PHONE CATALOG

About Grandstream

Grandstream has been a global provider of cloud-based unified communications services since 2002. Delivering award-winning IP voice/video telephony to businesses of all sizes, Grandstream offers a large portfolio of over 50 products, including basic, advanced and executive IP phones as well as cordless handsets, conferencing systems, analog adapters and other devices. Recognized worldwide for their quality, reliability and innovation, their SIP-based products and solutions help to lower communication costs, increase security and allow businesses to be more productive.

Grandstream DP720

Grandstream DP750

Grandstream DP760

Grandstream GAC2500

Grandstream GXP1630

Grandstream GXP2135

Grandstream GXP2140

Grandstream GXP2170

Grandstream GXP2200 EXT

Grandstream HT801

About Yealink

Grandstream has been a global provider of cloud-based unified communications services since 2002. Delivering award-winning IP voice/video telephony to businesses of all sizes, Grandstream offers a large portfolio of over 50 products, including basic, advanced and executive IP phones as well as cordless handsets, conferencing systems, analog adapters and other devices. Recognized worldwide for their quality, reliability and innovation, their SIP-based products and solutions help to lower communication costs, increase security and allow businesses to be more productive.

Yealink CP920

Yealink EXP40

Yealink SIP-T21(P) E2

Yealink SIP-T21P

Yealink SIP-T27G

Yealink SIP-T29G

Yealink SIP-T46S

Yealink W52P

Yealink W56P

Yealink W60P

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GRANDSTREAM

DP720

DECT Cordless HD Handset for Mobility

The DP720 is a DECT cordless VoIP phone that allows users to mobilize their VoIP network throughout any business, warehouse, retail store and residential environment. It is supported by Grandstream's DP750 DECT VoIP base station and delivers a combination of mobility and top-notch telephony performance. Up to five DP720 handsets are supported on each DP750 while each DP720 supports a range of up to 300 meters outdoors and 50 meters indoors from the base station. The DP720 touts a suite of top-notch telephony features including support for up to 10 SIP accounts per handse t, full HD audio, a 3.5mm headset jack, multi-language support, a speakerphone and more. When paired with Grandstream's DP750 DECT Base Station, the DP720 offers a powerful DECT VoIP handset that allo ws any business or residential user to create a cordless VoIP solution.

Features

10 LINES	Up to 10 SIP accounts per system; up to 10 lines per handset		DECT authentication & encryption technology to protect calls & account
3 WAY	3-way audio conferencing for easy conference calls	SIP	DP750 is fully compliant with the SIP/ DECT standard
	Automated provisioning options include TR-069 and XML config files	zero CONFIG	Use with Grandstream's UCM series of IP PBXs for Zero Config provisioning
HD	HD audio to maximize audio quality and clarity; full duplex speakerphone		Supports advanced telephony features, including call transfer, call forward, call-waiting, do not disturb, message waiting indication, multi-language prompts, flexible dial plan and more

Specifications

Air Interface
Telephony standards: DECT
Frequency bands:
1880 – 1900 MHz (Europe), 1920 – 1930 MHz (US),
1910 – 1920 MHz (Brazil), 1786 – 1792 MHz (Korea),
1893 – 1906 MHz (Japan), 1880 – 1895 MHz (Taiwan)
Number of Channels: 10 (Europe), 5 (US, Brazil or Japan), 3 (Korea), 8 (Taiwan)
Range: up to 300 meters outdoors and 50 meters indoors

Peripherals
1.8 inch (128x160) color TFT LCD
23 keys including 2 soft keys, 5 navigation/ menu keys, 4 dedicated function keys for SEND, POWER/END, SPEAKERPHONE, MUTE
3-color MWI LED
3.5mm headset jack
Removable belt clip
Micro-USB port for alternative charging and non-battery operation

Protocol/Standards
Hearing Aid Compatibility (HAC) compliant

Voice Codecs
G.722 codec for HD audio and G.726 codec for narrow band audio (G.711µ/a-law, G.723.1, G.729A/B, iLBC and OPUS are supported via companion DECT base station DP750), AEC, AGC, Ambient noise reduction

Telephony Features
Hold, transfer, forward, 3-way conference, call park, call pickup, downloadable phonebook, call waiting, call log, auto answer, click-to-dial, flexible dial plan, music on hold

Sample Applications
Weather, Currency (Pending)

HD Audio
Yes, both on Handset and Speakerphone

Security
DECT authentication & encryption

Multi-language
Chinese Simple, Chinese Tradition, Czech, Danish, Dutch, English, Estonian, Finnish, French, German, Hebrew, Hungarian, Japanese, Korean, Norwegian, Polish, Portuguese, Romanian, Spanish, Turkish

Upgrade/ Provisioning
Software Upgrade Over-The-Air (SUOTA), handset provisioning Over-The-Air

Multi-line Access
Each handset may access up to ten (10) lines

Power & Green Energy Efficiency
Universal Power Supply Input AC 100-240V 50/60Hz; Output 5VDC 1A; Micro-USB connection; Rechargeable 800mAh Ni-MH Low Self-Discharge (LSD) AAA batteries (250 hours of standby time and 20 hours of talk time)

Package Content
Handset unit, universal power supply, charger cradle, belt clip, 2 batteries, Quick Start Guide

Dimensions (H x W x D)
Handset: 155 x 50 x 26 mm, charger cradle: 35 x 63.5 x 54 mm

Weight
Handset: 138g; Charger Cradle: 71g; Universal Power Supply: 50g; Package: 360g

Temperature and Humidity
Operation: -10° to 50°C (14 to 122°F); Charging: 0 to 45°C (32 to 113°F)
Storage: -20° to 60°C (-4 to 140°F); Humidity: 10% to 90% non-condensing

Compliance
FCC: Part 15D, 47 CFR 2.1093 & IEEE1528-2013, Part 68, Part 15B
CE: EN60950, EN301489-1-6, EN301406, EN50360; EN62209-1
RCM: AS/NZS60950, AS/ACIF S004
ANATEL: #2288-16-9452

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GRANDSTREAM

DP750

Long-range DECT VoIP Base Station

The DP750 is a powerful DECT VoIP base station that pairs with up to 5 of Grandstream's DP720 DECT handsets to offer mobility to business and residential users. It supports a range of 300 meters outdoors and 50 meters indoors to give users the freedom to move around their work or home space, delivering efficient flexibility. This DECT VoIP base station supports up to 10 SIP accounts and 5 concurrent calls while also offering 3-way voice conferencing, full HD audio and integrated PoE. A shared SIP account on all handsets will add seamless unified features that gives users the ability to answer all calls regardless of location in real-time. The DP750 supports a variety of auto- provisioning methods and TLS/SRTP/ HTTPS encryption security. When paired with Grandstream's DP720 , the DP750 offers a powerful DECT VoIP base station that allows any business or residential user to create a cordless VoIP solution.

Features

10 LINES	Up to 10 SIP accounts per system; up to 10 lines per handset		DECT authentication & encryption technology to protect calls & account
3 WAY	3-way audio conferencing for easy conference calls	SIP	DP750 is fully compliant with the SIP/DECT standard
	Automated provisioning options include TR-069 and XML config files	zero CONFIG	Use with Grandstream's UCM series of IP PBXs for Zero Config provisioning
HD	HD audio to maximize audio quality and clarity; full duplex speakerphone		Supports advanced telephony features, including call transfer, call forward, call-waiting, do not disturb, message waiting indication, multi-language prompts, flexible dial plan and more

Specifications

Air Interface
Telephony standards: DECT
Frequency bands:
1880 – 1900 MHz (Europe), 1920 – 1930 MHz (US), 1910 – 1920 MHz (Brazil), 1786 – 1792 MHz (Korea), 1893 – 1906 MHz (Japan), 1880 – 1895 MHz (Taiwan)
Number of Channels: 10 (Europe), 5 (US, Brazil or Japan), 3 (Korea), 8 (Taiwan)
Range: up to 300 meters outdoors and 50 meters indoors

Peripherals
5 LED indicators: Power, Network, Register, Call, DECT
Reset button, Pairing/Paging button
One 10/100 Mbps auto-sensing Ethernet port with integrated PoE

Protocol/Standards
SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP/RARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TFTP, NTP, STUN, SIMPLE, LLDP-MED, LDAP, TR-069, 802.1x, TLS, SRTP, IPv6 (Pending)

Voice Codecs
G.711µ/a-law, G.723.1, G.729A/B, G.726-32, iLBC, G.722, OPUS, G.722.2/AMR-WB (special order), in-band and out-of-band DTMF (in audio, RFC2833, SIP INFO), VAD, CNG, PLC, AJB

Telephony Features
Hold, transfer, forward, 3-way conference, downloadable phonebook (XML, LDAP, up to 3000 entries), call waiting, call log (up to 300 records), auto answer, flexible dial plan, music on hold, server redundancy and fail-over

Sample Applications
Weather, Currency (Pending)

QoS
Layer 2 QoS (802.1Q, 802.1P) and Layer 3 QoS (ToS, DiffServ, MPLS)

Security
User and administrator level access control, MD5 and MD5-sess based authentication, 256-bit AES encrypted configuration file, TLS, SRTP, HTTPS, 802.1x media access control, DECT authentication & encryption

Multi-language
Chinese Simple, Chinese Tradition, Czech, Danish, Dutch, English, Estonian, Finnish, French, German, Hebrew, Hungarian, Japanese, Korean, Norwegian, Portuguese, Romanian, Spanish, Swedish, Turkish

Upgrade/ Provisioning
Firmware upgrade via TFTP/HTTP/HTTPS, mass provisioning using TR-069 or AES encrypted XML configuration file

Multiple SIP Accounts
Up to ten (10) distinct SIP accounts per system
Each handset may map to any SIP account(s)
Each SIP account may map to any handset(s)

Ring Group
Flexible options when multiple handsets share the same SIP account
Circular Mode: all phones ring sequentially from the phone next to the one that answered last
Linear Mode: all phones ring sequentially in the predesignated order
Parallel Mode: all phones ring concurrently and after one phone answers, the remaining available phones can make new calls
Shared Mode: all phones ring concurrently and always share the same line similar to analog phones

Power & Green Energy Efficiency
Universal Power Supply Input AC 100-240V 50/60Hz; Output 5VDC 1A; Micro-USB connection; PoE: IEEE802.3af Class 1, 0.44W–3.84W

Package Content
Base Unit, Universal Power Supply; Ethernet cable; Quick Start Guide, GPL statement

Dimensions (H x W x D)
28.5 x 130 x 90 mm

Weight
Base unit: 143g; Universal Power Supply: 50g; Package: 360g

Temperature and Humidity
Operation: -10 to 50°C (14 to 122°F); Storage: -20° to 60°C (-4 to 140°F)
Humidity: 10% to 90% non-condensing

Compliance
FCC: Part 15D, Part 15B
CE: EN60950, EN301489-1-6, EN301406
RCM: AS/NZS60950
ANATEL: #2288-16-9452

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GRANDSTREAM

DP760

Long-Range Wideband DECT Repeater

The DP760 is a powerful wideband DECT repeater (wireless relay station) that auto associates to Grandstream's DP750 DECT base station offering extended mobility to business and residential users. The DP760 extends an additional range of 300 meters outdoors and 50 meters indoors to give users the freedom to move around their home or work space. This Wideb and DECT Repeater relays up to 2 concurrent HD calls. The Ethernet connection provides PoE for convenient installation and a variety of remote features including provisioning, status monitoring and repeater firmware upgrades. When paired with Grandstream's DP750 DECT VoIP base station and DP720 handsets, the DP760 offers a powerful extended DECT solution for users looking to add coverage to their VoIP DECT system.

