IPX4000

IP-PBX

Features

- Secured, manageable, and feature rich IP-PBX
- Built-in PSTN trunks and IP trunks
- Built-in session border controller for NAT resolution
- MOH music-in port and Paging speaker-out port
- Easy installation and setup
- Star feature codes and personal web page

IPX4000 IP-PBX are based on SIP protocol and designed for the small-to medium-size enterprise. It offers all popular features of the PBX and maximizes the interoperability with existing vendors of SIP equipment. It works with most ITSP vendors that offer SIP services. The IP-PBX also provide four built-in FXO ports to be used to connect with the PSTN. It supports one WAN and one LAN Ethernet interfaces to be able to connect to the internet and the local network , as well as four FXO interfaces used to connect to the PSTN. It also includes the capability to convert RTP voice packets to PCM format , and vice versa. First , user may register SIP user agents (UA) , for example hard or soft phones , FAX or other SIP-compliant devices like conferences systems. Second , user may connect to the outside world via trunks.

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Specifications

Maximum Capacity
- 20 SIP Extensions
- 20 Voice Mail Accounts
- 10 Concurrent Calls
- 4 PSTN Line

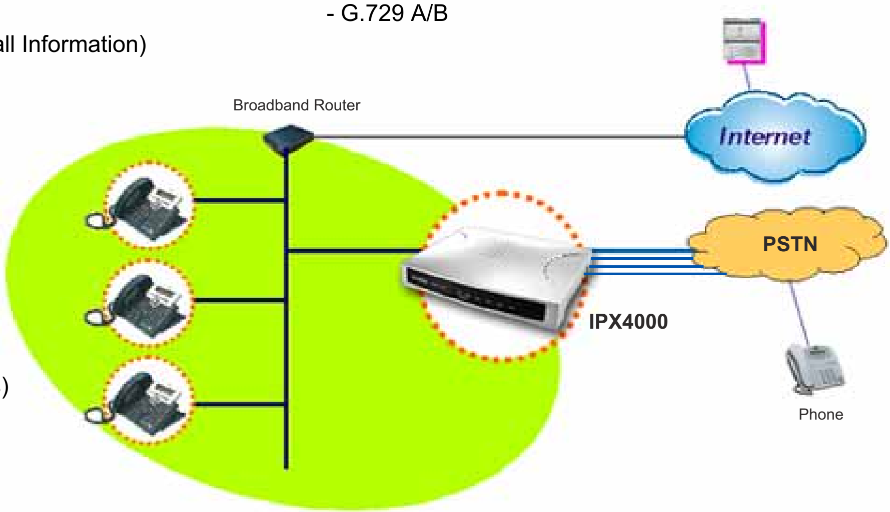
Hardware
- 1 RJ-45 Ethernet Ports
- 4 RJ-11 FXO Ports
- 1 Music-In Jack
- 1 Page-Out Jack

Supported Standards
- SIP (RFC 3261-3265, 3325, 3515, 3581, 3842, 3891, 2976,   3326, 4235), SDP (RFC 2327)
- RTP (RFC 1889, 2833), SRTP (RFC3711)
- TLS (RFC2246), STUN (RFC 3489), TFTP (RFC1350), DNS (RFC 2782, 2915)
- SNMP (RFC 1157), HTTP/HTTPS (RFC 2616, 2617)

PBX Features
- 3-Stage Hunt Group
- Address Book for Speed Dial and Caller ID Name Resolution
- Agent Group
- Anonymous Call Blocking
- Auto Attendant
- Call Data Record (Email the Last Call Information)
- Call Forward ¡V Always/Busy/No

Answer Forward
- Call Park
- Call Pickup
- Call Return
- Caller-ID Blocking
- Caller-ID Detection
- Conference Room
- Configurable RTP Port Range
- DISA (Direct Inward System Access)
- Do Not Disturb
- DTMF
- Internal Call
- IVR Node
- Multiple Language Support
- Multiple Registration per Extension
- Music on Hold (Input from .wav File or Music-In Jack)
- NAT Traversal with Built-in Session Border Controller
- Paging (to Phone or to Page-Out Speaker)
- PSTN Trunk
- Redial
- Service Flag (Night Service)
- SIP Registrar and SIP Proxy
- SIP Trunk
- SIP Trunk with Failover
- Voice Mail with Message Waiting Indication (MWI) and Email Forwarding Administration
- HTTP/HTTPS Web Based Configuration
- Extension Status Display
- Trunk Status Display
- Current Active Call Status Display
- Call Detail Record (CDR)
- Adjustable Log Level
- SIP Message Log
- SNMP

Supported Codecs
- G.711 u-Law and A-Law
- G.723.1
- G.726
- G.729 A/B



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TECOM
Converging Your Networks

TECOM CO., LTD.
23 R&D Road 2, Science-Based Industrial Park Hsin-Chu, Taiwan, R.O.C.
Tel : 886-3-5775141   Fax :886-3-5776854   web site: http://www.tecomproduct.com