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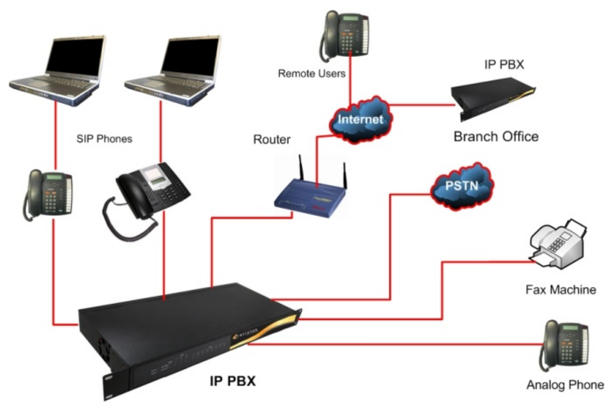
IP PBX

VX800 – SIP *Smart IP Communication System*

The Atizlan VX800 are highly versatile, integrating Analog Trunk, Analog Line, SIP Trunk, SIP Line and IP-PBX with in one box. With Built-in Registrar and Proxy Server. The VX800 can serve point to multi-point branches without external ISP services and an extra telephone fees. The VX800 can also work with existing PBXs to extend current telephone networks. This all-in-one VoIP device is the best solution for multi-point, small group connections

Function

- **Built-in DISA** provides recorded auto attendant messages and transfers incoming calls. Messages can be easily recorded through the telephone or downloaded by FTP.
- **Networks Operator** can be assigned to any extension and be reached by dialing “0” when calling from the PSTN**.**
- **Inbound, Outbound calls** are all supported. Even calls from the PSTN can be transited to other SIP number via the SIP trunk.
- **Calls transfer, call park and consult transfer** to any extension of local or remote sites through Analog or SIP lines.
- **Call forward** an incoming call to an analog line, a SIP line of local or remote VX800 or to a PSTN number.
- **Speed dialing** can be configured for 100 abbreviate numbers.
- **T.38 FAX** is supported to connect a fax machine to an analog port of the VX800.
- **Comprehensive Networking Management** through telephone set, Console, and WEB management and Telnet. Moreover, the VX800 allows remote management from external sites.



Key Features and Benefits

- **All-In-One Device, Integrates Analog Trunk, Analog Line, SIP Trunk, SIP Server, IP-PBX**
  There is no need to purchase several devices for a complete solution. Complete trunking is available using analog PSTN ports and the SIP trunk. Comprehensive extension numbers are available by employing analog and SIP line ports.
- **Built-in Registrar and proxy Server support SIP standard devices**
  RFC-3261 and RFC-2833 SIP-compliant devices, such as SIP Gateways and IP Phones, can register as an extension to the VX800.
- **Scalable up to 20 sites and 2000 extensions**
  Great adaptability and system planning are provided with 2 or 6 analog ports and scalability, which when set, works as a complete IP-PBX.
- **Register to ITSPs for additional, value-added service**
  Reduce communication costs with the additional ability to register to IP telecoms.
- **Original dialing behavior**
  Dialing plans can be maintained with E.164, offering high flexibility.
- **Multiple “Call Forwarding”**
  Never miss a call. The VX800 can variety of devices, such as GSM mobile phones, notebooks, IP Phones, etc in an order pre-arranged by the user, Call Forwarding can be set for busy, no answer, and all calls.
- **Private IP support giving SIP devices ”Plug and Play” mobility.**
  Public IP is not required. The VX800 uses Media Relay to reach devices behind NAT without additional configuration.
- **Upgrade software through Web interface.**
  User have simple, on-demand maintenance for configuration files, auto attendant greetings and software using this convenient feature.

VX800 Product Specification

System Capacity

Maximum SIP Registration: 100
Maximum Concurrent Calls: 20
Maximum Analog Trunks: 6
Maximum Scalability: 20 sites

Data/Voice Management

Protocols

- RTP (Transfer Protocol)
- Proprietary Call Control Protocol
- Standard SIP Protocol (RFC-3261, RFC-2833) Voice Codec
- G.711 a/-law, G.729AB Voice Quality
- VAD Voice Activity Detection
- CNG Comfortable Noise Generation
- Echo Cancellation G.165/G.168 16ms
- Adaptive Jitter Buffer Management Gain Control
- In/Out +/-6db

Emissions/Safety

Safety: CUL, CCC*,CB* *Future Release
PTT Regulations: FCC part 68
EMI Certifications: FCC Part 15 Class B, CE Mark

Mechanical

Power Supply External Power Adapter

Input: 100~240VAC, 50~60 Hz
Output: 12VDC, 3.0A

Management

Management

- WEB Browser
- System Console
- Telnet

Software Upgrade

- WEB
- Multiple configuration files

SIP PBX Features

Call Transfer, Call Hold, Call Park, Pickup Groups, Multiple Appearances, Intercom, DID, Forwarding, Redial, Speed Dialing, Presence Management, Call Waiting, Call Monitoring, Call Recording Traditional Conferencing, Music on Hold, Caller ID Display of Time & Date, call Forwarding, Barge In Support for multiple vendors, Report, PBX Status Meet Me Conference Rooms, Disable Line, Ring Groups, Conference Monitoring / Recording, Analog Stations / Trunks, Automated Attendant Voice Enabled PBX prompts, Disable Lines, Outgoing Calls Limit option, System Alarms, Individual Web Admin per user, Remote phones support, Multi Site Networking, Integrate with VoIP carriers.

Voice Messaging Features

Voice Message in Email, Personal Greeting, Caller ID in message, Support Forward, Listen Controls Envelope Information, Departmental Operators Integrated all in one solution, Append to message Remote message retrieval, Append to message Voice Folders, Prompting based on presence Record multiple greetings, Light / Extinguish Message waiting light.

Call Routing

Hunt Groups, Incoming DID Support, VoIP to PSTN Routing -and vice-versa-, VoIP to VoIP Routing Call Forward if Busy, Find Me Follow Me, Do not Disturb, Call Forward if Offline, Call Forward to Mobile phone, Call Forward if Unanswered, Call Forward Remote Configuration.

Environmental Specifications

Working Environment

- -10°C to 70°C Storage Temperature
- 0°C to 50°C Storage Temperature

Form Factor / Dimension

178mm (W) *311mm (D) *43mm (H)
U1 Rack Mountable

Warranty

1 Year Limited Warranty

The back view