2 LINES	Up to 2 concurrent HD calls		Extends an additional range of 300 meters outdoors and 50 meters indoors for added mobility
	Automatic or manual association to DP750, base station for easy use		Supports Plug-n-Play features like auto association, auto region detection and seamless call handover
	Automated provisioning options include XML config files	PoE	Ethernet connection provides PoE features

Specifications

Air Interface

Telephony standards:
DECT EN 301 406:2001
DECT GAP TBR22 EN 300 444:2001
DECT WRS EN 300 700, CAT-iq TS 102 527
Frequency bands:
1880 – 1900 MHz (Europe), 1920 – 1930 MHz (US), 1910 – 1920 MHz (Brazil)
Number of Channels: 10 (Europe), 5 (US, Brazil)
Range: up to 300 meters outdoors and 50 meters indoors

Peripherals

5 LED indicators: Power, Network, Association, Activity, DECT Signal Strength, Reset button, Dissociation button, One 10/100 Mbps auto-sensing Ethernet port with integrated PoE

Protocol/Standards

TCP/IP/UDP, HTTP/HTTPS, ARP/RARP, ICMP, DNS, DHCP, PPPoE, SSH, TFTP, NTP, LLDP-MED, UPnP

Voice Codecs

G.722 codec for HD audio and G.726 codec for narrow band audio

Telephony Features

Plug-n-Play, auto association, auto region detection and seamless call handover

Security

User and administrator level access control, MD5 and MD5-sess based authentication, 256-bit AES encrypted configuration file, HTTPS, 802.1x media access control

Multi-language

Arabic, Chinese Simple, Chinese Tradition, Czech, Dutch, English, French, German, Hebrew, Italian, Japanese, Korean, Polish, Portuguese, Russian, Serbian, Slovakian, Spanish, Swedish, Turkish

Upgrade/ Provisioning

Firmware upgrade via TFTP/HTTP/HTTPS, mass provisioning using AES encrypted XML configuration file

Association

Up to 5 repeaters in star
Relays up to 2 concurrent HD calls
Automatic or manual association to base station

Power & Green Energy Efficiency

Universal Power Supply Input AC 100-240V 50/60Hz; Output 5VDC 1A; Micro-USB connection; PoE: IEEE802.3af Class 1, 0.44W–3.84W

Package Content

Repeater Unit, Universal Power Supply; Ethernet cable; Quick Start Guide,
GPL statement

Dimensions (H x W x D)

28.5 x 130 x 90 mm

Weight

Repeater Unit: 143g; Universal Power Supply: 50g; Package: 360g

Temperature and Humidity

Operation: -10° to 50°C (14° to 122°F); Storage: -20° to 60°C (-4° to 140°F)
Humidity: 10% to 90% non-condensing

Compliance

FCC: Part 15B, Part 15D, MPE
CE: EN 60950-1, EN 301 489-1, EN 301 489-6, EN 301 406, EN 50385, EN 55032, EN 55024, EN 61000-3-2, EN 61000-3-3
RCM: AS/NZS 60950.1
ANATEL

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GRANDSTREAM

GAC2500

Android Enterprise Conference Phone

The GAC2500 is an Android-based Business Conference Phone that will redefine the choices, flexibility and mobility available in any workplace. It supports up to 6 lines and 6 SIP accounts while also offering full access to the Google Play Store in order to hold conferences through Skype, Google Hangouts and more. This Business Conference Phone offers a variety of mobility options by supporting Bluetooth for audio pairing and data syncing with mobile devices, as well as WiFi for wireless calling/conferencing in any location. A 7-way conference bridge allows the GAC2500 to easily create and hold a conference at any time and the 4.3” touch screen with familiar Android interface offers easy use. This Android Enterprise Conference Phone also offers daisy chain support. By combining all of these features, the GAC2500 offers a foundation that gives any business the choice and flexibility to customize their conference phone based on their communication needs.

Features

Runs Android 4.4 and offers full access to the Google Play Store; Create custom Android apps	Auto-sensing 10/100/1000mbps network port
Built-in Bluetooth for syncing headsets and mobile devices	Built-in PoE+ to power the device and give it a network connection
WiFi support offers mobility	Supports 6 SIP accounts and 7-way voice conferencing
HD audio to maximize audio quality	4.3 inch (800x480) capacitive touch screen
TLS and SRTP security encryption technology to protect calls and accounts	

Specifications

Protocols/Standards
SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TFTP, NTP, STUN, SIMPLE, LLDP, LDAP, TR-069, 802.1x, TLS, SRTP, IPV6 (pending), OpenVPN (pending)

Network Interface
Auto-sensing Gigabit Ethernet port with integrated PoE+ (IEEE 802.3at Class4)

Graphic Display
4.3" IPS LCD with 800x480 resolution

Mic
3 cardioid microphones; 12 ft. pickup distance

Speakerphone
Frequency: 220-18,000 Hz Volume: Up to 86 dB at 0.5 meter Audio full duplex

Bluetooth
Yes, integrated. Bluetooth 4.0

Wi-Fi
Yes, integrated. 802.11 b/g/n

Auxiliary Ports
3.5mm audio port, USB Micro-B, RJ48 daisy chain port

Voice Codecs
Support for G.711µ/a, G.722, G.726, iLBC, Opus, G.722.1 and G.722.1c (pending), in-band and out-of-band DTMF (In audio, RFC2833, SIP INFO), G.729A/B, VAD, CNG, AEC, PLC, AJB, AGC

Telephony Features
6 SIP accounts, hold, transfer, forward, 7-way conference, call park, call pickup, downloadable phonebook (XML, LDAP, up to 2000 items), call waiting, call log (up to 2000 records), XML customization of screen, auto answer, click-to-dial, flexible dial plan, hot desking, personalized music ringtones and music on hold, server redundancy and fail-over

Sample Applications
Skype, Google Hangouts, Skype for Business (Lync), Web browser, Adobe Flash, Facebook, Twitter, YouTube, Google calendar, mobile phone data import/export via Bluetooth, etc. API/SDK available for advanced custom application development

HD Audio
Yes, speakerphone with support for wideband audio

QoS
Layer 2 QoS (802.1Q, 802.1p) and Layer 3 (ToS, DiffServ, MPLS) QoS.

Security
User and administrator level passwords, MD5 and MD5-sess based authentication,
256-bit AES encrypted configuration file, TLS, SRTP, HTTPS, 802.1x media access
control

Multi-language
English, German, Italian, French, Spanish, Portuguese, Russian, Croatian, Chinese, Korean, Japanese, and more

Upgrade/Provisioning
Firmware upgrade via TFTP / HTTP / HTTPS or local HTTP upload, mass provisioning using TR-069 or AES encrypted XML configuration file

Power & Green Energy Efficiency
Universal Power Supply:Input:100-240VAC 50-60Hz; Output:12VDC,2A (24W)

Temperature and Humidity
Operation: 0°C to 40°C
Storage: -10°C to 60°C
Humidity: 10% to 90% Non-condensing

Package Content
GAC2500 phone, universal power supply, network cable, USB cable, RJ48 cascade cable, Quick Installation Guide

Compliance
FCC: Part 15 (CFR 47) Class B; UL 60950 (power adapter)
CE: EN55022 Class B, EN55024, EN61000-3-2, EN61000-3-3, EN60950-1, EN62479, RoHS
RCM: AS/ACIF S004; AS/NZS CISPR22/24; AS/NZS 60950; AS/NZS 4268

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GRANDSTREAM

GXP1630

A Small Business Gigabit IP Phone

The GXP1630 is a powerful IP phone for small-to-medium businesses (SMBs). This Linux-based model includes 3 lines, 3 XML programmable soft keys, 8 BLF keys and 4-way conferencing. A 132x64 (2.98'') backlit LCD screen creates a clear display for easy viewing. Additional features such as dual switched gigabit network ports, HD audio, multi-language support, integrated PoE and call-waiting allow the GXP1630 to be a high quality, versatile and dependable office phone.

Features

3 LINES	3 lines, 3 SIP accounts, 3 call apperanc-es, 3 softkeys	Gigabit	Dual-switched auto-sensing 10/100/1000mbps network ports
	Includes 8 dual-colored BLF/speed dial keys	PoE	Built-in PoE+ to power the device and give it a network connection
HD	HD audio to maximize audio quality and clarity, full-depulex speakerphone	zero CONFIG	Use with Grandstream's UCM series of IP PBXs for Zero Config provisioning
EHS	Electronic Hook Switch (EHS) support for Plantronics headsets		Automated provisioning options include TR-069 and XML config files
	TLS and SRTP security encryption technology to protect calls and accounts	4 WAY	4-way audio conferencing for easy conference calls

Specifications

Protocols/Standards
SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP/RARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TFTP, NTP, STUN, SIMPLE, LLDP-MED, LDAP, TR-069, 802.1x, TLS, SRTP

Network Interfaces
Dual switched auto-sensing 10/100/1000 Mbps Ethernet ports, integrated PoE

Graphic Display
32 x 64 (2.98”) backlit graphical LCD display

Feature Keys
3 line keys with dual-color LED and 3 SIP accounts, 3 XML programmable context sensitive soft keys, 5 (navigation, menu) keys, 8 BLF keys, 13 dedicated function keys for MUTE, HEADSET, TRANSFER, CONFERENCE, SEND and REDIAL, SPEAKER-PHONE, VOLUME, PHONEBOOK, MESSAGE, HOLD, PAGE/INTERCOM, RECORD, HOME

Voice Codecs
Support for G.711µ/a, G.722 (wide-band), G.723, G.726-32, G.729 A/B, iLBC, in-band and out-of-band DTMF (In audio, RFC2833, SIP INFO), VAD, CNG, AEC, PLC, AJB, AGC

Telephony Features
Hold, transfer, forward (unconditional/no-answer/busy), call park/pickup, 4-way conference, shared-call-appearance (SCA) / bridged-line-appearance (BLA), down-loadable phone book (XML, LDAP, up to 1000 items), call waiting, call history (up to 200 records), off-hook auto dial, auto answer, click-to-dial, flexible dial plan, hot desking, personalized music ringtones, server redundancy & fail-over

Headset Jack
RJ9 headset jack (allowing EHS with Plantronics headsets)

HD Audio
Yes, HD handset and speakerphone with support for wideband audio

Base Stand
Yes, 2 angled positions available, wall mountable

Wall Mountable
Yes

QoS
Layer 2 QoS (802.1Q, 802.1P) and Layer 3 QoS (ToS, DiffServ, MPLS)

Security
User and administrator level access control, MD5 and MD5-sess based authenti-cation, 256-bit AES encrypted configuration file, TLS, SRTP, HTTPS, 802.1x media access control

Multi-language
English, German, Italian, French, Spanish, Portuguese, Russian, Croatian, simplified and traditional Chinese, Korean, Japanese and more

Upgrade/Provisioning
Firmware upgrade via TFTP / HTTP / HTTPS, mass provisioning using TR-069 or AES encrypted XML configuration file

Power and Green Energy Efficiency
Universal Power Supply Input 100-240VAC 50-60Hz; Output +5VDC, 600mA PoE: IEEE802.3 af Class 2, 3.84W-6.49W

Physical
Dimension: 222.5mm (L) x 208.5mm (W) x 76.2mm (H) (with handset) Unit weight: 0.8kg; Package weight: 1.2kg

Temperature and Humidity
Operation: 0°C to 40°C, Storage: -10°C to 60°C , Humidity: 10% to 90% Non-condensing

Package Content
GXP1630 phone, handset with cord, base stand, universal power supply, network cable, Quick Installation Guide, brochure, GPL License

Compliance
FCC: Part 15 (CFR 47) Class B CE : EN55022 Class B, EN55024 Class B; EN61000-3-2, EN61000-3-3, EN60950-1
RCM: AS/ACIF S004; AS/NZS CISPR22/24; AS/NZS 60950; AS/NZS 60950.1

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GRANDSTREAM

GXP2135

A Multi-line High Performance IP Phone

The GXP2135 is an enterprise-grade IP phone that supports Gigabit speeds and up to 32 virtual BLF/speed-dial keys, making it ideal for busy workers. This Enterprise IP Phone features up to 8 lines/line keys and 4 SIP accounts using a 2.8-inch color display LCD and full HD audio. The GXP2135 includes up to 32 digital, on-screen speed dial/BLF keys to help users be more productive and efficient. This Enterprise IP Phone supports the fastest possible connection speeds with dual Gigabit network ports, features integrated PoE and includes built-in Bluetooth for syncing with mobile devices and Bluetooth headsets. The GXP2135 is the perfect choice for business users looking for a powerful and reliable IP phone with advanced functionality

