Techroutes NETWORK

DATASHEET

TX60

VoIP PBX for SME

Product Overview

TX60 is an IP telephony system, used to help small and medium-sized enterprises establish a convenient and high-efficient communication way.

TX60 provides 2 FXO ports and 2 FXS ports, and meanwhile it can be extended based on the Session Initiation Protocol (SIP).Interworking with traditional PBX and third party IPPBX system, it allows users to communicate through voice, fax, data or video.

TX60 supports VPN, encryption and security strategies, and thus ensures secure communication. It can be widely used in small and medium-sized call centers, enterprise branches to improve work efficiency and save communication cost.

Key Features

Physical Interface

- Supports up to 200 SIP users and 30 concurrent calls
- Supports 2 FXO and 2 FXS ports with lifeline capability
- Flexible dial rules based on time, number or source IP etc.
- Supports Multi-level IVR(Interactive Voice Response)
- Built-in VPN server/client
- Support voicemail/ Voice recording
- User-friendly web interface, classification of web user’s rights

Physical Interface

- FXS: 2 *FXS Ports
- FXO: 2 * FXO Ports
- USB Interface: 1* USB2.0
- Ethernet Interfaces
  - 1 WAN, 100/1000M, Base-T, RJ45
  - 1 LAN, 100/1000M, Base-T, RJ45
- 1 * SD card slot
- 1 * Console port
- LED Indicators: PSTN Line, FXS, USB,SD,PWR/RUN

FXS/FXO

- Connector: RJ11
- Caller ID: Bell core Type 1&2, ETSI, BT, NTT and DTMF based CID
- Dialing Mode: Pulse Dialing and DTMF
- Answer and Disconnect Signaling: Polarity Reversal for call connect/Disconnect Signaling

Environmental

- Power Supply: 12VDC, 2A
- Power Consumption: 18W
- Operating Temperature. 0 ˚C~ 40 ˚C
- Storage Temperature: -10 ˚C~90 ˚C
- Humidity: 10%-90% Non-Condensing
- Dimensions (W/D/H): 240×180×35mm
- Unit Weight: 1kg

VOIP

Voice Capabilities

- VoIP Protocols: SIP over UDP/TCP/TLS,RTP/SRTP
- Codecs: G.711a/µ law,G.723.1, G.729A/B
- Silence Suppression
- Comfort Noise Generator(CNG)
- Voice Activity Detection(VAD)
- Echo Cancellation: G.168 with up to 128ms
- Dynamic Jitter Buffer
- Adjustable Gain Control
- Automatic Gain Control （AGC）
- Call Progress Tones: Dial Tone, Ring Back Tone, Busy Tone
- FAX: T.38and Pass-through
- NAT: STUN/UPnP
- DTMF: RFC2833/Signal/Inband
- VPN Server/Client
- Software Features

Maintenance

- Web GUI Configuration
- Telnet Management
- Configuration Restore/Backup
- Multiple Languages Supported
- HTTP/TFTP/FTP Firmware Upgrade
- Auto Provision
- Syslog
- Ping and Trace route
- Traffic Statistics: TCP, UDP, RTP
- Network Capture
- NTP
- Classification of Web Users' Rights
- HTTP&HTTPS/NATS API

PBX Functionality

- Call Forward (Unconditional/No Answer/Busy)
- Call Waiting
- Call Holding
- Call Transfer
- Hotline
- Do-not-disturb
- 3-Way Conference
- Ring Group
- Call Queue
- Routing Groups
- Caller/Called Number Manipulation
- Routing based on Time Period
- Routing based on Caller/Called Prefixes
- Routing based on Source Trunks
- Dial Rules
- Failover Routing
- Multi-level IVR
- Auto-attendant Function
- CDRs
- Voicemail
- Local Recording (Support USB Storage)
- Up to 200 SIP Extensions
- Up to 30 Concurrent Calls
- Paging
- Event Report
- Email Client
- Voicemail to Email

VOIP

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