Tadiran Telecom

Endpoints Specifications

AEONIX EVERLASTING

Unified Communications
& Collaboration Solution

Aeonix Endpoints

Tadiran offers a broad range of SIP phones, softphones, wireless phones, and attendant consoles.

The phones offer both executives and office workers superior high definition voice quality in every call.

Users can personalize their settings directly via the phone or via a user-friendly, web-based administration, as well as benefit from both local and centralized phone directories.

Aeonix offers a range of IP phones from entry-level to mid-range for remote workers and executives, with a wide range of capabilities including graphic display and Video Conferencing.

Aeonix preconfigured phones allow for smooth and easy implementation.

Aeonix's adherence to SIP open standards allows companies to choose their own SIP compliant phones (BYOD), including smartphones

Tadiran T320 (P) – HD 2 Line PoE and non-PoE Phone

Tadiran has expanded its line-up of IP phones with a new entry-level product – the SIP-T320 (P). This phone is equipped with a TI TITAN chipset with a 2x15 character LCD. It offers two VoIP accounts, high-definition audio, a broad range of voice codecs and security protection for privacy. It also provides rich features that include headset, PoE and PnP Auto-provisioning. It allows users to make calls in a simple, convenient and reliable manner while meeting all basic business feature requirements. The SIP-T320P is easy to install and is a perfect fit for both corporate office and residential users.

Features:
- TI TITAN chipset and TI voice engine
- 3-line LCD (2 x 15 characters and an icon line)
- 2 VoIP accounts
- HD Voice: HD Codec, HD Handset, HD Speaker
- 31 keys including 9 function keys
- Voicemail, Intercom
- Localized language, Local phonebook
- FTP/TFTP/HTTP, PnP Auto-provisioning
- SRTP/HTTPS/TLS, VLAN, QoS
- PoE, Headset, Wall-Mounted

Phone Features:
- Hotline, emergency call waiting, call transfer, call forward
- Hold, mute, flash, auto-answer, redial, 3-way conference, DND, speed dial
- Phonebook import/export, call history
- Volume adjustment, ring tone selection
- Tone scheme, System log
- Multi-language (more than 20)

IP PBX System Integration:
- Dial plan, dial-now
- Voicemail, MWI
- Intercom, Paging
- Call park, call pickup
- Distinctive ringtone

Codecs and Voice Features:
- Wideband codec: G.722
- Narrowband codec: G.711, G.723.1, G.726, G.729AB
- VAD, CNG, AEC, PLC, AJB, AGC
- Full-duplex hands-free speakerphone with AEC

Network Features:
- SIP v1 (RFC2543), v2 (RFC3261)
- NAT Traversal: STUN mode
- DTMF: In-Band, RFC2833, SIP Info
- Proxy mode and peer-to-peer SIP link mode
- IP Assignment: Static/DHCP/PPPoE
- TFTP/DHCP/PPPoE client
- Telnet/HTTP/HTTPS server
- DNS client
- NAT/DHCP server

Management:
- FTP/TFTP/HTTP/PnP auto-provisioning
- Configuration: browser/phone/auto-provisioning
- Direct IP call without SIP proxy
- Dial number and URL via SIP server
- Supports IPV6

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

Physical Features:
- 4 LEDs: 1xpower, 2xline, 1xmessage
- 1xRJ9 handset port / 1xRJ9 headset port
- 2xRJ45 10/100M Ethernet ports
- Power adapter: AC 100~240V input / DC 5V/1.2A Output
- Power over Ethernet (IEEE 802.3af)
- Power consumption: 1.4-2.6W
- Net weight: 0.77kg
- Dimensions (mm): 185 x 200 x 90
- Operating humidity: 10~95%
- Storage temperature: up to 60°C

Tadiran T322 – SIP HD 3 Line Graphic Display PoE Phone

The Tadiran SIP-T322P features an intuitive user-interface and enhanced functionality to make user interaction and operation very easy and efficient. The TI TITAN chipset and TI leading VoIP voice engine provide enhanced high-definition audio, outsourced management options, flexible installation and the addition of third-party communications applications. As a cost-effective IP solution, the T322P helps users to streamline their business systems and delivers a powerful, secure and consistent communication experience for both small and large office environments.