Features

8 LINES	8 lines, 8 dual-color line keys (with 4 SIP accounts), 4 XML programmable context-sensitive soft keys	EHS	Supports EHS compatible Plantronics headsets
Gigabit	Dual switched auto-sensing 10/100/1000Mbps Gigabit network ports		Automated provisioning options include TR-069 and XML config files
	32 digitally programmable & customizable BLF/fastdial keys		TLS and SRTP security encryption technology to protect calls and accounts
	Built-in Bluetooth for syncing headsets and mobile devices for contact books, calendars & call transferring	4 WAY	4-way audio conferencing for easy conference calls
HD	HD audio to maximize audio quality, full-duplex speakerphone	zero CONFIG	Use with Grandstream's UCM series IP PBX appliance for Zero-Config provisioning, 1-touch call recording & more
PoE	Built-in PoE to power the devices and give it a network connection		

Specifications

Protocols/Standards
SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, TELNET, TFTP, NTP, STUN, SIMPLE, LLDP, LDAP, TR-069, 802.1x, TLS, SRTP, IPV6

Network Interfaces
Dual switched auto-sensing 10/100/1000 Mbps Gigabit Ethernet ports with integrated PoE

Graphic Display
2.8-inch (320x240) TFT color LCD

Bluetooth
Yes, integrated

Feature Keys
8 line keys with up to 4 SIP accounts, 4 XML programmable context sensitive soft keys, 5 navigation/menu keys, 11 dedicated function keys for: MESSAGE (with LED indicator), PHONEBOOK, TRANSFER, CONFERENCE, HOLD, HEADSET, MUTE, SEND/REDIAL, SPEAKERPHONE, VOL+, VOL

Voice Codecs
Support for G.729A/B, G.711μ/a-law, G.726, G.722(wide-band), in-band and out-of-band DTMF (in audio, RFC2833, SIP INFO)

Auxiliary Ports
RJ9 headset jack (allowing EHS with Plantronics headsets)

Telephony Features
Hold, transfer, forward, 4-way conference, call park, call pickup, shared-call appearance (SCA)/bridged-line-appearance(BLA), downloadable phonebook (XML, LDAP, up to 2000 items), call waiting, call log (up to 500 records), XML customization of screen, off-hook auto dial, auto answer, click-to-dial, flexible dial plan, hot desking, personalized music ringtones and music on hold, server redundancy and fail-over

HD Audio
Yes, HD handset and speakerphone with support for wideband audio

Base Stand
Yes, 2 angled positions available, wall mountable

Wall Mountable
Yes

QoSs
Layer 2 QoS (802.1Q, 802.1P)
and Layer 3 (ToS, DiffServ, MPLS) QoS

Security
User and administrator level passwords, MD5 and MD5-sess based authentication, 256-bit AES encrypted configuration file, SRTP, TLS, 802.1x media access control

Multi-language
English, German, Italian, French, Spanish, Portuguese, Russian, Croatian, Chinese, Korean, Japanese

Upgrade/Provisioning
Firmware upgrade via TFTP / HTTP / HTTPS, mass provisioning using TR-069 or AES encrypted XML configuration file.

Power and Green Energy Efficiency
Universal power adapter included: Input:100-240V; Output: +12V, 0.5A ; Integrated Power-over-Ethernet (802.3af) Max power consumption: 6.4W (power adapter) or 6.49W (PoE)

Physical
Dimension: 201mm(W) x 193mm(L) x 85mm(H);
Unit weight: 0.85kg;
Package weight:1.12kg

Temperature and Humidity
Operation: 0°C to 40°C
Storage: -10°C to 60°C
Humidity: 10% to 90% Non-condensing

Package Content
GXP2135 phone, handset with cord, base stand, universal power supply, network cable, Quick Installation Guide, GPL license

Compliance
FCC: Part 15 (CFR 47) Class B
CE: EN55022 Class B; EN55024 Class B; EN61000-3-2; EN61000-3-3;EN60950-1
RCM: AS/ACIF S004; AS/NZS CISPR22/24; AS/NZS 60950.1

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GRANDSTREAM
GXP2140
A versatile Enterprise IP Phone

A versatile Enterprise IP phone, the GXP2140 is a Linux-based device that includes 4 lines, 5 XML programmable soft keys, and 5-way conferencing. A 4.3-inch color LCD screen and HD audio allow for a crisp display and high quality calls. The GXP2140 comes equipped with Bluetooth, USB and EHS capabilities for flexibility. The phone also comes pre-loaded with weather & currency exchange apps. Add up to four GXP2200EXT modules to view an additional 160 lines, and customize your language for global use.

Features

4 LINES	4 lines, with up to 4 SIP accounts, 4 dualcolored line keys	EHS	Electronic Hook Switch (EHS) support for Plantronics
Gigabit	Dual switched auto-sensing 10/100/1000Mbps Gigabit network ports		Automated provisioning options include TR-069 and XML config files
	Supports the GXP2200 EXT Module for up to 160 speed dial/BLF contacts		TLS and SRTP security encryption technology to protect calls and accounts
	Built-in Bluetooth for syncing headsets and mobile devices for contact books, calendars & call transferring		Built-in USB ports for importing/exporting data
HD	HD audio to maximize audio quality and clarity, full-duplex speakerphone	5 WAY	5-way audio conferencing for easy conference calls
PoE	Built-in PoE to power the devices and give it a network connection		

Specifications

Protocols/Standards
SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE,TELNET, TFTP, NTP, STUN, SIMPLE, LLDP, LDAP, TR-069, 802.1x, TLS, SRTP, IPv6

Network Interfaces
Dual switched auto-sensing 10/100/1000 Mbps
Gigabit Ethernet ports with integrated PoE

Graphic Display
4.3-inch (480x272) TFT color LCD

Bluetooth
Yes, Bluetooth V2.1

Feature Keys
4 line keys with up to 4 SIP accounts, 5 programmable context-sensitive soft keys, 5 navigation/menu keys,
11 dedicated function keys for: MESSAGE (with LED indicator), PHONEBOOK, TRANSFER, CONFERENCE, HOLD, HEADSET, MUTE, SEND/REDIAL, SPEAKERPHONE, VOL+, VOL

Voice Codecs
Support for G.729A/B, G.711µ/a-law, G.726, G.722 (wide-band), and iLBC,in-band and out-of-band DTMF (in audio, RFC2833, SIP INFO)

Auxiliary Ports
RJ9 headset jack (allowing EHS with Plantronics headsets), USB, extension module port

Telephony Features
Hold, transfer, forward, 5-way conference, call park, call pickup, shared-call-appearance (SCA)/bridged-line-appearance (BLA), downloadable phonebook (XML, LDAP, up to 2000 items), call waiting, call log (up to 500 records), customization of screen, off-hook auto dial, auto answer, click-to-dial, flexible dial plan, hot desking, personalized music ringtones and music on hold, server redundancy and fail-over

Sample Applications
Weather, currency, news, XML

HD Audio
Yes, both on handset and speakerphone

Extension Module
Yes, can power up to 4 GXP2200EXT modules which feature a 128x384 graphic LCD, 20 quick-dial/BLF keys with dual-color LED, 2 navigation keys, and less than 1.2W power consumption per unit.

Base Stand/Wall Mountable
Yes, allow 2 angle positions

QoSs
Layer 2 (808.1Q, 802.1p) and Layer 3 (ToS, DiffServ, MPLS) QoS

Security
User and administrator level passwords, MD5 and MD5-sess based authentication, AES-based secure configuration file, SRTP, TLS, 802.1x media access control

Multi-language
English, Arabic, Chinese, Croatian, Czech, Dutch, German, French, Hebrew, Hungarian, Italian, Japanese, Korean, Polish, Portuguese, Russian, Slovenian, Spanish, Turkish

Upgrade/Provisioning
Firmware upgrade via TFTP/HTTP/HTTPS, mass provisioning using TR-069 or encrypted XML configuration file.

Power and Green Energy Efficiency
Universal power adapter included: Input: 100-240V; Output: +12V, 1.0A; Integrated Power-over-Ethernet (802.3af) Max power consumption: 6W (without GXP2200EXT), 10W(with 4 cascaded GXP2200EXTs)

Physical
Dimention: 228mm (W) x 206mm (L) x 46.5mm (H);
Unit weight: 0.98kg; Package weight: 1.55kg

Temperature and Humidity
Operation: 0°C to 40°C
Storage: -10°C to 60°C
Humidity: 10% to 90% Non-condensing

Package Content
GXP2140 phone, handset with cord, base stand, universal power supply, network cable, Quick Start Guide

Compliance
FCC: Part 15 (CFR 47) Class B; EN55022 Class B, EN55024, EN61000-3-2, EN61000-3- 3, EN 60950-1, EN62479 AS/NZS CISPR 22 Class B, AS/NZS CISPR 24, RoHS; UL 60950 (power adapter)

GRANDSTREAM

GXP2170

An Enterprise IP Phone for High-Volume Users

The GXP2170 is a powerful enterprise-grade IP phone that is ideal for busy users who handle high call volumes. This top-of-the-line Enterprise IP Phone features up to 12 line keys/line appearances and 6 SIP accounts using a 4.3-inch color display LCD and full HD audio. It includes up to 48 digital, on-screen speed dial/BLF keys to help users be more productive and efficent. The GXP2170 supports the fastest possible connection speeds with dual Gigabit network ports, features integrated PoE and includes built-in Bluetooth for syncing with mobile devices and Bluetooth headsets. This Enterprise IP phone can connect to up to four GXP2200 EXT modules with LCD display to access up to 160 speed dial/BLF contacts. The GXP2170 is the perfect choice for enterprise users looking for a top-notch executive IP phone with advanced functionality.

Features

12 LINES	12 dual-color line keys (with 6 SIP accounts), 5 XML programmable context sensitive soft keys	EHS	Supports EHS compatible Plantronics headsets
Gigabit	Dual switched auto-sensing 10/100/1000Mbps Gigabit network ports		Automated provisioning options include TR-069 and XML config files
	48 digitally programmable & customizable BLF/fast dial keys, and supports up to 4 cascaded XP2200EXT Modules		Built-in USB ports for importing/exporting data
	Built-in Bluetooth for syncing headsets and mobile devices for contact books, calendars & call transferring		TLS and SRTP security encryption technology to protect calls and accounts
HD	HD audio to maximize audio quality and clarity; full duplex speakerphone	5 WAY	5-way audio conferencing for easy conference calls
PoE	Built-in PoE to power the devices and give it a network connection	zero CONFIG	Use with Grandstream's UCM series IP PBX appliance for Zero-Config provisioning, 1-touch call recording & more

Specifications

Protocols/Standards
SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, TELNET, TFTP, NTP, STUN, SIMPLE, LLDP, LDAP, TR-069, 802.1x, TLS, SRTP, IPV6VAD, CNG, AEC, PLC, AJB, AGC

Network Interfaces
Dual-switched auto-sensing 10/100/1000 Mbps Gigabit Ethernet ports with inte-grated PoE

Graphic Display
4.3-inch (480x272) TFT color LCD

Bluetooth
Yes, integrated

Feature Keys
12 line keys with up to 6 SIP accounts, 5 XML programmable context-sensitive soft keys, 5 navigation/menu keys,
11 dedicated function keys for : MESSAGE(with LED indicator), PHONEBOOK, TRANSFER, CONFERENCE, HOLD, HEADSET, MUTE, SEND/REDIAL, SPEAKERPHONE, VOL+, VOL

Voice Codecs
Support for G7.29A/B, G.711µ/a-law, G.726, G.722 (wide-band), in-band and out of-band DTMF (in audio, RFC2833, SIP INFO)

Auxiliary Ports
RJ9 headset jack (allowing EHS with Plantronics headsets), USB, extension module port

Telephony Features
Hold, transfer, forward, 5-way conference, call park, call pickup, shared-call appearance (SCA)/bridged-line-appearance (BLA), downloadable phonebook (XML, LDAP, up to 2000 items), call waiting, call log (up to 500 records), XML customization of screen, off-hook auto dial, auto answer, click-to-dial, flexible dial plan, hot desking, personalized music ringtones and music on hold, server redundancy and fail-over

HD Audio
Yes, HD handset and speakerphone with support for wideband audio

Extension Module
Yes, can power up to 4 GXP2200EXT modules which feature a 128x384 graphic LCD, 20 quick-dial/BLF keys with dual-color LED, 2 navigation keys, and less than 1.2W power consumption per unit.

