



UTG7100-IP/IPE/IPO/IPEO

Multi-Function VoIP SIP Telephone

Cost effective VoIP phone for your home or office



FEATURES

- Supports SIP protocol that compatible with most IP PBX and can be register up to 3 SIP accounts
- Provides optional features of PoE, FXO, and Headset
- Enhanced VPN security features to protect LAN
- Supports 3-way call, 140 phone book numbers, four type ringer tones and Call Waiting function
- Supports 64 entries each for incoming & dialed call
- Supports multiple voice codec G.711, G.723.1, G.726, and G.729a/b codec algorithm
- Call functions for all forward, busy forward, no answer forward and unconditional status
- Adjustable Ringer, handset and speaker volume
- Upgrade program by FTP / TFTP / HTTP modes
- Supports Asterisk Open Source IP PBX
- Supports Plan setting, Hotline setting, Alarm setting
- Supports Block setting for AI I/ By Time / Duration
- LCD display for dial data, caller name, caller number
- Supports Auto Provision by FTP / TFTP / HTTP

Package Contents

- One VoIP SIP Telephone
- One User's Manual CD-ROM
- One RJ-45 Network Cable
- One Base Stand and one Handset RJ-11 Coil Cable
- One Power Adapter(PoE Model Optional)

OVERVIEW

EUSSO UTG7100 VoIP Telephone series is SIP 2.0 (RFC3261) compliant. It is feature-rich, toll-quality, cost-effective, and standard-compliant VoIP telephone that can be easily plugged to network Cat. 5 cable directly from ADSL router or switch at your home or office. UTG7100 Series is compatible with most VoIP service providers and it operates like a regular telephone. With a VoIP phone service or connect to VoIP Gateway Router in office, this VoIP phone enables significant cost savings when making international or long distance phone calls.

UTG7100 series is self-contained, service-integrated and intelligent phone features device. With optional functions of 802.3af Power over Ethernet (PoE) or FXO port, it provides flexibility installation alternatively for different VoIP application environments.

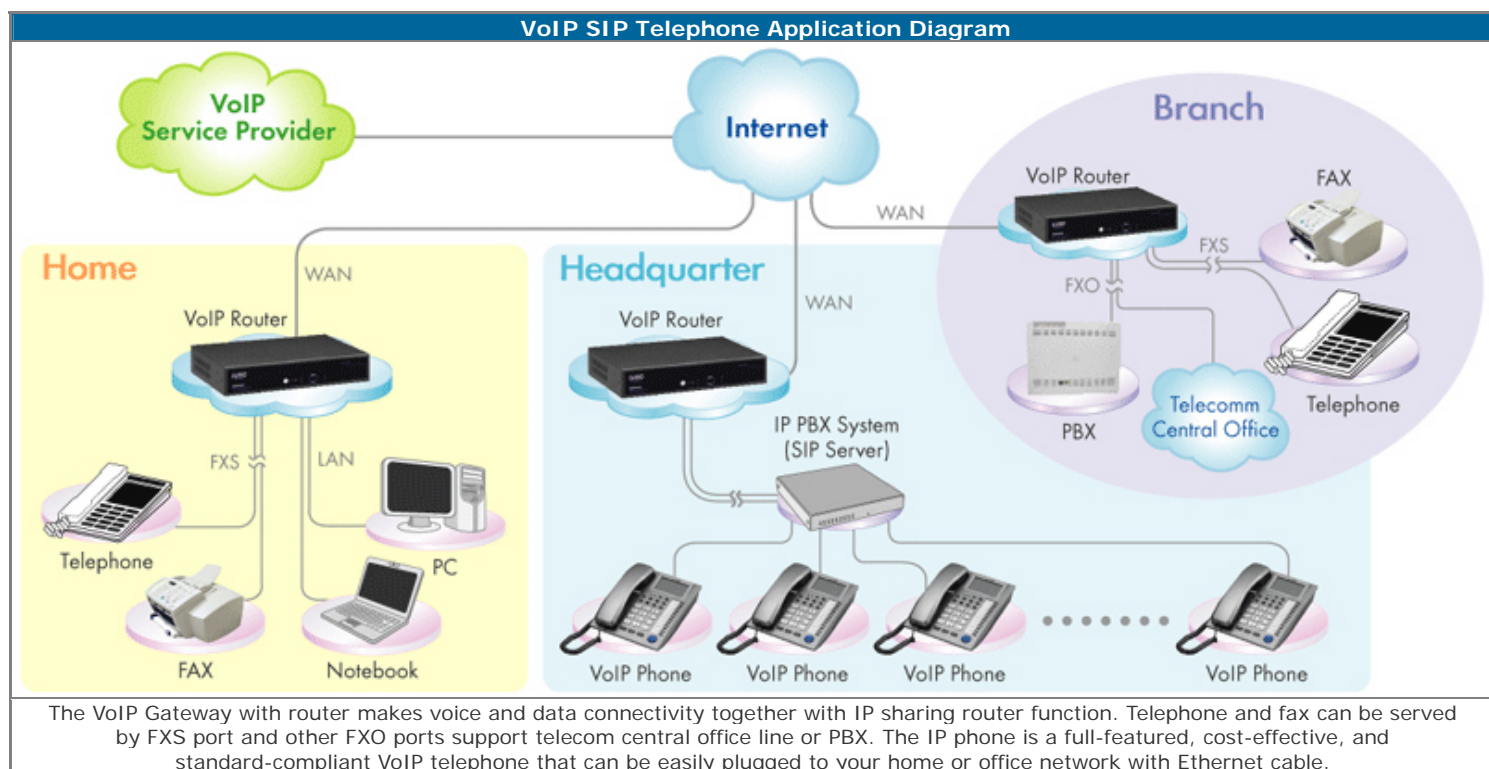
EUSSO SIP IP Phone included an integrated speakerphone, as well as a 2 x 16 LCD screen to display caller ID, number dialed, speed dial entries, and address book entries, those call information. The convenience 5 memory function keys give you instant access to your most frequently called numbers. The VoIP SIP Phone also supports rich features such as redial, mute, hold, transfer, 3-way conference call, call waiting, call forwarding, message light, and inserts dials.