Features:
- TI TITAN chipset and TI voice engine
- 132x64 graphic LCD
- 3 VoIP accounts
- HD Voice: HD Codec, HD Handset, HD Speaker
- 32 keys including 4 soft keys
- SMS, Voicemail, Intercom
- Localized language, XML phonebook
- FTP/TFTP/HTTP, PnP Auto-provisioning
- SRTP/HTTPS/TLS, VLAN, QoS

Phone Features:
- Hotline, emergency call
- Call waiting, call transfer, call forward
- Hold, mute, flash, auto-answer, redial
- 3-way conference, DND, speed dial
- Phonebook (each record with 3 phone numbers, 300 entries), black list
- XML Phonebook search/import/export
- Lists of missed, received, dialed and forwarded calls (100 entries in all)
- Volume adjustment, ring tone selection
- Tone scheme, System log
- Multi-language (more than 20)

IP PBX System Integration:
- Dial plan, dial-now
- SMS, Voicemail, MWI
- Intercom
- Call park, call pickup
- Distinctive ringtone

Codecs and Voice Features:
- Wideband codec: G.722
- Narrowband codec: G.711, G.723.1, G.726, G.729AB
- VAD, CNG, AEC, PLC, AJB, AGC
- Full-duplex speakerphone with AEC

Management:
- FTP/TFTP/HTTP/PnP auto-provisioning
- Configuration: browser/phone/auto-provisioning
- Direct IP call without SIP proxy
- Dial number and URL via SIP server
- Supports IPV6

Network Features:
- SIP v1 (RFC2543), v2 (RFC3261)
- NAT Traversal: STUN mode
- DTMF: In-Band, RFC2833, SIP Info
- Proxy mode and peer-to-peer SIP link mode
- IP Assignment: Static/DHCP/PPPoE
- TFTP/DHCP/PPPoE client
- Telnet/HTTP/HTTPS server
- DNS client
- NAT/DHCP server

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

Physical Features:
- 5 LEDs: 1xpower, 3xline, 1xmessage
- 1xRJ9 handset port / 1xRJ9 headset port
- 2xRJ45 10/100M Ethernet ports
- PoE, Headset, Wall-Mounted
- Power adapter: AC 100~240V input / DC 5V/1.2A Output
- Power over Ethernet (IEEE 802.3af)
- Power consumption: 1.4-2.6W
- Net weight: 0.77kg
- Dimension (mm): 185 x 200 x 90
- Operating humidity: 10~95%
- Storage temperature: up to 60°C

Tadiran T328 – SIP HD 6 Line Graphic Display PoE 10 DSS Phone

The Tadiran SIP-T328P represents the next generation VoIP phone designed for business users who need rich telephony features, a friendly user-interface and superb voice quality. Equipped with the TI TITAN chipset, it offers high-definition voice quality through a TI voice engine, HD handset, HD speaker and HD codec (G.722). The large, high-resolution graphical display, combined with up to 48 keys, guarantees an excellent user experience in terms of configuration options, making calls, and accessing the express XML browser. To ensure that your audio data remains confidential, the Tadiran SIP-T328P also supports security standards TLS, SRTP, HTTPS, 802.1x, Open VPN and AES encryption. These features guard against electronic eavesdropping and data theft. The T328P supports up to six X39 expansion modules.