Base Stand
Yes, 2 angle positions available, Wall Mountable

QoSs
Layer 2 QoS (802.1Q, 802.1P) and
Layer 3 (ToS, DiffServ, MPLS) QoS

Security
User and administrator level passwords, MD5 and MD5-sess based authentication, 256-bit AES encrypted configuration file, SRTP, TLS, 802.1x media access control

Multi-language
English, German, Italian, French, Spanish, Portuguese, Russian, Croatian, Chinese, Korean, Japanese

Upgrade/Provisioning
Firmware upgrade via TFTP / HTTP / HTTPS, mass provisioning using TR-069 or AES encrypted XML configuration file.

Power and Green Energy Efficiency
Universal power adapter included: Input:100-240V ; Output: +12V, 1.0A ; Integrated Power-over-Ethernet (802.3af)
Max power consumption : 5.4W(without GXP2200EXT) or 9.2W(with 4 cascaded GXP2200EXTs)

Physical
Dimension: 228mm(W) x 206mm(L) x 46mm(H);
Unit weight:0.98kg ; Package weight:1.43kg

Temperature and Humidity
Operation: 0°C to 40°C
Storage: -10°C to 60°C
Humidity: 10% to 90% Non-condensing

Package Content
GXP2170 phone, handset with cord, base stand, universal power supply, network cable, Quick Installation Guide, GPL license

Compliance
FCC: Part 15 (CFR 47) Class B
CE: EN55022 Class B; EN55024 Class B; EN61000-3-2; EN61000-3-3;EN60950-1
RCM: AS/ACIF S004; AS/NZS CISPR22/24; AS/NZS 60950.1

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GRANDSTREAM

GXP2200 EXT

Expansion Module

GXP2200EXT delivers additional functionality, versatility and flexibility to Grandstream's GXP2140 and GXP2170 Enterprise IP Phones and GXV3240 Video IP Phone for Android™. The GXP2200EXT module features a large 128 x 384 graphic LCD and 20 programmable buttons (each with dual color LED). It offers up to 40 extensions per module by using the 2 page switch keys and up to 160 buttons when 4 extension modules are daisy-chained together.This extension module is connected, powered, and controlled by the host phone to provide the benefit of additional extension keys instantly. The GXP2200EXT supports the traditional call features on each of its programmable buttons, BLF (busy lamp field, standard or eventlist), call park/pick-up, speed dial, presence, intercom, and conference/ transfer/forward. The GXP2200EXT is the ideal solutions for any receptionist or businesses managing high call volume. When using the GXP2200EXT, the telephone attendant can ensure maximum productivity by efficiently monitoring and dispatching multiples incoming calls.

Specifications

Lines
20 per page (each module contains 2 pages, for up to 40 lines per module
Up to 160 with 4 daisy-chained modules

Compatible Grandstream IP phones
GXP2140, GXP2170 and GXV3240

Feature Support
Local GUI with animation driven from the host GXP2140 or GXP3240 phone; Multiple line/call appearances

Power
Powered by the host phone

Firmware Upgrades
Delivered by the host phone

Dimensions (L x W x H)
206 mm x 117mm x 32mm

Weight
0.38kg

Temperature
0 ~ 40°C (32 ~ 104°F)

Humidity
10%-90% Non-condensing

Compliance
FCC/CE/C-Tick

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GRANDSTREAM

HT801

An easy-to-use 1 port ATA

The HT801 is a single port analog telephone adapter (ATA) that allows users to create a high-quality and manageable IP telephony solution for residential and office environments. Its ultra-compact size, voice quality, advanced VoIP functionality, security protection and auto provisioning options enable users to take advantage of VoIP on analog phones. It also allows service providers to offer high quality IP service to their market. The HT801 is an ideal ATA for individual use as well as commercial IP voice deployments worldwide.

Features

10 LINES	Supports 1 SIP profile through a single FXS port and a single 10/100Mbps port		TLS and SRTP security encryption technology to protect calls and accounts
	Automated provisioning options include TR-069 and XML config files	3 WAY	Supports 3-way voice conferencing
	Failover SIP server automatically switches to secondary server if main server loses connection		Supports T.38 Fax for creating Fax-over-IP
	Supports a wide range of caller ID formats	zero CONFIG	Use with Grandstream's UCM series of IP PBXs for Zero Configuration provisioning
	Supports advanced telephony features, including call transfer, call forward, callwaiting, do not disturb, message waiting indication, multi-language prompts, flexible dial plan and more		

Specifications

Telephone Interfaces
One (1) FXS port

Network Interfaces
One (1) 10/100Mbps auto-sensing ethernet port (RJ45)

LED Indicators
POWER, INTERNET, PHONE

Factory Reset Button
Yes

Telephony Features
Caller ID display or block, call waiting, flash, blind or attended transfer, forward hold, do not disturb, 3-way conference

Voice Codecs
G.711 with Annex I (PLC) and Annex II (VAD/CNG), G.723.1, G.729A/B, G.726, iLBTC, OPUS, dynamic jitter buffer, advanced line echo cancellation

Fax Over IP
T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through

Short/Long Haul Ring Load
2 REN: Up to 1km on 24 AWG

Caller ID
Bellcore Type 1 & 2, ETSI, BT, NTT, and DTMF-based CID

Disconnect Methods
Busy Tone, Polarity Reversal/Wink, Loop Current

Network Protocols
TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP/RARP, ICMP, DNS, DHCP, NTP, TFTP, SSH, STUN, SIP (RFC3261), SIP over TCP/TLS, SRTP, TR-069

QoS
Layer 2 (802.1Q VLAN, SIP/RTP 802.1p) and Layer 3 (ToS, DiffServ, MPLS)

DTMF Method
In-audio, RFC2833 and/or SIP INFO

Provisioning and Control
HTTP, HTTPS, SSH, TFTP, TR-069, secure and automated provisioning using AES encryption, syslog

Media
SRTP

Control
TLS/SIPS/HTTPS

Management
Syslog support, SSH, remote management using web browser

Universal Power Supply
Input: 100-240VAC, 50-60Hz
Output: 5.0VDC/1.0A

Environmental
Operational: 32° – 104°F or 0° – 40°C
Storage: 14° – 140°F or -10° – 60°C
Humidity: 10 – 90% Non-condensing

Dimension and Weight
Dimensions: 100mm x 100mm x 29.5mm
Weight: 102 g

Compliance
FCC: Part15B
CE: EN55032, EN55024, EN61000-3-2, EN61000-3-3, EN60950-1
RCM: AS/NZS CISPR22, AS/NZS60950.1, S003 K.21

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YEALINK

CP920

Touch-sensitive HD IP Conference Phone

It is time to optimize your conference room with a Yealink modern conference phone – the Yealink Touch-sensitive HD IP Conference Phone CP920. With user-centric design philosophy, this new release from Yealink combines simplicity of use with sophistication of features, being perfect for a small-to-medium-size conference room. In regard to its crystal-clear audio quality, your conversation will sound natural and bright anywhere with well-designed CP920. The Yealink CP920 can be paired with your mobile staff – smartphone or PC/tablet via Bluetooth. It is also a good choice for the companies that use a public-switched telephone network (PSTN) after combining with CPN10 PSTN Box. As a valuable complement for your conference room, CP920 conference phone strikes an excellent balance between ease-of-use and powerful features, giving you a simply and clearly engaging business conference experience.

Features

Superior audio quality	PSTN/SIP	Built-in Wi-Fi	Built-in Bluetooth	Local 3-way PSTN conference	Call recording	Hybrid UC meeting

Sensitive Touch, Elegant Control

This Y-shape phone released from Yealink, representing the first letter Y of Yealink, has a sensitive touch keypad which perfectly cancels the keypad noise and facilitates your conference room experience by simply putting all of the key conferencing functions at your fingertips.

Superior Audio Quality

The Yealink CP920 conference phone, marrying the Yealink Noise Proof Technology, unburdens the business conversation by reducing annoying noise and minimizing distractions to set audio experience into a new level. Thanks to its built-in 3-microphone array, CP920 has a 20-foot (6-meter) and 360-degree voice pickup range. It's as if all participants are sitting across the table from each other!

Simultaneous Analog-plus-IP

To protect a business owner's investments by supporting the migration to VoIP, CP920 allows you to connect Yealink PSTN box CPN10 with traditional analog phone lines. Without purchasing any extra PSTN server, cascading two CPN10 can directly implement a local three-way PSTN conference and offers flexible modes for business users: analog, IP or simultaneous analog-plus IP.

Hybrid UC Meeting

With Yealink CP920, up to five parties can join a conference call from different locations, helping your company cut costs and save time. Pairing Yealink CP920 with your smartphone or PC/tablet via Bluetooth allows you to turn the Yealink CP920 into a loudspeaker or microphone with ease. In addition, you can merge your smartphone or PC/tablet, SIP call and PSTN call into one hybrid UC meeting

- Optimal HD audio, full duplex technology
- Yealink Noise Proof Technology
- 20-foot (6-meter) and 360-degree voice pickup
- Built-in 3-microphone array
- Sensitive touch keypad
- 3.1" 248x120-pixel graphical LCD with backlight
- Power over Ethernet
- 5-way conference call
- Hybrid UC meeting
- Built-in Wi-Fi (2.4GHz, 802.11.b/g/n)
- Built-in Bluetooth 4.0
- Local USB call recording
- Local 3-way PSTN conference via Yealink CPN10

Specifications

Audio Features

- Optimal HD audio
- Yealink Noise Proof Technology
- Background noise suppression
- 20-foot (6-meter) microphone pickup range
- Apply to small to medium conference room
- Built-in 3-microphone array, 360-degree voice pickup
- 56mm diameter and 5w speaker
- Full-duplex speakerphone with AEC
- Echo cancellation tail length is up to 320ms
- Codecs: G722, G722.1C, G726, G729, G723, iLBC, Opus, PCMA, PCMU
- DTMF: In-band, Out-of-band (RFC 2833) & SIP INFO
- VAD, CNG, PLC, AJB, AGC

Phone Features

- 1 VoIP account
- Call hold, mute, DND, call recording, hotline
- 5-way conference call
- Flash
- Redial, call waiting, emergency call
- Call forward, call transfer, call return, dial plan
- Ring tone selection/import/delete
- Set date time manually or automatically
- Volume adjustment
- Pairing via Bluetooth

Directory

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

IP-PBX Features

- Intercom
- Multicast paging
- Anonymous call
- Anonymous call rejection
- Voice mail
- Distinctive ringtone
- Call pickup

Display

- 3.1" 248x120-pixel graphical LCD with backlight
- 25 total keys:
  4 soft keys, off-hook key, on-hook key,
  12-key numerical keypad, Bluetooth, mute,
  volume keys, 2 navigation keys, OK key
- Phone lock
- Multilingual user interface
- Caller ID with name and number

Interface

- 1 x RJ45 10/100M Ethernet port
- Built-in Wi-Fi (2.4GHz，802.11 b/g/n)
- Built-in Bluetooth 4.0
- Power over Ethernet (IEEE 802.3af), class 3
- 1 x USB 2.0 port
- 1 x Security slot

Network and security

- SIP v1 (RFC2543), v2 (RFC3261)
- SIP server redundancy supported
- IPv4/IPv6
- NAT traversal: 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 encryption
- 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
- Zero-sp-touch, TR-069, SNMP
- Reset to factory, reboot
- Package tracing export, system log