UTG7100 series provides Quality of Service (QoS) functions to deliver clear, high-quality voice communication in a variety of network condition. With QoS, voice packets get a higher priority over data packets. Other features to improve call quality included echo cancellation, comfort noise generation, (CNG), Voice activity detection (VAD), Adaptive Jitter Buffer and packet Loss Compensation technology.

SPECIFICATIONS

VoIP Models Specifications				
Model	UTG7100-IP	UTG7100-IPE	UTG7100-IPO	UTG7100-IPEO
Description	SIP Phone	SIP Phone with PoE	SIP Phone with FXO	SIP Phone with PoE & FXO
VoIP Standard	SIP	SIP	SIP	SIP
Headset	Optional	Optional	Optional	Optional
Network Protocols	SIP v1 (RFC2543), v2 (RFC3261) IP/TCP/UDP/RTP/RTCP IP/ICMP/ARP/RARP/SNTP TFTP Client/DHCP Client/PPPoE Client Telnet/Http Server DNS Client VLAN Setting, DMZ Setting, MAC Clone Setting Virtual Server			
Codec	G.711: 64k bit/s (PCM) G.723.1: 6.3k / 5.3k bit/s G.726: 16k / 24k / 32K / 40k / bit/s (ADPCM) G.729A: 8k bit/s (CS-ACELP) G.729B adds VAD & CNG to G.729			
Voice Quality	VAD: Voice activity detection CNG: Comfort noise generation LEC: Line echo canceller Packet Loss Compensation Adaptive Jitter Buffer			
Call Functions	Call Hold Call Waiting Call Forward Caller ID 3-Way Conference			
Phone Functions	Volume Adjustment Speed dial, Phone Book Flash Speaker Phone CPC (Calling Party Control)			
Tone	Ring Tone Ring Back Tone Dial Tone Busy Tone User Programming Tone			
IP Assignment	Static IP DHCP PPPoE			
Security	HTTP 1.1 basic/digest authentication for Web setup MD5 for SIP authentication (RFC2069/RFC2617)			
DTMF Functions	In-Band DTMF Out-of-Band DTMF SIP Info			
Advanced Features	SIP Server: registrar Server (three SIP accounts); Outbound Proxy QoS: ToS Field Configuration: Web Browser, Console/Telnet, Keypad NAT Traversal: STUN			

VoIP SIP Telephone Model Hardware Specifications				
Model	UTG7100-IP	UTG7100-IPE	UTG7100-IPO	UTG7100-IPEO
Description	SIP Phone	SIP Phone with PoE	SIP Phone with FXO	SIP Phone with PoE & FXO
LAN Interface	1 x 10Mbps/100Mbps RJ-45 (PoE: Power over Ethernet 802.3af for PoE Models)			
WAN Interface	1 x 10Mbps/100Mbps RJ-45 Port			
LCD Display	2 x 16 characters*			
Function Keys	26 function keys besides standard keys 0-9, #, *			
Speaker	Full Duplex hand free speaker phone			
Installation	Desktop, Wall Mountable			
Power Supply	DC12V, 1A			
Temperature	Operating: 0° ~ 40°C; Storage: -10 ~ 50 °C			
Humidity	10 ~ 90% (Non-Condensing)			
Dimension	183 (L) x 225 (W) x 120 (D) mm			
Weight	680 g			
Certification	CE, FCC			



ORDERING INFORMATION

Model	Description
UTG7100-IP	VoIP SIP Phone
UTG7100-IPE	VoIP SIP Phone with PoE Port
UTG7100-IPO	VoIP SIP Phone with FXO Port
UTG7100-IPEO	VoIP SIP Phone with PoE & FXO Ports

EUSSO Technologies, Inc. is a dedicated data communication and networking company. With professional experiences in design, production, marketing and service support, we deliver the full range networking products including Gigabit Ethernet, Fiber Optic, Wireless LAN, Switches, Hubs, LAN cards, PCMCIA adapters, Converter, Transceivers. As well as Internet Telephony Gateway, Print Servers, Internet Sharing Servers and many others.

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