ADSL2+ VoIP Router

AVR11/B

❖ Call free for long distance or international calls

❖Plug and Play for VoIP application

Features

- ANSI T1.413 Issue 2, ITU-T G.992.1(G.dmt) and G.992.2 (G.lite) compliant for Annex A, B
- ITU-T G.992.3, G.992.4 for ADSL2, ITU-T G.992.5 for ADSL2+
- Downstream and upstream data rates up to 24Mbps and 2Mbps
- Compliant with IEEE 802.3, 802.3u 10/100M Ethernet standard
- Supports SIP 3261 (Radvision SIP stack), TCP/UDP/IP, RTP/RTCP, HTTP, ICMP, ARP, DNS, and DHCP.
- Supports popular voice-coders including G.711 (A-law and µ-law), G.723.1, G.726 and G.729A/B
- Supports standard voice features such as Caller ID, Consultation Hold, Call Waiting, Blind Call Transfer, Attended Call Transfer, Call Forwarding, Mid-Call DTMF (only using RTP) and 3-Way Conferencing
- Supports Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), and Line Echo Cancellation (G.168)
- Supports DIGEST authentication (MD5 only)
- Provides easy configuration through a web interface and limited configuration through the handset dial-pad
- Standards-based implementation (ITU-T, IETF compliant)
- PSTN Backup support through Hardware Relay—applicable when power is down or when WAN interface is down
- Distinctive dial tones (PSTN, IP Dialing, and Normal) are played according to the PSTN/WAN/Registration status

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Specifications

ADSL Features
- Compatible with all leading DSLAMs
- Supports DMT line modulation
- Supports Full Rate ADSL: up to 24Mbps downstream, up to 2Mbps upstream
- Supports ITU-T G.992.1 Annex A & T1.413 Issue 2
- Supports ITU-T G.992.1 (G.dmt) Annex B / ETSI TS 101 388 / DTS TM-06006
- Supports ITU-T G.992.1 Annex B - Deutsch Telecom / U-R2
- Supports G.Lite ADSL: up to 1.5Mbps downstream
- Up to 512Kbps upstream (G.992.2)
- G.992.3 / G.992.4 (ADSL2) : Downstream and upstream data rates up to 12Mbps and 1Mbps
- G.992.5 (ADSL2+) : Downstream and upstream data rates up to 24Mbps and 2Mbps
- Supports DSL handshaking (G.994.1)

Standard
- IEEE 802.3 / 802.3u 10/100M Ethernet Standard
- SIP v1(RFC2543), v2(RFC 3261)
- Voice codec : G.711(A-law, μ-law), G.723.1, G.726, G.729A/B

Physical Interface
- 10/100Mbps Ethernet, RJ45 x 4
- RJ11 Port for ADSL Line x 1
- FXS RJ11 Port for Phone x 1
- FXO RJ11 Port for PSTN x 1
- Power DC 9V, 1A

Voice Feature
- Voice codec
  G.711 : 64k bit/s (PCM)
  G.723 : 6.3k / 5.3k bit/s
  G.726 : 16k / 24k / 32k / 40k bit/s (ADPCM)
  G.729A : 8k bit/s (CS-ACELP)
  G.729B : adds AD & CNG to G.729
- Supports silence Suppression, VAD(Voice activity detection), CNG (Comfortable noise generator) and LEC (Line echo canceller)
- Supports Packet Loss Compensation

Call Function
- Call Hold
- Call Waiting
- Call Forward
- Caller ID
- Flash
- Volume Adjustment
- Speed dial key
- 3-Way Conferencing

PSTN
- Supports PSTN Backup

Management Feature
- Web Browser
- Telnet
- Voice configuration
- HTTP

Security
- HTTP 1.1 basic/digest authentication for Web setup
- MD5 for SIP authentication (RFC2069/RFC2617)

Led indicator
- Phone, Line, Ethernet and power indicator

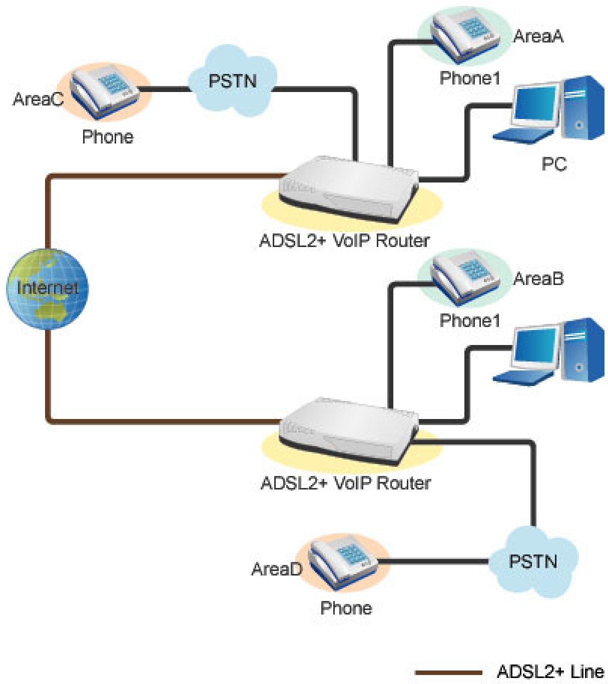
Temperature
- 32~122°F (0 ~50°C)

Humidity
- Max. 95% (Non-Condensing)

Regulatory Approvals
- FCC, CE

Application:

- AreaA(Phone1) ⟷ AreaB(Phone1) - Call Free
- AreaC(Phone) ⟷ AreaB(Phone1) - Local fee for international call
- Overseas Branch office application
- AreaD(Phone) ⟷ AreaC(Phone) - Both sides are local phone expense via VoIP (saving international call expense)



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