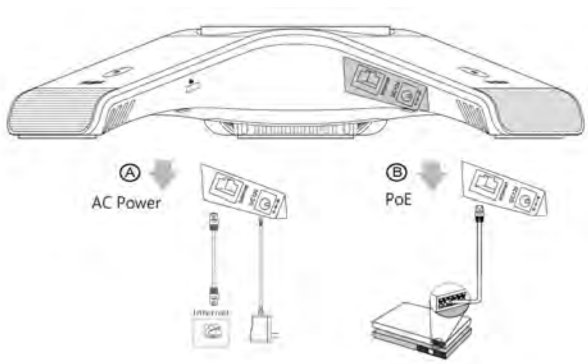
Other physical features

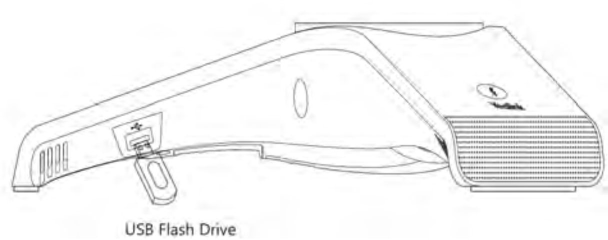
- External Yealink power adapter:
- AC 100~240V input and DC 12V/1A output
- Power consumption (PoE): 3.9w-7.6w
- Dimension (W*D*H): 307.8mm*298.5mm*66.9mm
- Operating humidity: 10~90%
- Storage temperature: -10~40°C (+14~104°F)

Package Features

- Package content:
  - Yealink CP920 IP conference phone
  - Ethernet Cable (7.5m CAT5E UTP cable)
  - Quick Start Guide
  - Yealink Power Adapter
- Qty/CTN: 5 PCS
- N.W/CTN: 8.046 kg
- G.W/CTN: 9.062 kg
- Giftbox size: 336mm*364mm*112mm
- Carton Meas: 587mm*348mm*376mm

Compliance





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YEALINK

EXP40

High-Performance LCD Expansion Module

The EXP40 Expansion Module for the SIP-T46S, SIP-T46G, SIP-T48S and SIP-T48G, expands the functional capability of your SIP phone to a whole new level. It features a large graphic LCD. Two pages of 20 flexible buttons are shown on the display can be programmed up to 40 various features, the productivity-enhancing features include BLF/BLA, speed dialing, call forward, transfer, park, pickup, etc.

Revolutionarily new design

New design includes tiny details to match with the look and feel of T46G. Rubber covers on the underside help prevent the phone from sliding, The new foot stand allows two positions for the device. The backlit display eliminates the need for external light.

A rich visual experience for applications

Equipped with a 160x320 graphic LCD with Backlight. 20 physical keys each with a dual-color LED and two page views are possible, this allows 40 additional programmable keys that can be used for speed dialing, BLF/BLA, call forward, transfer, park, pickup, etc. 2 independent control keys are used for fast switch pages.

Expandability

Supports up to 6 Expansion Modules for an attendant console application, adding up to 240 additional buttons. It has to be powered by a conventional power supply (5V/1.2A) if it exceeds 2 connections.

- An exclusive collection with quality and detailed designs
- Rich visual experience with 160x320 graphic LCD
- 20 physical keys each with a dual-color LED
- 2 independent control keys are used for fast switch pages
- Stand with 2 adjustable angles
- Wall mountable

Specifications

Display

- 160x320 graphic LCD with 16-level grayscales
- LCD Backlight
- Two page views are possible
- Different icons for each function shown on the LCD

Features Keys and Indicator

- 20 physical keys each with a dual-color LED
- 20 additional keys through page switch
- 2 independent control keys are used for fast switch pages
- Programmable for shared line, BLF List, call park, conference, forward, group pickup, group listening, LDAP, Pick UP, XML Browser

Physical Feature

- Wall mountable
- Stand with 2 adjustable angles
- Expansion module (≤2) is powered by the host phone
- Supports up to 6 modules daisy-chain
- 2xRJ12（6P6C）ports for data in and out
- Dimension (W*D*H*T): 127mm*213mm*167mm*45mm
- Applies to Yealink SIP-T46S, SIP-T46G, SIP-T48S, SIP-T48G
- Operating temperature: -10~50°C (+14~122°F) Operating humidity: 10-95%

Package Features

- Qty/CTN: 10 PCS
- N.W/CTN: 6.7 kg
- G.W/CTN: 7.6 kg
- Giftbox size: 237mm*154mm*100mm
- Carton Meas: 520mm*317mm*247mm

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YEALINK

SIP-T21(P) E2

Entry-level IP phone with 2 Lines & HD voice

Yealink's new SIP-T21(P) E2 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 with backlight showing a clear 5-line data display, it offers a smoother user experience, much more visual information at a glance, plus HD Voice characteristics. Meanwhile, the dual 10/100 Mbps network ports with integrated PoE makes T21P E2 an ideal choice for extended network use. The T21(P) E2 supports two VoIP accounts, 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 T21(P) E2 maximizes productivity in both small and large office environments.

Features

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, the SIP-T21(P) E2 provides clearer, more lifelike voice communications.

Enhanced Call Management

The SIP-T21(P) E2 supports vast productivity- enhancing features such as XML Browser, call park, call pickup, BLF, call forward, call transfer, 3-way conference, which makes it the natural and obvious efficiency tool for today's busy small and large offices environment.

Efficient Installation and Provisioning

The Yealink SIP-T21(P) E2 supports efficient provisioning and effortless mass deployment with Yealink's Redirection and Provisioning Service (RPS) and Boot mechanism to help you realize the Zero Touch Provisioning without any complex manual settings, which makes it simple to deploy, easy to maintain and upgrade.

Highly secure transport and interoperability

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

- Yealink HD Voice
- 132 x 64-pixel graphical LCD with backlight
- Two-port 10/100M Ethernet Switch
- PoE support (T21P E2)
- Opus codec support
- Up to 2 SIP accounts
- Headset support
- Wall mountable
- Simple, flexible and secure provisioning options

Specifications

Audio Features

- HD voice: HD handset, HD speaker
- Wideband codec: Opus, 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

- 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
- Integrated screenshots
- RTCP-XR
- Enhanced DSS Key

Directory

- Loal phonebook up to 1000 entries
- Black list
- XML/LDAP remote phonebook
- Smart dialing
- 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, voice mail
- Flexible seating
- Call park, call pickup
- Executive and Assistant
- Centralized call recording
- Visual voice mail
- Call recording

Display and Indicator

- 32 x 64-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
- Multilingual user interface
- Caller ID with name and number
- Power saving

Feature keys

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

Interface

- 2 x RJ45 10/100M Ethernet ports
- Power over Ethernet (IEEE 802.3af), class 1 (T21P E2)
- 1 x RJ9 (4P4C) handset port
- 1 x RJ9 (4P4C) headset port

Other Physical Features

- Wall mountable
- External Yealink AC adapter：
  AC 100~240V input and DC 5V/600mA output
- Power consumption (PSU): 0.8-1.4W
- Power consumption (PoE): 1.3-1.8W (T21P E2)
- Dimension (W*D*H*T):
  209 mm*188 mm*150 mm*41 mm
- Operating humidity: 10~95%
- Operating temperature: -10~50°C (+14~122°F)

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

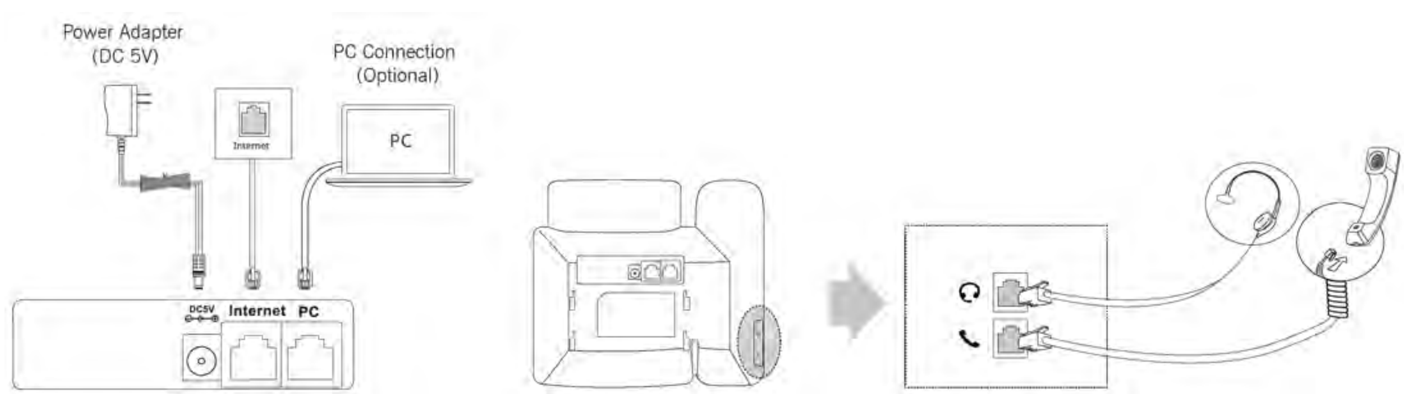
- SIP v1 (RFC2543), v2 (RFC3261)
- Call server redundancy supported
- NAT traversal: STUN mode
- Proxy mode and peer-to-peer SIP link mode
- IP assignment: static/DHCP
- HTTP/HTTPS web server
- Time and date synchronization using SNTP
- UDP/TCP/DNS-SRV (RFC 3263)
- QoS: 802.1p/Q tagging (VLAN), Layer 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
- LLDP/CDP/DHCP VLAN
- ICE

Package Features

- Package content:
  - Yealink SIP-T21(P) E2 IP phone
  - Handset with handset cord
  - Ethernet Cable (1.5m CAT5E UTP Cable)
  - Stand
  - Quick Start Guide
  - Power Adapter:
    T21 E2 (Standard)/T21P E2 (Optional)
- Qty/CTN: 10 PCS
- N.W/CTN: 11.7 kg
- G.W/CTN: 12.5 kg
- Giftbox size: 215 mm*200 mm*118 mm
- Carton Meas: 615 mm*436 mm*208 mm

Compliance

REACH ISO 9001



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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 accounts, 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 maximizes productivity in both small and large office environments.

Features

Optima HD Voice	Opus Codec

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. Provides clearer, more lifelike voice communications.

Enhanced Call Management

The SIP-T21P supports vast productivity-enhancing features 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 office environments.

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 and ensures 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

Specifications

Audio Features
- HD voice: HD handset, HD speaker
- Wideband codec: G.722
- Narrowband codec: G.711(A/μ), G.723.1, 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
- 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
- Ringtone selection/import/delete
- Set date time manually or automatically
- Dial plan
- XML Browser, action URL/URI

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

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 and 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) handset port

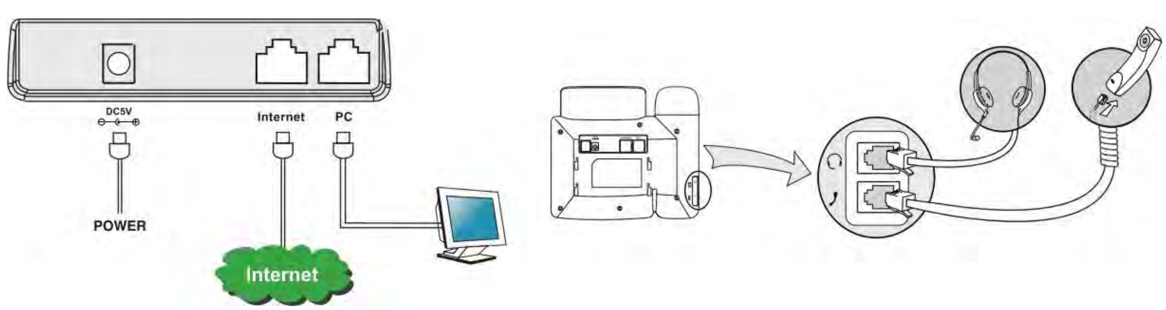
Other Physical Features
- Wall mountable
- External universal AC adapter (optional) : AC 100~240V input and DC 5V/600mA output
- Power consumption (PSU): 1.2-1.9W
- Power consumption (PoE): 1.8-2.3W
- Dimension(W*D*H*T): 209mm*188mm*150mm*41mm
- Operating humidity: 10~95%
- Operating temperature: -10~50°C

Package Features
- Qty/CTN: 10 PCS
- N.W/CTN: 10.8 kg
- G.W/CTN: 12.3 kg
- Giftbox size: 215mm*200mm*118mm
- Carton Size: 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
CE FC RoHS REACH ISO 9001



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YEALINK

SIP-T27G

Standard and Affordable SIP Phone for Business

With faster response on the phone's user interface and better device performance, the SIP-T27G IP phone, as the upgraded product of T27P, is Yealink's latest feature-rich tool unifying superior voice capabilities and increased function extension capability for business. Yealink's Optima HD technology and wideband codec of Opus deliver a superb sound quality and bring you life-like voice communications. With an all-new USB port, the SIP-T27G boasts unparalleled functionality and expansibility with Bluetooth, Wi-Fi and USB recording features. Seamlessly migrated to a GigE-based network infrastructure, SIP-T27G IP phone is also built with Gigabit Ethernet, technology for rapid call handling. By using standard encryption protocols, the SIP-T27G performs highly secure remote provisioning and software upgrades.

