Air Live®
Powered by OvisLink Corp.

VoIP-211RS

VoIP Gateway Router

The AirLive VoIP-211RS is a very cost effective, highly reliable Voice over IP (VoIP) gateways that offer toll quality voice and real-time fax data over IP networks. With its integrated IP sharing router, so users use one Internet line for both voice or data function.

Supporting SIP protocol in proxy mode. The VoIP-211RS is compatible with Cisco ATA-186 and most proxy service around the world. It has been tested with the most popular proxy services including FWD(freeworld dialup), Iptel, Snomag.de, Addaline.com, BellLab, and MCI.

VoIP-211RS comes with one FXS port to connect with a phone or fax, turning your regular phone into an IP phone. It also comes with a FXO port for connection to your PSTN line or PBX, so you can access the VoIP through regular telephone and share the VoIP connection in the office environment

Buy the VoIP-211RS today and enjoy free IP Telephony instantly.

*The router is available in 3 different configuration.

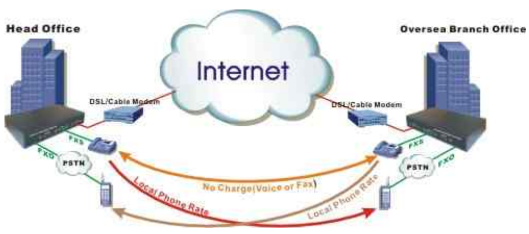
- VoIP-211RS: 1FXS and 1FXO ports
- VoIP-210RS: 1FXS ports
- VoIP-220RS: 2FXS ports

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.

Manufacturer
OvisLink Corp.

2F, No. 8, Lane 130, Min-Chuan Rd, Hsin-Tien City, Taipei, Taiwan.

Specification

CPU

- DSP & Control Processor
- System On Chip (SOC)
- MIPS-X5 unified RISC and DSP core (up to 180 DSP MIPS)
- 384K bytes on-chip RAM, 16-way interleaved with single cycle access
- 16-K byte cache
- Low power, 1.8V core voltage, 3.3V I/O voltage
- 2M bytes flash memory

Call Control Protocol

- Compliant with SIP v2.0 (RFC 3261)

Voice Function

- 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
- Comfort Noise Generation (CNG)
- User programmable Call Progress detection/ generation
- Voice Activity Detection (VAD)

Control Processor

- 32-bit ARM7 TDMI 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 RJ-45 LAN port
- 1 x 10/100-BaseTX RJ-45 WAN port
- 1 x RJ11 interfaces for FXS
- 1 x RJ11 interfaces for FXO

Management Tools

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

Fax Facsimile protocol:

- Facsimile protocol(option): T.30 Group 3, T.38
- Modulation formats: V.21, V.27ter, V.29, V.17
- Real-time fax over IP
- DTMF tone detection/regeneration

Environment & Power

- 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 (12VDC, 100-240 VAC, auto-ranging, 50-60 Hz.)

EMI

- CE
- FCC part 15 A

Ordering Information

eLive VoIP-211RS SIP VoIP Router with 1 FXS & 1 FXO port

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