Features:
- TI TITAN chipset and TI voice engine
- 320x160 graphic LCD with 4-level grayscales
- 6 VoIP accounts
- HD Voice: HD Codec, HD Handset, HD Speaker
- 48 keys including 16 programmable keys
- BLF/BLA, SMS, Voicemail, Intercom
- Localized language, XML phonebook
- FTP/TFTP/HTTP, PnP Auto-provisioning
- SRTP/HTTPS/TLS, VLAN, QoS
- PoE, Headset, 2xRJ45, Expansion module

Phone Features:
- Hotline, emergency call
- Call waiting, call transfer, call forward
- Hold, mute, flash, auto-answer, redial
- 3-way conference, DND, speed dial
- XML Phonebook search/import/export
- Black list, call history (100 entries)
- Volume adjustment, ring tone selection
- Tone scheme, System log
- Multi-language (more than 20)
- Supports up to 6 expansion modules

IP PBX System Integration:
- Dial plan, dial-now
- SMS, Voicemail, MWI
- BLF/BLA, intercom, paging
- Call park, call pickup
- Distinctive ringtone

Codecs and Voice Features:
- Wideband codec: G.722
- Narrowband codec: G.711, G.723.1, G.726, G.729AB
- VAD, CNG, AEC, PLC, AJB, AGC
- Full-duplex speakerphone with AEC

Management:
- FTP/TFTP/HTTP/PnP auto-provisioning
- Configuration: browser/phone/auto-provisioning
- Direct IP call without SIP proxy
- Dial number and URL via SIP server
- Supports IPV6

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

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

Physical Features:
- 1xRJ9 (4P4C) handset port / 1xRJ9 (4P4C) headset port
- 2xRJ45 10/100M Ethernet ports
- 1XRJ12 (6P6C) expansion module port
- Power adapter: AC 100~240V input / DC 5V/1.2A output
- Power over Ethernet (IEEE 802.3af)
- Power consumption: 1.6-2.6W
- Net weight: 1.05KG
- Dimension (mm): 273x204x42
- Operating humidity: 10~95%
- Storage temperature: up to 60°C

T32G – SIP HD 3 Line Color Display PoE 1G Ethernet Phone

The SIP-T32G IP Phone is one of the most recent innovations for managers with demanding integrated communication needs. It has been designed as a future-proofed infrastructure investment that provides a seamless migration to modern GigE-based networks. Dual-port Gigabit Ethernet connections ensure flexible installation options and lower cabling costs. With its large, high-resolution TFT color display, the T32G IP Phone offers a brilliant presentation of caller information, with a user interface designed for clarity and intuitive operation. Equipped with the TI Aries chipset, HD handset, HD speaker and HD codec (G.722), the T32G IP Phone gives an unrivalled, lifelike audio experience. To ensure that your audio data remains confidential, it also supports security standards TLS, SRTP, HTTPS, 802.1x, Open VPN and AES encryption. These features guard against electronic eavesdropping and data theft.

Features:
- TI Aries chipset and TI voice engine
- Dual-port Gigabit Ethernet (Router & Switch)
- Supports IPV6
- Power over Ethernet
- 3" TFT-LCD, 400 x 240 pixel, 262K colors
- Color Picture Caller-ID, Screensaver, Wallpaper
- Convenient and intuitive user structure
- Support HD wideband codec: G.722
- HD Voice: HD Codec, HD speaker, HD handset
- Full-duplex speakerphone with AEC
- 3 VoIP accounts
- Auto-provisioning via TFTP / FTP / HTTP /HTTPS / PNP
- SRTP / HTTPS / TLS, VLAN, QoS, Open VPN
- National Language Support
- BLF/BLA, Hot-desking, OpenVPN
- Broad and Deep Interoperability
- Inherits all the features of T2x V60

Phone Features:
- Codecs and Voice Features
- Physical Features
- Hotline, Emergency call
- Call hold, Call waiting, Call forward, Call return
- Call transfer (Blind/Semi-attended/Attended)
- Caller ID display, Redial, Mute, DND
- Auto-answer, 3-way conferencing
- Speed dial, SMS, Voicemail
- Message Waiting Indication (MWI) LED
- Tone scheme, Volume control
- Direct IP call without SIP proxy
- Ring tone selection/Import/Delete
- Phonebook (1000 entries), Black list
- Call history: Dialed/Received/Missed/Forwarded
- Menu-driven user interface
- Localized language and input method
- Soft keys programmable