Features

Optima HD Voice	Opus Codec	Gigabit	USB 2.0	Paperless

HD Audio

Yealink Optima HD Voice technology combines cutting-edge hardware and software with wideband technology for maximum acoustic performance. Being a totally open, highly versatile audio codec, Opus is designed to perform a higher HD audio quality than other wideband codecs in a high network bandwidth. However, if your current network quality is poor, Opus can provide you with a better audio quality than other narrowband codecs.

Easy Customization and High Expandability

With an all-new USB port, the SIP-T27G is a powerful and expandable office phone with Bluetooth, Wi-Fi and USB recording features. It has three pages of flexible buttons which can be programmed with up to 21 paperless DSS keys.

Efficient Installation and Provisioning

The Yealink SIP-T27G supports efficient provisioning and effortless mass deployment. Yealink's Redirection and Provisioning Service (RPS) and Boot mechanism helps you carry the Zero Touch Provisioning without any complex manual settings. it's simple to deploy, easy to maintain and upgrade.

Secure Transport and Interoperability

The SIP-T27G uses SIP over Transport Layer Security (TLS/SSL), which is the latest network security technology. It's also compatible with leading soft switch suppliers.

- Yealink Optima HD voice
- 3.66" 240x120-pixel graphical LCD with backlight
- Gigabit
- USB 2.0
- Opus* codec support
- Up to 6 SIP accounts
- Paper label free design
- PoE support
- Headset, EHS support
- Supports expansion modules

Specifications

Audio Features
- HD voice: HD handset, HD speaker
- Wideband codec: Opus*, G.722
- Narrowband codec: Opus*, G.711(A/μ), G.723.1, G.729, 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
- 6 VoIP accounts
- One-touch speed dial, redial
- Call forward, call waiting
- Call transfer, call hold
- Call return, group listening
- Mute, auto answer, DND
- 3-way conference call
- Direct IP call without SIP proxy
- Ringtone selection/import/delete
- Hotline, emergency call
- Set date time manually or automatically
- Dial plan, XML Browser, Action URL/URI
- Integrated Screenshots
- RTCP-XR
- USB port (2.0 compliant) Bluetooth earphone through BT40, Wi-Fi through WF40, USB call recording through USB flash drive
- Enhanced DSS key

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, voice mail
- Flexible seating
- Call park, call pickup
- Executive and Assistant
- Centralized call recording
- Visual voice mail

Display and Indicator
- 3.66” 240x120-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
- Multilingual user interface
- Caller ID with name and number
- Power saving

Feature keys
- 8 line keys with LED
- 8 line keys can be programmed up to 21 paperless DSS keys (3-page view)
- 8 feature keys: message, headset, conference, mute, hold, transfer, redial, hands free speakerphone
- 4 context-sensitive “soft” keys
- 6 navigation keys
- Volume control keys
- Illuminated mute key
- Illuminated headset key

Interface
- Dual-port Gigabit Ethernet
- Power over Ethernet (IEEE 802.3af), Class 3
- 1 x USB port (2.0 compliant)
- 1 x RJ9 (4P4C) handset port
- 1 x RJ9 (4P4C) headset port
- 1 x RJ12 (6P6C) EHS port
- 1 x RJ12 (6P6C) EXT port: Supports up to 6 Expansion Modules

Other Physical Features
- Stand with 2 adjustable angles
- Wall mountable
- External Yealink AC adapter (optional): AC 100~240V input and DC 5V/2A output
- Power consumption (PSU): 1.4-3.0W
- Power consumption (PoE): 1.7-7.0W
- Dimension (W*D*H*T): 265mm x 210mm x 170.5mm x 53.5mm
- Operating humidity: 10-95%
- Operating temperature: -10~50°C (+14~122°F)

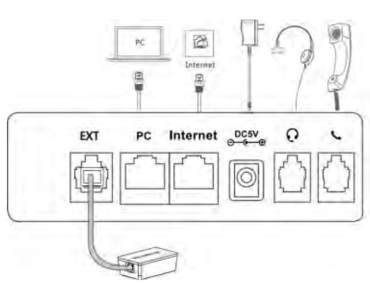
Management
- Configuration: browser/phone/auto-provision
- Auto provision via FTP/TFTP/HTTP/HTTPS for mass deploy
- Auto provision with PnP
- Zero-sp-touch, TR-069
- Phone lock for personal privacy protection
- Reset to factory, reboot
- Package tracing export, system log

Network and Security
- SIP v1 (RFC2543), v2 (RFC3261)
- Call server redundancy supported
- NAT traversal: 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
- LLDP/CDP/DHCP VLAN
- ICE

Package Features
- Package content:
  - Yealink SIP-T27G IP phone
  - Handset with handset cord
  - Ethernet Cable (1.5m CAT5E UTP Cable)
  - Stand
  - Quick Start Guide
  - Power Adapter (Optional)
  - Wall Mount Bracket (Optional)
- Qty/CTN: 5 PCS
- N.W/CTN: 6.6 kg
- G.W/CTN: 7.4 kg
- Giftbox size: 295mm x 224mm x 115mm
- Carton Size: 602mm x 308mm x 236mm

Compliance

REACH ISO 9001



*Opus: Support 8 kHz (narrowband) and 16 kHz (wideband) sampling rate

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YEALINK
SIP-T29G
Professional Gigabit Phone with Color LCD

SIP-T29G IP Phone is the most advanced model in the Yealink T2x IP terminal series. It has a high-resolution TFT color display, delivers a rich visual experience. Yealink Optima HD technology enables rich, clear, life-like voice communications. Supports Gigabit Ethernet, a variety of device connections, including EHS headset and USB. With programmable keys, the IP Phone supports vast productivity enhancing features.

Features

Optima HD Voice	Paperless	USB 2.0	Gigabit

New updated and improved model

Yealink 's SIP Phones continue to evolve, the function of new SIP-T2 Series has become more abundant, the new user-interface with color display has become more friendlier.

HD Audio

Yealink Optima 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. Creating an amazing face-to-face live experience.

A rich visual experience for applications

SIP-T29G Supports vast productivity-enhancing feature such as XML Browser, SCA, BLF List, call forward, call transfer, 3-way conferencing, the 4.3 inch backlit color display enables rich visual presentation and easier navigation of the menu. Three pages of 10 flexible buttons are shown on the display can be programmed up to 27 various features.

Advanced connectivity and expandability

For network connectivity, the SIP-T29G comes with two Gigabit Ethernet ports, one of them suitable for Power over Ethernet. Supports a variety device connections, including EHS headset and USB. A built-in USB 2.0 port can also be used for Bluetooth, Wi-Fi and USB recording.

Supports up to 6 Expansion Modules, adding up to 240 additional buttons with a screen based LCD display and LED system

- 4.3" 480 x 272-pixel color display with backlight
- Yealink Optima HD voice
- USB 2.0
- USB recording
- Wi-Fi via WF40
- Bluetooth via BT40
- Up to 16 SIP accounts
- Dual-port Gigabit Ethernet
- PoE support
- Paperless label design
- Headset, EHS support
- Integrated stand with 2 adjustable
- Wall mountable

Specifications

Audio Features
- HD voice: HD handset, HD speaker
- Wideband codec: Opus*, G.722
- Codecs: G.722, G.711(A/μ), G.723, 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
- 16 VoIP accounts
- One-touch speed dial, hotline
- Call forward, call waiting, call transfer
- Group listening, SMS, emergency call
- Redial, call return, auto answer
- 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
- Integrated screenshots
- RTCP-XR
- USB port (2.0 compliant) for:
  Bluetooth earphone through BT40,
  Contact synchronization through BT40,
  Wi-Fi through WF40,
  USB call recording through USB flash drive
- Enhanced DSS Key
- 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
- Hotdesking, voice mail
- Music on hold
- Message Waiting Indicator (MWI)
- Call park, call pickup
- Intercom, paging
- Call completion, call recording
- Flexible seating
- Executive and Assistant
- Centralized call recording
- Visual Voice Mail

Display and Indicator
- 4.3" 480 x 272-pixel color display with backlight
- 16 bit depth color
- LED for call and message waiting indication
- Dual-color (red or green) illuminated LEDs for line status information
- Wallpaper
- Multilingual user interface with icons and soft keys
- Caller ID with name, number and photo
- Power saving
- Screensaver

Feature keys
- 10 line keys with LED
- 10 line keys can be programmed up to 27 various features (3-page view)
- 8 features keys: message, headset, conference, hold, mute, transfer, redial, hands-free speakerphone
- 4 context-sensitive "soft" keys
- 6 navigation keys
- Volume control keys
- Illuminated headset key

Interface
- Dual-port Gigabit Ethernet
- Power over Ethernet (IEEE 802.3af), class 0
- 1 x USB port (2.0 compliant)
- 1 x RJ9 (4P4C) handset port
- 1 x RJ9 (4P4C) headset port
- 1 x RJ12 (6P6C) EXT port

Supports up to 6 Expansion Modules for an attendant console application

Other Physical Features
- Stand with 2 adjustable angles
- Wall mountable
- External Yealink AC adapter (optional) : AC 100~240V input and DC 5V/2A output
- Power consumption (PSU): 1.6-3.9W
- Power consumption (PoE): 2.1-5.7W
- Dimension (W*D*H*T): 244mm*213mm*185mm*54mm
- Operating humidity: 10~95%
- Operating temperature: -10~50°C (+14~122°F)

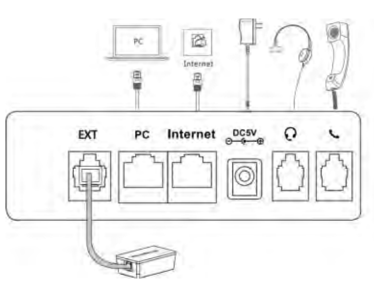
Management
- Configuration: browser/phone/auto-provision
- Auto provision via FTP/TFTP/HTTP/HTTPS for mass deploy
- Autoprovision with PnP
- BroadSoft device management
- Zero-sp-touch, TR-069
- Phone lock for personal privacy protection
- Reset to factory, reboot
- Package tracing export, system log

Network and Security
- SIP v1 (RFC2543), v2 (RFC3261)
- Call server redundancy supported
- NAT traversal: 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
- LLDP/CDP/DHCP VLAN,
- ICE

Package Features
- Package content:
  - Yealink SIP-T29G IP phone
  - Handset with handset cord
  - Ethernet Cable (1.5m CAT5E UTP Cable)
  - Stand
  - Quick Start Guide
  - Power Adapter (Optional)
- Qty/CTN: 5 PCS
- N.W/CTN: 6.6 kg
- G.W/CTN: 7.4 kg
- Giftbox size: 295mm*224mm*115mm
- Carton Size: 602mm*308mm*236mm

Compliance

CE FC RoHS REACH ISO 9001



*Opus: Support 8 kHz (narrowband) and 16 kHz (wideband) sampling rate

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YEALINK

SIP-T46S

A Revolutionary SIP Phone for Enhancing Productivity

The SIP-T46S IP phone is the ultimate communications tool for busy executives and professionals. In addition to offering better overall performance than the T46G, this device has a faster interface with a rich, high-resolution TFT color display. Built with Yealink Optima HD technology and wideband codec of Opus, this device enables crystal clear communications. The SIP-T46S is also built with Gigabit Ethernet technology, for rapid call handling and use with accessories like a Bluetooth USB Dongle and a Wi-Fi USB Dongle. The new T4S series offers the same elegant appearance of the T4 line, but with improvements for greater collaboration.

