

## VoIP-400R series

### VOIP Gateway Router



The AirLive VoIP-400R series are designed for the best versatility in a IP sharing environment. The Gateways come equipped with a built-in 4-port NAT router that ensure the quality of Voice transmission while allowing your entire network to share one internet connection. It comes in varies 4-port FXO/FXS configuration and allow up to 4 simultaneous voice/fax communications.

#### 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-400R ensure high quality voice transmission. Furthermore, its support of 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.

#### Simplified Configuration

The AirLive VoIP gateway can be configured through web browser, Telnet, or console port. Each model comes with detailed instruction manual and set up example to help users start their VoIP operation quickly. Having the broadband built-in also means quantity of service and dynamic IP issues are solved automatically. It is an all-in-one solution for the VoIP and Broadband sharing environment.

#### Configurations

- VoIP 422R: Metal housing, 2FXO + 2FXS, 4 LAN ports (H.323 only)
- VoIP 404R: Metal housing, 4FXO, 4 LAN ports (H.323 only)
- VoIP 440R: Metal housing, 4FXS, 4 LAN ports (H.323 only)

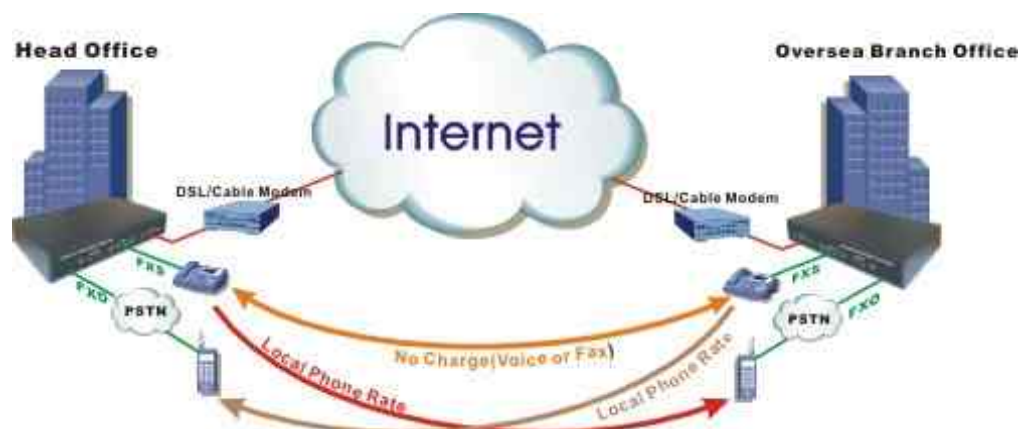


**FXO:** The FXO ports connect to the PSTN line (analogue telephone line from the telephone operator) or company's PBX line. It allows the VoIP device to communicate with the ordinary telephone system. Having the FXO port is necessary if you want your device to serve as a gateway for remote caller to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

**FXS:** The FXS ports connect to your ordinary telephone or FAX. Through your telephone or FAX, you 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
- 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
- NAT Routing Function
- Support dynamic bandwidth management and QoS

### 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



## Specification

- Support Packet filter and basic firewall functions
- DHCP Server and client support
- DMZ support
- Dynamic DNS and Virtual Server support
- Bulk Call successful rate over 99.98%

### Call Control Protocol

- H.323 version 3

### Voice Coder Support

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

### Digital Signal Processors

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

### Control Processor

- 32-bit ARM7TDMI 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

- 4 x 10/100 BaseTX RJ45 LAN port
- 4 x RJ11 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

### 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

### 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

<b>eLive VoIP-422R</b>	4-port VoIP Gateway Router, 2 FXO + 2 FXS ports
<b>eLive VoIP-404R</b>	4-port Gateway Router, 4 FXO ports
<b>eLive VoIP-440R</b>	4-port VoIP Gateway Router, 4 FXS ports