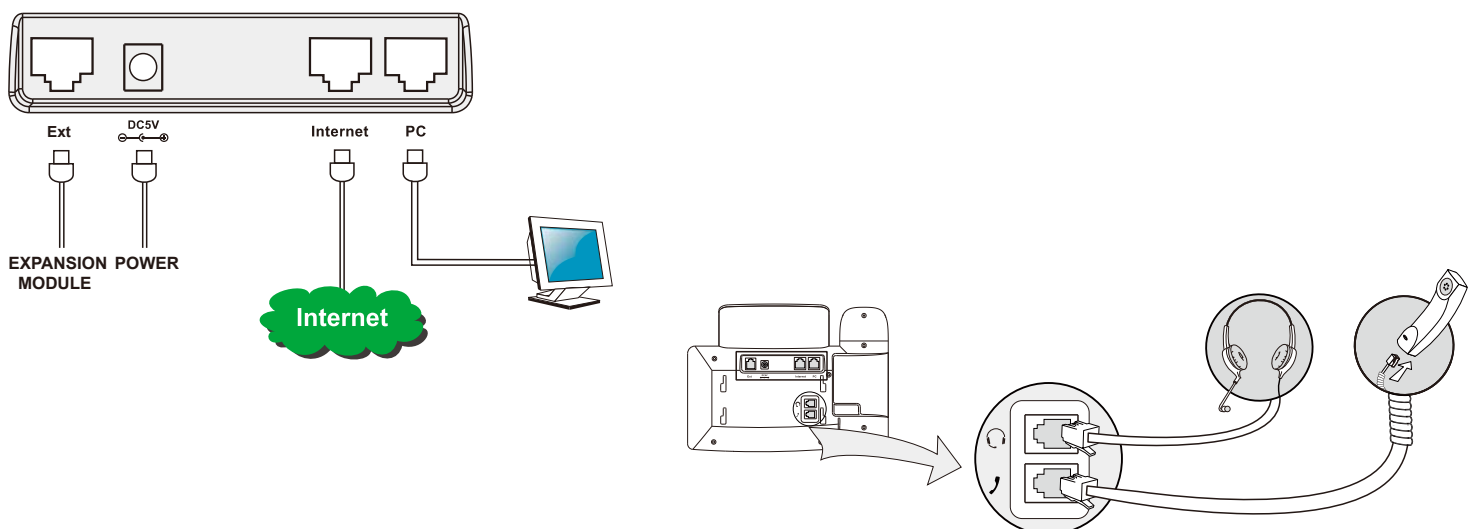
Physical Features

2xRJ45 10/100/1000Mbps Ethernet ports
TI Aries chipset
4.3" TFT-LCD, 480 x 272 pixel, 16.7M colors
48 keys including 16 programmable keys
1xRJ9 (4P4C) handset port
1xRJ9 (4P4C) headset port
1XRJ12 (6P6C) EXT port
Power adapter: AC 100~240V input and DC 5V/2A output
Power over Ethernet (IEEE 802.3af)
Power consumption approx.: 4.6W
Net weight: 1.05KG
Dimension: 273x204x42MM
Operating humidity: 10~95%
Storage temperature: up to 60°C

Package Features

Qty/CTN: 5 PCS
N.W/CTN: 8.065KG
G.W/CTN: 8.915KG
Measurement: 0.062CMB
Carton Meas: 580*295*250MM

Certifications



Powerful Features in the Advanced, Executive Level IP Video Phone

- > 7" Touch Screen
- > HD Voice
- > 3-way video Conferencing
- > Total Directory solution
- > Open API



The Yealink VP530, innovation of the advanced, executive level IP Video Phone. Unified with audio, video, applications, the VP530 is a powerful business video phone. Its large display and ease of use make the VP530 an ideal all-in-one tool for today's busy executives and managers, whether they are in office, soho, healthcare, etc. With its excellent user experience and rich business features, the VP530 creates an immersive, face-to-face experience over the network, empowering users collaborate with each other like never before.

Superb user experience

Featuring a 7" color high resolution touch screen, the VP530 offers a graphical business grade user interfaces, touchable Dsskeys/softkeys and virtual keypad. Intergating wideband software and hardware, the VP530 delivers life-like communication with HD voice, full-duplex speaker and handset in considerably fast response of the icons and interfaces switching. Furthermore, the VP530 offers 3-way video conferencing, through which users can demonstrate and deliver the most what they want other sides to get, without the expensive MCU.

Maximize productivity

The VP530 offers a total directory solution on the intuitive and icon-driven interfaces. It enables to search any entries on the phone or overall searching on the dialing interface with intelligent searching way. For example, if you want to get "Mike", only "6453" need to be entered. Then there is a "Mike" entry. When you want to get someone you don't know his name or number, you could still reach it through remote phonebook which is displayed by departments within simply 3-touch. The directory solution will give you unprecedented experience.

Simplified deployment

The VP530 is easy to deploy and manage. Its enterprise-grade, web-based, intuitive configuration method gives administrators the ability to easily provision and maintain a large number of phones throughout the entire enterprise. From initial deployment and configuration to future enhancements and upgrades, the VP530 is designed for this. Integrated Power-over-Ethernet allows easy deployment with centralized powering and backup.

Highly customizable and expandable

The VP530 features an Open API, XML Browser, Push XML, Action URI/URL, that enables the third-party developers to integrate the VP530 with business applications, such as Enterprise Resource Planning (ERP), Customer Relationship Management (CRM), and specialized applications for certain industries, such as healthcare, hotel, education, with customized language locally.