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

IP PBX System Integration:
- Busy lamp field (BLF), BLF list, (BLA)
- DND & Forward synchronization
- Intercom, Paging, Music on hold
- Call park, Call pickup
- Call recording, Call completion
- Group listening, Group pickup
- Anonymous call, Anonymous call rejection
- Network conference
- Distinctive ringtone
- Dial Plan, Dial-now
- Wideband codec: G.722
- Narrowband codec: G.711μ/A, G.723.1,G.726, G.729AB
- VAD, CNG, AEC, PLC, AJB, AGC
- Full-duplex speakerphone with AEC

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

Network Features:

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

Security:

- Open VPN, 802.1x, VLAN QoS (802.1pq), LLDP
- Transport Layer Security (TLS)
- HTTPS (server/client), SRTP (RFC3711)
- Digest authentication using MD5/MD5-sess
- Secure configuration file via AES encryption
- Phone lock for personal privacy protection
- Admin/VAR/User 3-level configuration mode
- 2xRJ45 10/100/1000Mbps Ethernet ports
- 3" TFT-LCD, 400 x 240 pixel, 262K colors
- 32 keys including 3 programmable keys
- 5 LEDs: 1 x power, 3 x line, 1 x message
- Wall-mountable
- 1xRJ9 (4P4C) handset port / 1xRJ9 (4P4C) headset port
- Power adapter: AC 100~240V input and DC 5V/2A output
- Power over Ethernet (IEEE 802.3af)
- Power consumption: 4.0W
- Net weight: 0.77kg
- Dimensions (mm): 185 x 200 x 90
- Operating humidity: 10~95%
- Storage temperature: up to 60°C

T38G – SIP HD 6 Line Color Display PoE 1G Ethernet 10 DSS Phone

The SIP-T38G Series IP Phone is one of the most recent innovations for managers with demanding integrated communication needs. It has been designed as a future-proofed infrastructure investment that provides a seamless migration to modern GigE-based networks. Dual-port Gigabit Ethernet connections ensure flexible installation options and lower cabling costs. With its large, high-resolution TFT color display, the T38G IP Phone offers a brilliant presentation of caller information, with a user interface designed for clarity and intuitive operation. Equipped with the TI Aries chipset, HD handset, HD speaker and HD codec (G.722), it gives an unrivalled, lifelike audio experience, plus a host of telephony features to increase efficiency. The T38P supports up to six X39 expansion modules.

Features:
- TI Aries chipset and TI voice engine
- Dual-port Gigabit Ethernet (Router & Switch)
- 6 VoIP accounts
- Supports IPV6
- Power over Ethernet
- 4.3" TFT-LCD, 480 x 272 pixel, 16.7M colors
- Color Picture Caller-ID, Screensaver, Wallpaper
- Convenient and intuitive user structure
- Headset, EHS support, LCD Expansion module

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

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

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

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

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

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

Physical Features:

- 2xRJ45 10/100/1000Mbps Ethernet ports
- 48 keys including 16 programmable keys
- 1xRJ9 (4P4C) handset and headset ports
- 1XRJ12 (6P6C) EXT port
- Power adapter: AC 100~240V input and
- DC 5V/2A output
- Power over Ethernet (IEEE 802.3af)
- Power consumption approx.: 4.6W
- Net weight: 1.05KG
- Dimensions (mm): 273x204x42
- Operating humidity: 10~95%
- Storage temperature: up to 60°C

EXP 39 20 DSS LCD Label Expansion Module

The EXP39 LCD Expansion Module has been designed to improve the power and flexibility of the advanced SIP T328P Tadiran IP phones. It features a 160x320 graphic LCD and 20 physical keys, each with a dual-color LED. These show very clearly which function has been selected and make operation very easy. What's more, you can add 20 additional keys via a page switch, bringing the number of programmable keys up to 40. Each programmable key supports functions that include speed dialing, plus BLF/BLA, intercom, call forward/transfer/hold/park/pickup/return. This allows up to one or two EXP39 units to be daisy-chained together. The Tadiran EXP39 is ideal for receptionists, administrative assistants, call center agents, power-users, and executives who need to monitor and manage a large volume of calls on a regular basis. Features include:

