Air Live®
Powered by OvisLink Corp.

VoIP 110R

VoIP Gateway Router

The AirLive VoIP-110R is designed for the best versatility in a IP sharing environment. The Gateways come equipped with a built-in NAT router that ensure the quality of Voice transmission while allowing your entire network to share one internet connection. It comes in 1-port FXS configuration to allow 1 channel of 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. It ensures 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.

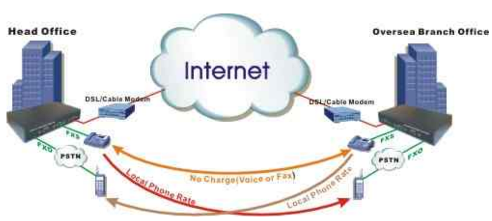
Built for SOHO/SMB environment
With our point to point setup guide, you can easily setup your multi-point VoIP connection with other H.323 gateways. This is the ideal solution to establish high quality VoIP connection between friends or offices. Why spend more on toll call, use AirLive H.323 router today.

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.

PSTN Backup Port: The PSTN backup port is different from FXO. It allows the user to press certain key to use the PSTN line for phone call. But it can not forward call from VoIP to PSTN.

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.



Manufacturer
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www.airlive.com

Specification

Key Feature

- Effective QoS scheme allowing simultaneous VoIP and Data communication
- 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
- 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 TI TMS320VC5409 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 ARM7TDMI core
- 8K byte unified cache
- 4Kworde Write buffer
- Embedded on-chip Ethernet MAC with associated BDMA
- Local 2M x 16 SDRAM and 1M x 16 Flash

I/O Standard

- 1 x 10/100 BaseTX RJ45 LAN port
- 1 x RJ11 Loop Start interfaces for FXS

Management Tools

- Web Management
- Telnet Server
- TFTP and flash memory for remote software download and upgrade

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

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: 179mm x 135mm x 45mm, 0.5kg
- Operating temperature: 32 to 122 F (0℃ to 50℃)
- Operating humidity: 10% to 95% (non-condensing)
- Storage temperature: 14 to 140 F (-10 to 60℃) 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-110R 1-port VoIP Gateway Router, 1 FXS port + PSTN backup