Features

Optima HD Voice	Opus Codec	Unified Firmware	USB 2.0	HAC	Gigabit	Paperless

HD Audio

Yealink Optima HD Voice technology combines cutting-edge hardware and software with wideband technology for maximum acoustic performance. Being a totally open, highly versatile audio codec, Opus is designed to perform a higher HD audio quality than other wideband codecs in a high network bandwidth. However, if your current network quality is poor, Opus can provide you with a better audio quality than other narrowband codecs. And its hearing aid compatible (HAC) handset helps the hearing impaired hear more clearly.

Easy Customization and High Expandability

The SIP-T46S comes with two Gigabit Ethernet ports, one of which is suitable for Power over Ethernet (PoE). A built-in USB 2.0 port can also be used for Bluetooth, Wi-Fi and USB recording. The SIP-T46S supports up to six expansion models, for up to 240 additional buttons with a screen-based LCD display and LED system. Plus, it has three pages of buttons which can be programmed with up to 27 paperless DSS keys.

Efficient Installation and Provisioning

The Yealink T4S series supports efficient provisioning and effortless mass deployment. Yealink's Redirection and Provisioning Service (RPS) and Boot mechanism helps you carry out the Zero Touch Provisioning without any complex manual settings. This makes the T4S series simple to deploy, easy to maintain and upgrade. Furthermore, a unified firmware and Auto-P template that applies to all T4S phone models (T41S, T42S, T46S and T48S), saves even more time and costs for businesses, and simplifies the management and maintenance.

Secure Transport and Interoperability

The SIP-T46S uses SIP over Transport Layer Security (TLS/SSL), which is the latest network security technology. It's also compatible with leading soft switch suppliers.

- 4.3" 480 x 272-pixel color display with backlight
- Opus* codec support
- USB 2.0
- T4S Auto-P template unified
- T4S firmware unified
- Up to 16 SIP accounts
- Dual-port Gigabit Ethernet
- PoE support
- Paperless label design
- Headset, EHS support
- Wi-Fi via WF40
- Bluetooth via BT40
- USB recording
- Supports expansion modules
- Stand with 2 adjustable angles

Specifications

Audio Features

- HD voice: HD handset, HD speaker
- Hearing aid compatible (HAC) handset
- Codecs: Opus*, G.722, G.711(A/μ), G.723.1, 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

- 16 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
- 3-way conferencing
- Direct IP call without SIP proxy
- Ringtone selection/import/delete
- Set date time manually or automatically
- Dial plan, XML Browser, Action URL/URI
- RTCP-XR (RFC3611), VQ-RTCPXR (RFC6035)
- USB port (2.0 compliant) for: Bluetooth earphone through BT40, Contact synchronization through BT40, Wi-Fi through WF40, USB call recording through USB flash drive
- Enhanced DSS key

Directory

- Local phonebook up to 1000 entries
- Black list
- XML/LDAP remote phonebook
- Smart dialing
- 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, voice mail
- Flexible seating
- Call park, call pickup
- Executive and Assistant
- Centralized call recording
- Visual voice mail
- Call recording

Display and Indicator

- 4.3" 480 x 272-pixel color display with backlight
- 16 bit depth color
- LED for call and message waiting indication
- Dual-color (red or green) illuminated LEDs for line status information
- Wallpaper
- Intuitive user interface with icons and soft keys
- Multilingual user interface
- Caller ID with name and number
- Screensaver
- Power saving

Feature keys

- 10 line keys with LED
- 10 line keys can be programmed up to 27 paperless
- DSS keys (3-page view)
- 7 feature keys: message, headset, hold, mute, transfer, redial, hands free speakerphone
- 4 context-sensitive "soft" keys
- 6 navigation keys
- Volume control keys
- Illuminated mute key
- Illuminated headset key
- Illuminated hands-free speakerphone key

Interface

- Dual-port Gigabit Ethernet
- Power over Ethernet (IEEE 802.3af), class 3
- 1 x USB port (2.0 compliant)
- 1 x RJ9 (4P4C) handset port
- 1 x RJ9 (4P4C) headset port
- 1 x RJ12 (6P6C) EXT port: Supports up to 6 Expansion Modules for an attendant console application

Other Physical Features

- Stand with 2 adjustable angles
- Wall mountable
- External Yealink AC adapter (optional): AC 100~240V input and DC 5V/2A output
- Power consumption (PSU): 1.9-4.0W
- Power consumption (PoE): 2.7-5.5W
- Dimension (W*D*H*T): 244mm*213mm*185mm*54mm
- Operating humidity: 10~95%
- Operating temperature: -10~50°C (+14~122°F)

Management

- Configuration: browser/phone/auto-provision
- Auto provision via FTP/TFTP/HTTP/HTTPS for mass deploy
- Auto-provision with PnP
- 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

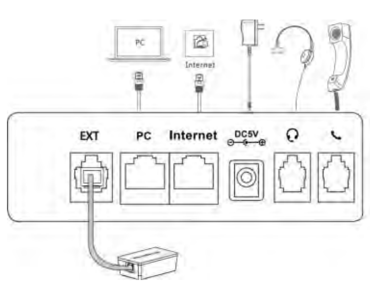
- SIP v1 (RFC2543), v2 (RFC3261)
- Call server redundancy supported
- NAT traversal: STUN mode
- Proxy mode and peer-to-peer SIP link mode
- IP assignment: static/DHCP/PPPoE
- HTTP/HTTPS web server
- Time and date synchronization using SNTP
- UDP/TCP/DNS-SRV(RFC 3263)
- QoS: 802.1p/Q tagging (VLAN), Layer 3 ToS 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, LLDP/CDP/DHCP VLAN, ICE

Package Features

- Package content:
  - Yealink SIP-T46S IP phone
  - Handset with handset cord
  - Ethernet Cable (2m CAT5E FTP Cable)
  - Stand
  - Quick Start Guide
  - Power Adapter (Optional)
  - Wall Mount Bracket (Optional)
- Qty/CTN: 5 PCS
- N.W/CTN: 7.6 kg
- G.W/CTN: 8.4 kg
- Giftbox size: 274mm*255mm*128mm
- Carton size: 660mm*286mm*263mm

Compliance

REACH ISO 9001



*Opus: Support 8 kHz (narrowband) and 16 kHz (wideband) sampling rate

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YEALINK

W52P

Flexible VoIP solution for small businesses

Yealink W52P is a SIP Cordless Phone System designed for small business and SoHo who are looking for immediate cost saving but scalable SIP-based mobile communications system.

Features

HD Voice	Up to 5 users	4 simultaneous calls

Combining the benefits of wireless communication with rich business features of Voice over IP telephony, User can benefit from freedom of movement, lifelike voice communications, multi-tasking convenience, professional features like intercom, transfer, call forward, 3-way conferencing, PoE, etc.

DECT technology

CAT-iq2.0 focuses on high quality Audio VoIP (wideband), as well as low bit - rate data applications, fully backward compatible to DECT GAP

- Exceptional HD sound with wideband technology
- Up to 4 simultaneous external calls
- Up to 2 simultaneous calls per handset
- Up to 5 DECT cordless handsets
- Up to 5 VoIP accounts
- 1.8" color display with intuitive user interface
- 5~10 hours talk time, 100 hours standby time
- Integrated PoE（Class 1）
- Desktop or wall mountable
- Up to 5 DECT Cordless Handsets Per base depending on your needs
- DECT radio coverage up to 50m indoors and 300m outdoors
- Energy-saving ECO features

Specifications

Phone Features
- Up to 4 simultaneous calls
- Up to 2 simultaneous calls per handset
- Up to 5 handsets
- Up to 5 VoIP accounts
- Handset select for receiving call
- Handset and Number select for making call
- Paging, Intercom, Auto answer
- Call hold, Call transfer
- Three-way conferencing
- Switching between calls
- Call waiting, Mute, DND
- Caller ID display, Redial
- Anonymous call, anonymous call rejection
- Call forward (always/busy/no answer)
- Speed dial, Voicemail, Silence
- Message Waiting Indication (MWI)
- Local phonebook for up to 500 entries (store in the base)
- .Remote phonebook
- Phonebook search/import/export
- Call history (outgoing/missed/accepted)
- Direct IP call without SIP proxy
- Reset to factory, Reboot
- Keypad lock, Emergency call
- Dial Plan, Music on hold
- Broadsoft directory, BroadSoft Call Log
- Broadworks feature key synchronization
- Shared Call Appearance (SCA)

Personalization
- 9 ringer melodies
- Screen saver, two kind of color schemes
- Multi-language support

Management
- Auto-provision via FTP/TFTP/HTTP/HTTPS
- Auto-provision with PnP
- Handset upgrade:
  OTA (Over-The-Air)/USB port
- Configuration:
  browser/phone/auto-provision
- Trace package and system log export

Voice and Codecs Features
- Full-duplex speakerphone
- Receiver volume control: 5 steps
- Ringer volume control: 5 steps+off
- Multiple advisory tones
- Acoustic warning for low battery status
- DTMF
- Wideband codec: G.722
- Narrowband codec:
  G.711µ/A, G.723, G.726, G.729, iLBC
- VAD, CNG, AEC, PLC, AJB
- Support VQ-RTCPXR (RFC6035)

Network Features
- SIP v1 (RFC2543), v2 (RFC3261)
- SNTP/NTP
- VLAN (802.1Q and 802.1P)
- 802.1x, LLDP, PPPoE
- STUN Client (NAT Traversal)
- UDP, TCP
- IP assignment: static/DHCP

Security
- Open VPN
- Transport Layer Security (TLS)
- HTTPS (server/client)
- SRTP (RFC3711)
- Digest authentication using MD5
- Secure configuration file via AES encryption
- Admin/Var/User 3-level configuration mode

DECT
- Frequency bands:
  1880 – 1900 MHz (Europe),
  1920 – 1930 MHz (US)
- DECT Standards: CAT-iq2.0

Connectors
- 1 x RJ45 10/100M Ethernet port
- Power over Ethernet (IEEE 802.3af)
- Headset jack (2.5 mm)
- A mini USB Port

Physical Features
- Indoor Range: 20m~50m (The ideal distance is 50m)
- Outdoor Range: 300m (In ideal conditions)
- Standby Time: 100h (In ideal conditions)
- Talk Time: 5~10h (The ideal talk time is 10h)
- 1.8'' 128x160 pixels color display
- Desktop or wall mountable
- LCD backlit, key backlit
- Energy-saving ECO mode/ECO Mode+
- 12 key numerical keypad, 5 navigation keys
  2 softkeys, 6 function keys, 6 shortcut keys
- 3 LEDs on Base:
  1 x power, 1 x Network, 1 x Call
- Base station: DC 5V/600mA Output
- Charger: DC 5V/600mA Output
- Phone size: 144mm x 50mm x 24mm
- Base station size:
  153.5mm x 108.5mm x 45mm
- Operating humidity: 10 ~ 95%
- Operating temperature: —10 ~ +50°C

Package Features
- Package content:
  - W52H Handset
  - Base Station
  - Charger Cradle
  - Two Power Adapters (one is optional)
  - Belt Clip
  - Ethernet Cable
  - Two Rechargeable Batteries
  - Quick Start Guide
- Qty/CNT: 10pcs
- Giftbox size: 214mm*200mm*101mm
- Carton meas: 529mm*410mm*223mm
- N.W: 7.4 kg
- G.W: 8.1 kg

Special Features
- Increase range with up to 5 repeaters

Compliance
REACH ISO 9001

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YEALINK

W56P

Cordless VoIP solution for small businesses

Yealink W56P is the next-generation SIP Cordless Phone System combining quality, reliability and flexibility.

Note: Yealink W56P IP DECT Phone consists of one Base for W52P/W56P and one W56H Handset.

