

VoIP 400 Series

VoIP Gateway



The AirLive VoIP-400 series Voice over IP gateway is designed for performance and expansion. The series comes equipped with either 4FXO, 4FXS, or 2FXO+2FXS configurations. They allow 4 channels of simultaneous voice and fax transmission.

High Quality Internet Telephony

With AirLive eLive series of VoIP gateways, the dream of making unlimited long distance phone call and fax using conventional phone is realized. Once setup, the VoIP gateway will automatically take call request from conventional telephone handset or through company's PBX network, then make connections through Internet with the remote gateway to make traditional phone calls. All this process is transparent to users. They will enjoy the freedom to communicate freely without worrying about the telephone charge.

New Dimension of Communication freedom

The latest Voice over IP technology completely open a new dimension of freedom in real time communication. For corporate environment, VoIP allows branch offices to keep in touch as effectively as in the local office. Using the gateway technology, company can even provide remote support to oversea customers without incurring extra cost. For family or friends living far away from each other, VoIP make the world much closer. Best of all, all these technology does not require you to operate the phone call any differently. All your need is the VoIP gateways and your ordinary telephone set.

Quality and Standard Compliance

Equipped with high performance RISC and DSP processors running on real-time OS. The VoIP-400 ensure high quality voice transmission. Furthermore, its support of SIP (firmware upgrade) and H.323 protocol ensure compatibility with most VoIP equipments including Cisco's ATA186 and IP phones. QoS Support for VoIP packets maintain the quality of transmission for VoIP packets.

Built for Enterprise

With our point to point setup guide, you can easily setup your multi-point VoIP connection with other H.323 gateways. This is the ideal solution to establish high quality VoIP connection between offices. The gateway can be setup to connect with PBX (digital PBX require analogue extension) and then your VoIP call will operate identically as your regular phone call in your PBX system. Why spend more on toll call, use AirLive H.323 Enterprise gateway today.

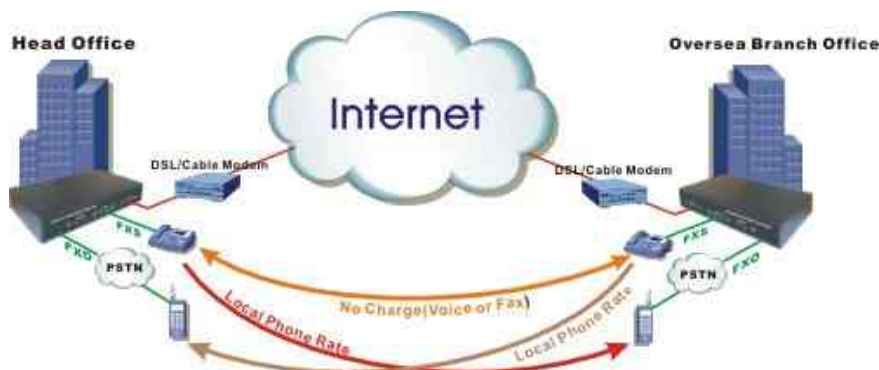
FXO: The FXO ports connect to the PSTN line (analogue telephone line from the telephone operator) or company's PBX line. It allows the VoIP device to communicate with the ordinary telephone system. Having the PXO port is necessary if you want your device to serve as a gateway for remote caller to reach your local telephone line.

FXS: The FXS ports connect to your ordinary telephone or FAX. Through your telephone or FAX, your can dial out through the gateway to other VoIP gateway or IP Phone.



WAN port: VoIP gateway has a WAN port that connects to the Internet directly or to a broadband router. User also setup the web or telnet configuration through the WAN port.

Call Forward/Transfer: This function require FXO port. It allow remote PSTN users to dial into the FXO port and use the gateway's dial-plan to make VoIP calls.



*The FXO or FXS port works with analogue telephone standard only.
If you have ISDN phone line or digital PBX, separate digital to analogue converter is required.

Specification

Key Feature

- Effective QoS scheme allowing simultaneous VoIP and Data communication
- Up to 4 channel of real-time voice and fax simultaneously
- Built-in TFTP and Flash memory for firmware upgrade via network
- Call forward and Call Transfer
- Support multiple VoIP Gateway on a LAN port
- Dynamic IP/PPPoE client support
- DHCP client support
- Dynamic DNS support
- Bulk Call successful rate over 99.98%

Call Control Protocol

- H.323 version 3 (point 2 point/multipoint)
- SIP (proxy mode)

Voice Coder Support

- G.711, G.723.1, and G.729 A/B

Real-time FAX Relay:

- T.38

Speech

- Compression algorithms: ITU G.711, G.723.1, and G.729A/B.
- Hybrid echo cancellation G.168 (16 ms)
- Auto switch between Fax and voice
- DTMF tone detection/regeneration
- Channel: four channels per module
- Comfort Noise Generation (CNG)
- User programmable Call Progress detection/generation
- Voice Activity Detection (VAD)
- User programmable Gain Control

Fax Facsimile protocol:

- T.30 Group 3
- Modulation formats: V.21, V.27ter, V.29, V.17
- Real-time fax over IP
- DTMF tone detection/regeneration



Specification

Digital Signal Processors

- One TITMS320VC5409 DSP
- 100 MIPS per DSP processor
- On-chip memory: 32K word of SRAM, 16K word of ROM
- Local SRAM 128K x 16 for each DSP

Control Processor

- 32-bit ARM7 TDMI core
- 8K byte unified cache
- 4K word Write buffer
- Embedded on-chip Ethernet MAC with associated BDMA
- Local 2M x 16 SDRAM and 1M x 16 Flash

I/O Standard

- 10/100 BaseTX RJ45 interface
- RJ 11 Loop Start interfaces for FXS/FXO

Management Tools

- RS 232 console port interface
- Web Management
- Telnet Server
- TFTP and flash memory for remote software download and upgrade

H.323 Protocol Stack

- RAS sub-stack for Terminals and Gatekeepers: supports all mandatory and optional messages (Tx and Rx) as specified in table 19/H.255.0
- H.245 sub-stack: supports the Signaling Entities of Master Slave Determination, Capability Exchange, Open Logical Channels, and Close Logical Channels
- Q.931: supports all mandatory messages as specified in table 4/H.255.0
- Compliant with H.323 Version 1 and Version 2

Environment & Power

- Dimension: 240 x 134 x 45 mm, 1.9kg
- Operating temperature: 32 to 122 F (0°C to 50°C)
- Operating humidity: 10% to 95% (non-condensing)
- Storage temperature: 14 to 140 F (-10 to 60°C) AC-to-DC
- power supply (90-260 VAC, auto-ranging, 50-60 Hz.)

EMI

- CE
- FCC part 15 A
- FXS/FXO (Compliant with ITU-T G.712)
- UL

Ordering Information

eLive VoIP-422	4-port VoIP, 2 FXO+ 2FXS ports
eLive VoIP-440	4-port VoIP, 4 FXS ports
eLive VoIP-404	4-port VoIP, 4 FXO ports