Benefits

- > Superb user experience with intelligent features integrated in a ease of use phone
- > Maximize productivity for managers, executives
- > Efficiently cooperate with each other by video call, video conferencing with lower travel expenses and faster decision making
- > Easy to deploy and simple to administer, upgrade, maintain
- > Reduced carbon footprint with energy-saving PoE for a green world
- > Highly customizable and expandable

Video Features

- > Video codec: H.264 and H.263
- > Image codec: JPEG, PNG, BMP
- > Video call format: CIF/QCIF
- > Bandwidth selection: 128kbps-1Mbps
- > Frame rate selection: 10-30fps
- > Adaptive bandwidth adjustment
- > I-frame adjustable
- > Picture-in-Picture (PIP)
- > Full screen for remote side
- > Video control of local side
- > Door phone application

Audio Features

- > HD voice: HD codec, HD handset, HD speaker
- > Wideband codec: G.722
- > Narrowband codec: G.711(A/μ), G.723.1, G.729AB
- > DTMF: In-band, Out-of-band(RFC 2833) and SIP INFO
- > Full-duplex hands-free speakerphone with AEC
- > Voice activity detection
- > Comfort noise generation
- > Adaptive jitter buffers
- > Packet loss concealment

IP-PBX and BroadSoft Features

- > Busy lamp field (BLF), BLF list
- > Bridged line appearance(BLA)/SCA
- > Message waiting indicator (MWI)
- > Intercom, paging, Music on hold
- > Call park, Call pickup
- > Call completion
- > Anonymous call, Anonymous call rejection
- > DND & forward synchronization
- > Dial Plan, Dial-now

Directory

- > Local phonebook up to 1000 entries
- > Phonebook with contact picture
- > Group manager, Favorites, Black list
- > XML/LDAP remote phonebook
- > Intelligent search method

Phone Features

- > 4 VoIP accounts, Video/Voice call
- > 18 one-touch soft DSS keys
- > One-touch speed dial, redial
- > Call forward, Call waiting, Call transfer, Call hold
- > Call return, Group listening, Group pick up
- > Mute, Auto answer
- > DND, Caller ID display, Call log
- > Voice mail, MWI
- > 3-way video conferencing
- > Direct IP call without SIP proxy
- > New message and missed call notification
- > Volume control, Ring tone selection
- > Wall paper
- > Set date time manually or automatically
- > National language selection
- > Backlight time selection
- > Icon-driven menu

Open API

- > XML Browser
- > Push XML
- > Action URI/URL

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/PPPoE
- > 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, and DSCP
- > SRTP for voice and video
- > Transport Layer Security (TLS)
- > HTTPS certificate manager
- > AES encryption for configuration file
- > Digest authentication using MD5/MD5-sess
- > OpenVPN, IEEE802.1X

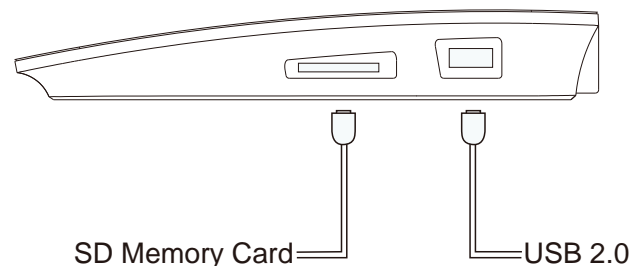
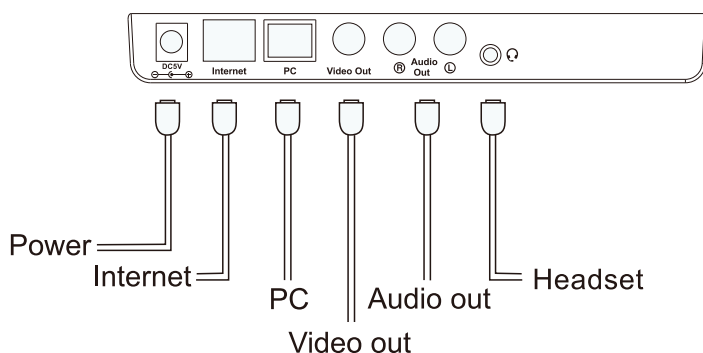
Management

- > 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
- > Recovery mode
- > Reset to factory, Reboot
- > Package tracing export, System log

Physical Features

- > TI DaVinci dual-core chipset, resistive touch screen
- > 7" digital TFT-LCD with 800x480 pixels resolution
- > Rotatable CMOS sensor camera with 2M pixels
- > 128MB flash and 256MB DDR2 memory
- > 27 keys including 4 soft keys
- > 6 feature keys: Mute/Camera/Phonebook/Transfer/Redial/Hands-free
- > 2xLEDs for power and status indication
- > 2xRJ45 Ethernet 10/100M ports
- > 2.5mm headset port
- > A/V out (pending)
- > SD Memory Card, USB 2.0 (pending)
- > Power adapter: AC 100-240V input and DC 5V/3A output
- > Power over Ethernet (PoE) optional: IEEE 802.3af, Class 0
- > Power consumption: 4-10W
- > Net weight: 1.2Kg
- > Dimension: 286x89x45mm
- > Operating humidity: 10-95%
- > Storage temperature: up to 60°C

Certifications



Learn More

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Addr: 4th-5th Floor, South Building, No.63 Wanghai Road, 2nd Software Park, Xiamen, China

Web: www.yealink.com

Tel: +86-592-5702000

Email: sales@yealink.com