Features

HD Voice	Quick Charge	USB Charger Cradle	3.5mm `Headset Jack

Offering the convenience of cordless with a simple add-on device without losing the SIP features, it brings a seamless call management for our users while “on-the-go.” With more functions, lines and mobility, it empowers users with the convenience of wireless communication along with the widely accepted benefits and feature richness of Voiceover- IP telephony.

In addition, the Yealink W56P delivers the typical benefits of the DECT world such as long talk time, high standby time and superior speech quality, providing excellent value for money and is ideally suited for small and medium-sized businesses.

DECT technology:

CAT-iq2.0 focuses on high quality Audio VoIP (wideband), as well as low bit - rate data applica-tions, fully backward compatible to DECT GAP.

- High-end ID design
- Exceptional HD sound with wideband technology
- Up to 4 simultaneous voice calls
- Up to 5 DECT cordless handsets
- Up to 5 Multiple Lines
- 2.4" 240 x 320 color screen with intuitive user interface
- Up to 30 hours talk time, Up to 400 hours standby time
- Quick charging: 10 mins charge time for 2 hours talk time
- USB Charger Cradle
- Headset connection via 3.5 mm jack
- Charger wall mountable
- New belt clip with better user experience
- Up to 5 DECT Cordless Handsets Per base depending on your needs.
- DECT radio coverage up to 50m indoors and 300m outdoors.
- Energy-saving ECO features

Specifications

Phone Features
- Up to 4 simultaneous calls
- Up to 5 handsets
- Up to 5 VoIP accounts
- Handset select for receiving call
- Handset and Number select for making call
- Paging, intercom, auto answer
- Call hold, call transfer, 3-way conferencing
- Switching between calls
- Call waiting, mute, DND
- Caller ID display, redial
- Anonymous call
- Anonymous call rejection
- Call forward (always/busy/no answer)
- Speed dial, voicemail, silence
- Message Waiting Indication (MWI)
- Local phonebook for up to 500 entries (store in the base)
- Remote phonebook/LDAP
- Phonebook search/import/export
- Call history outgoing/missed/accepted)
- Direct IP call without SIP proxy
- Reset to factory, reboot
- Keypad lock
- Emergency call
- Dial Plan
- Music on hold
- Broadsoft directory
- BroadSoft Call Log
- Broadworks feature key synchronization
- Shared Call Appearance (SCA)

Personalization
- 9 ringer melodies
- Screen saver
- Multi-language support

Management
- Auto-provision via FTP/TFTP/HTTP/HTTPS
- Auto-provision with PnP
- Handset upgrade: OTA (Over-The-Air)
- Configuration:
  browser/phone/auto-provision
- Trace package and system log export

Voice and Codecs Features
- Full-duplex speakerphone
- Hearing Aid Compatibility (HAC) compliant
- Receiver volume control: 5 steps
- Ringer volume control: 5 steps+off
- Multiple advisory tones
- Acoustic warning for low battery status
- DTMF
- Wideband codec: G.722
- Narrowband codec:
  G.711μ/A, G.723, G.726, G.729, iLBC
- VAD, CNG, AEC, PLC, AJB
- Support VQ-RTCPXR (RFC6035)

Network Features
- SIP v1 (RFC2543), v2 (RFC3261)
- SNTP/NTP
- VLAN (802.1Q and 802.1P)
- 802.1x, LLDP, PPPoE
- STUN Client (NAT Traversal)
- UDP, TCP
- IP assignment: static/DHCP
- Support outbound proxy server backup

Security
- Open VPN
- Transport Layer Security (TLS)
- HTTPS (server/client)
- SRTP (RFC3711)
- Digest authentication using MD5
- Secure configuration file via AES encryption
- Support SHA256/SHA512/SHA384
- Admin/Var/User 3-level configuration mode

DECT
- Frequency bands:
  - 1880 – 1900 MHz (Europe),
  - 1920 – 1930 MHz (US),
  - 1900 – 1906 MHz (Thailand)
- DECT Standards: CAT-iq2.0

Connectors
- 1 x RJ45 10/100M Ethernet port
- Power over Ethernet (IEEE 802.3af)
- Headset jack (3.5 mm)

Optional accessory
- Handset protective case

Physical Features
- Indoor Range: 20m~50m (The ideal distance is 50m)
- Outdoor Range: 300m (In ideal conditions)
- Standby Time: 400h (In ideal conditions)
- Talk Time: 30h
- 2.4" 240x320 pixels color display
- Desktop or wall mountable
- LCD backlit, key backlit
- Energy-saving ECO mode/ECO Mode+
- 12 key numerical keypad, 5 navigation keys, 2 softkeys, 6 function keys, 6 shortcut keys
- 3 LEDs on Base:
  1 x power, 1 x Network, 1 x Call
- Base station: DC 5V/600mA Output
- Charger: DC 5V/600mA Output
- Maximum transmission power: 10 milliwatts
- Phone size: 175mm x 53mm x 20.3mm
- Base station size:
  53.5mm x 108.5mm x 45mm
- Operating humidity: 10 ~ 95%
- Operating temperature: -10 ~ +50°C

Package Features
- Package content:
  - W56H Handset
  - Base for W52P/W56P
  - Belt Clip
  - Rechargeable Lithium Battery
  - USB Charger Cradle
  - Two Power Adapters
  - Ethernet Cable
  - Quick Start Guide
- Qty/CNT: 10pcs
- Giftbox size: 205mm*196mm*95mm
- Carton meas: 512mm*414mm*213mm
- N.W: 8.1kg
- G.W: 9.0kg

Special Features
- Increase range with up to 5 repeaters

Compliance
CE FC RoHS REACH ISO 9001

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YEALINK

W60P

High-performance DECT IP phone system with user-centric design

The Yealink W60P, being a high-performance SIP cordless phone system, is the ideal solution for small and medium-sized businesses. Paring with up to a total of 8 Yealink W52H/W56H DECT handsets, it allows you enjoy superb mobility and efficient flexibility immediately as well as significantly eliminates additional wiring troubles and charges. To provide a better and higher performance, this DECT IP phone not only supports up to 8 VoIP accounts and 8 concurrent calls, but also speeds up its startup and signal connection, slashes its upgrade downtime as well.

Features

Eight Concurrent Calls	TLS & SRTP security encryption	Backup System	Centralized Deployment	Quick Charge	Long Standby Time

By supporting Opus codec, W60P consistently delivers excellent and professional audio quality in both high-bandwidth and poor network conditions, comparing with other wideband or narrowband audio codecs. Offering the convenience of cordless with a simple add-on device without losing the SIP features, it brings a seamless call management for our users while “on-the-go.” Owning more functions, lines and mobility, it empowers users with the convenience of wireless communication along with the widely accepted benefits and feature richness of Voice-over-IP telephony.

The Yealink DECT IP phone W60P supports efficient provisioning and effortless mass deployment with Yealink’s Redirection and Provisioning Service (RPS) and Boot mechanism to help you realize the Zero Touch Provisioning without any complex manual settings, which makes it simple to deploy, easy to maintain and upgrade, saving even more time and IT costs for businesses.

DECT technology:

CAT-iq2.0 focuses on high quality Audio VoIP (wideband), as well as low bit - rate data applications, fully backward compatible to DECT GAP.

- Up to 8 DECT cordless handsets per base depending on your needs
- DECT radio coverage up to 50m indoors and 300m outdoors
- Energy-saving ECO features
- High-performance SIP cordless phone system
- 2.4" 240 x 320 color screen with intuitive user interface
- Up to 8 concurrent calls
- Up to 8 DECT cordless handsets
- Up to 8 VoIP accounts
- Support Opus audio codec
- Up to 30-hour talk time
- Up to 400-hour standby time
- Quick charging: 10-min charge time for 2-hour talk time
- TLS and SRTP security encryption
- Noise Reduction System
- Headset connection via 3.5 mm jack
- Charger wall mountable

Specifications

Audio Features

- Full-duplex speakerphone
- Hearing Aid Compatibility (HAC) compliant
- Receiver volume control: 5 steps
- Ringer volume control: 5 steps+off
- Multiple advisory tones
- Acoustic warning for low battery status
- DTMF
- Wideband codec: Opus, AMR-WB (optional), G.722
- Narrowband codec:
  PCMU, PCMA, G.726, G.729, iLBC VAD, CNG, AGC, PLC, AJB
- AEC (supported by W52H and W56H)
- Support VQ-RTCPXR (RFC6035)

Phone Features

- Up to 8 simultaneous calls
- Up to 8 handsets
- Up to 8 VoIP accounts
- Up to 2 simultaneous calls per handset
- Up to 5 repeaters per base station
- Handset selection for receiving call
- Handset and number selection for placing call
- Paging, intercom, auto answer, dial plan
- Call hold, call transfer, 3-way conference
- Switching between calls
- Call waiting, mute, silence, DND
- Caller ID with name and number
- Anonymous call, Anonymous call rejection
- Call forward (always/busy/no answer)
- Speed dial, voicemail, redial
- Message Waiting Indication (MWI)
- Local phonebook for up to 500 entries (store in the base)
- Remote phonebook/LDAP
- Phonebook search/import/export
- Call history (all/missed/placed/received)
- Direct IP call without SIP proxy
- Reset to factory, reboot
- Keypad lock, emergency call
- Broadsoft directory, BroadSoft call log
- Broadworks feature key synchronization
- Shared Call Appearance (SCA)

Personalization

- 9 ringer melodies
- Screen saver
- Multilingual user interface

Management

- Auto-provision via TFTP/FTP/HTTP/HTTPS/RPS
- Auto-provision with PnP
- Handset upgrade: OTA (Over-The-Air)/USB port
- Configuration: browser/phone/auto-provision
- Trace package and system log export

Network Features

- SIP v1 (RFC2543), v2 (RFC3261)
- SNTP/NTP
- VLAN (802.1Q and 802.1P)
- 802.1x, LLDP, PPPoE
- STUN Client (NAT Traversal)
- UDP/TCP/TLS
- IP assignment: static/DHCP
- Support outbound proxy server backup

Security

- Open VPN
- Transport Layer Security (TLS)
- HTTPS (server/client), SRTP (RFC3711)
- Digest authentication using MD5
- Secure configuration file via AES encryption
- Support SHA256/SHA512/SHA384
- Three-level configuration mode: Admin/Var/User

DECT

- Frequency bands:
  1880 – 1900 MHz (Europe), 1920 – 1930 MHz (US)
- DECT Standards: CAT-iq2.0

Interface

- 1 x RJ45 10/100M Ethernet port
- Power over Ethernet (IEEE 802.3af), Class 1
- Headset jack (3.5 mm)

Physical Features

- Indoor Range: 20m~50m (The ideal distance is 50m)
- Outdoor Range: 300m (in ideal conditions)
- Standby Time: 400 hours (in ideal conditions)
- Talk Time: 30 hours
- 2.4'' 240x320 pixels color display
- Desktop or wall mountable
- LCD backlit, key backlit
- Energy-saving ECO mode/ECO Mode+
- 12 key numerical keypad, 5 navigation keys,
- 2 softkeys, 6 function keys, 6 shortcut keys
- Three LED indicators on W60B:
  1 x Registration LED
  1 x Network Status LED
  1 x Power Indicator LED
- External Yealink AC adapter:
  AC 100-240V Input and DC 5V/600mA Output
- Phone size: 175mm x 53mm x 20.3mm
- Base station size: 130mm x 100mm x 25.1mm
- Operating humidity: 10~95%
- Operating temperature: -10~+50°C (+14~122°F)

Package Features

- Package content:
  - W56H Handset
  - W60B Base Station
  - Base Stand
  - USB Charger Cradle
  - Two Power Adapters
  - Ethernet Cable
  - Belt Clip
  - Rechargeable Battery
  - Quick Start Guide
  - Handset Protective Case (optional)
- Qty/CNT: 10 PCS
- Giftbox size: 205mm*196mm*95mm
- Carton meas: 495mm*406mm*223mm
- N.W: 7.5 kg
- G.W: 8.3 kg

Compliance

REACH ISO 9001