Features:

- Rich visual experience with 160x320 graphic LCD
- 20 physical keys each with a dual-color LED
- 20 additional keys through page switch
- Daisy-chain 2 modules up to 50 keys
- Supports BLF/BLA, speed dialing, call pickup

Specifications:

- 160x320 graphic LCD with 16-level gray scales
- Supports up to 6 modules daisy-chain
- Different icons for each function shown on the LCD
- Dual-color LED for status information
- Intercom, hold, transfer, voicemail, forward, DND, etc.
- Applies to Tadiran IP Phone T328P and SIP-T38G
- Expansion module is powered by the host phone

Physical Features:

- 2xRJ12 (6P6C) ports for data in and out
- Power adapter: AC 100~240V input / DC 5V/3A output
- Weight: 290g
- Dimensions (mm): 145x182x100
- Operating humidity: 10~90%
- Storage temperature: max 40°C

VP530 – SIP HD Video Touch Screen Phone

The Tadiran VP530 is a further innovation of our advanced, executive-level IP Video Phone. With integrated audio, video and applications, the VP530 is a very powerful business video phone. Its large display and easy use make it an ideal all-in-one tool for today's busy business leaders, regardless of where they happen to be. With excellent performance and rich business features, the VP530 offers an all-round, face-to-face network experience that allows users to interact and communicate like never before. Features include: Integrated intelligent features that make it exceptionally easy to operate; Maximum productivity for managers and executives; Efficient person-to-person contact by video call or video conferencing, leading to lower travel costs and faster decision-making; Easy to install and simple to administer, upgrade and maintain; A reduced carbon footprint with energy-saving PoE for a greener world; Highly customizable and expandable service features.

Features:

- TI DaVinci dual-core chipset
- 7" 800x480 digital TFT-LCD, resistive touch screen
- HD Voice, full-duplex speakerphone
- 4 VoIP accounts, 3-way video conferencing
- BLF, Intercom, 18 one-touch soft DSS keys
- Total directory solution
- Door phone application
- Supports IPV6

Phone Features:

- Video Features
- Video codec: H.264 and H.263
- Image codec: JPEG, GIF, PNG, BMP
- Video capacity: up to D1 (720x480) @ 30 fps
- Video call format: CIF/QCIF
- Bandwidth selection: 128kbps~1Mbps
- Frame rate selection: 10~30fps
- Adaptive bandwidth adjustment
- I-frame adjustable
- Picture-in-Picture (PIP)
- Full screen for remote side
- Video control of local side
- Door phone application

Audio Features:

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

IP PBX System Integration:

- Busy lamp field (BLF), BLF list
- Message waiting indicator (MWI)
- Intercom, Music on hold
- Call park, Call pickup
- Anonymous call, Anonymous call rejection
- DND & forward synchronization
- Dial Plan, Dial-now

Additional Phone Features:

- Video/Voice call
- 18 one-touch soft DSS keys, Speed dial
- Dial/answer call type selection (video or voice)
- Call forward, Call waiting, Call transfer, Call hold
- Mute, Redial, Auto answer
- DND, Caller ID display, Call history
- Voice mail, Local 3-way audio conference
- Direct IP call without SIP proxy
- Phonebook with contact picture
- Group manager, Black list
- XML/LDAP remote phonebook
- New message and missed call notification
- Volume control, Ring tone selection
- Wallpaper
- Set date time manually or automatically
- National language selection
- Icon-driven menu

Network and Security:

- SIP v1 (RFC2543), v2 (RFC3261)
- NAT transverse: STUN mode
- Proxy mode and peer-to-peer SIP link mode
- IP assignment: static/DHCP/PPPoE
- HTTP/HTTPS web server
- Time and date synchronization using SNTP
- UDP/TCP/DNS-SRV(RFC 3263)
- QoS: 802.1p/Q tagging (VLAN), Layer 3 ToS, and DSCP
- SRTP for voice and video
- Transport Layer Security (TLS)
- HTTPS certificate manager
- AES encryption for configuration file
- Digest authentication using MD5/MD5-sess

Management:

- Configuration: browser/phone/auto-provisioning
- Auto provisioning via FTP/TFTP/HTTP/HTTPS
- Auto-provisioning with PnP
- TR069 Protocol optional
- Zero-sp-touch
- Reset to factory, Reboot
- Package tracing export, System log

Physical Features:

- Rotatable CMOS sensor camera with 2M pixels
- 128MB flash and 256MB DDR2 memory
- 27 keys including 4 soft keys
- 6 feature keys: Mute/Camera/Phonebook/Transfer/Redial/ Hands-free
- 2xLEDs for power and status indication
- 2xRJ45 Ethernet 10/100M ports
- 1xUSB2.0 port, 1xSD card slot
- 2.5mm headset port
- Power adapter: AC 100~240V input / DC 5V/3A output
- Power over Ethernet (PoE) optional: IEEE 802.3af, Class 0
- Power consumption: 4~10W
- Net weight: 1.2Kg
- Dimensions (mm): 286x89x45
- Operating humidity: 10~95%
- Storage temperature: up to 60°C

EXP 39 20 DSS LCD Label Expansion Module

The EXP39 LCD Expansion Module has been designed to improve the power and flexibility of the advanced SIP T328P Tadiran IP phones. It features a 160x320 graphic LCD and 20 physical keys, each with a dual-color LED. These show very clearly which function has been selected and make operation very easy. What's more, you can add 20 additional keys via a page switch, bringing the number of programmable keys up to 40. Each programmable key supports functions that include speed dialing, plus BLF/BLA, intercom, call forward/transfer/hold/park/pickup/return. This allows up to one or two EXP39 units to be daisy-chained together. The Tadiran EXP39 is ideal for receptionists, administrative assistants, call center agents, power-users, and executives who need to monitor and manage a large volume of calls on a regular basis. Features include:

Features:

- Rich visual experience with 160x320 graphic LCD
- 20 physical keys each with a dual-color LED
- 20 additional keys through page switch
- Daisy-chain 2 modules up to 50 keys
- Supports BLF/BLA, speed dialing, call pickup

Specifications:

- 160x320 graphic LCD with 16-level gray scales
- 20 physical keys each with a dual-color LED
- 20 additional keys through page switch
- Supports up to 6 modules daisy-chain
- Different icons for each function shown on the LCD
- Dual-color LED for status information
- Supports Busy Lamp Field (BLF)
- Bridge Line Appearance (BLA)
- Programmable for speed dialing, call pickup
- Intercom, hold, transfer, voicemail, forward, DND, etc.
- Applies to Tadiran IP Phone T328P and SIP-T38G
- Expansion module is powered by the host phone

Physical Features:

- 2xRJ12 (6P6C) ports for data in and out
- Power adapter: AC 100~240V input / DC 5V/3A output
- Weight: 290g
- Dimensions (mm): 145x182x100
- Operating humidity: 10~90%
- Storage temperature: max 40°C

Tadiran Telecom

About Tadiran

Tadiran Telecom (TTL) L.P., part of Afcon Industries, is an established global leader, innovator, and supplier of IP business telephony and telecommunications solutions. For nearly 50 years, Tadiran has been serving businesses of all sizes, including some of the world's largest companies and organizations in various market segments across 41 countries worldwide. With more than 100,000 satisfied end users and over 14 million installed ports worldwide, Tadiran strives to lead the industry in providing superior support and service to our global customer base. Tadiran features a comprehensive family of products including IP PBXs, Softswitches, Contact Centers, IP phones, as well as Mobility and Desktop applications. This highly versatile offering is designed to serve an ever growing list of leading companies in multiple vertical markets as varied as government, healthcare, education, hospitality, utilities, finance, transportation and more.

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Israel Tel +972-3-9262000 Russia Tel/Fax +7-495-7750855 China Tel +86-10-58696418
USA Tel +1-516-632-7200 India Tel +91-11-25850446/25854212