

ADSL2+ Gateway with VoIP

PRODUCT DATA

Intelligent ADSL Gateway Solution for Internet Connectivity and VoIP



High Speed Connections

High-speed ADSL2/+ Modem gives you a fast, "Always On" connection to the Internet

Connect PCs via the built-in Router and 4-port Switch to jump start your Ethernet network and share the Internet

Advanced firewall and security features protect your PCs and your data

Quality Voice and Carrier-Grade Feature Support

Reduce voice services costs by automatically routing PSTN telephone calls to a VoIP service provider via your ADSL broadband connection.

The AG310 features the ability to connect a standard telephone or fax machine to IP-based data networks with the additional benefit of an integrated connection for legacy telephone network "hop-on, hop-off" applications. AG310 users will be able to leverage their broadband phone service more than ever by automatically routing local calls from mobile phones and land lines over to VoIP service providers --and vice versa. If power is lost to the unit or Internet service is down, calls can be redirected to a traditional carrier via the FXO port or connect to a Telco or PBX circuit.

Phone Features

Telephony

- Service Authentication via PIN, Digest, Caller ID (Bellcore Type 1)
- Independent Configurable Dial Plans - Up to 8
- Call Waiting, Cancel Call Waiting, Call Waiting Caller ID Detection (Bellcore Type 1)
- Caller ID Blocking
- Call Forwarding to PSTN or VoIP Service: No answer, Busy, All
- Do Not Disturb
- Call Transfer
- Three-way Conference Calling with Local Mixing
- Message Waiting Indication - Visual and Tone Based
- Call Return
- Call Back on Busy
- Call Blocking with Toll Restriction
- Delayed Disconnect
- Distinctive Ringing - Calling and Called Number
- Off-hook Warning Tone
- Selective/Anonymous Call Rejection
- Hot line and Warm Line Calling
- Speed Dialing of 8 Numbers/Addresses
- Fax: G.711 Pass Through or Real Time Fax over IP via T.38

Product Specific

- VoIP to PSTN (USA) Service Call Origination and Termination
- PSTN (USA) to VoIP Service Call Origination and Termination
- Single Stage and Two Stage Dialing
- Forward Calls to VoIP service - Selective, Authenticated, All
- Forward Calls to PSTN service - Selective, Authenticated, All
- PSTN Line Sharing with Multiple Extensions
- Automatic PSTN Fallback (Loss of Power or IP Service to Unit - with Quiescence to Normal Operations)
- Advanced Inbound and Outbound Call Routing
- Force PSTN Disconnection
- Sequential Dialing Support

VoIP to PSTN Authentication and Routing Features

- VoIP to PSTN Gateway Enable/Disable
- VoIP Caller Auth Method (None, PIN, HTTP Digest)
- VoIP PIN Max Retry Setting
- One Stage Dialing Enable/Disable
- VoIP Caller ID Pattern Matching
- VoIP Access Allowed Caller List (No Further Authentication)
- VoIP Caller PIN and Associated Dial Plan
- PSTN to VoIP Gateway Enable/Disable
- Ring Through to FXS Enable/Disable
- Ring Through Tone - Configurable
- Caller ID (Bellcore Type 1) for VoIP Service Access
- Caller ID Enable/Disable
- PIN Max Retry Settings
- Caller PIN and Associated Dial Plan
- Least Cost Routing (Via Outbound VoIP – Line1 Dial Plan)

FXO Behavior Features

- VoIP Answer Delay Timer
- PSTN Answer Delay Timer
- VoIP PIN Digit Time-Out Timer
- PSTN PIN Digit Time-Out Timer
- PSTN-to-VoIP Call Max Dur Timer
- VoIP-to-PSTN Call Max Dur Timer
- PSTN Ring Through Delay Timer

- PSTN Dialing Delay Timer
- VoIP DIG Refresh Interval Timer
- PSTN Ring Time-out Timer

PSTN Disconnection Detection Features

- CPC (Removal of Tip/Ring Voltage Momentarily)
- Polarity Reversal
- Long Silence (Configurable Time Setting)
- Disconnect Tone (e.g. Reorder Tone)
- Silence Threshold

International Control Features

- FXO Port Impedance - Configurable to 16 settings
- Ring Frequency - Configurable
- SPA to PSTN and PSTN to SPA Gain Settings
- Ring Frequency - Maximum Setting
- Ring Validation Time Setting
- Tip/Ring Voltage Adjustment Setting
- Ring Indication Delay Setting
- Operational Loop Current Minimum Value
- Ring Time-out Setting
- On-Hook Speed Setting
- Ringer Impedance Setting
- Line-in-Use Voltage Setting

Gateway Features

- Supports Static and Dynamic Routing (RIP1 and 2)
- DHCP server capability to assign IP addresses automatically
- Remote administration and remote upgrades available over the Internet
- Supports Fixed IP, DHCP client, PPTP client or PPPoE to access remote server
- Supports Universal Plug and Play (UPnP)
- Supports ALG for FTP, TFTP, RTSP, PPTP/GRE, SIP, H.323, ESP, IRC, TALK, MMS.
- Supports IP multicasting (RFC 2236, RFC 3376)
- Supports IP QoS (IEEE 802.1q)
- Supports DHCP Option (1,6,12,43,50,51,53,54,55,60,61,66,82,121)
- Supports programmable MTU, Path MTU discovery (RFC1191)
- Supports SNTP (Simple Network Time Protocol) for synchronizing with real-time server

Security

- Password protected configuration for web access
- PAP and CHAP authentication
- URL filtering, and keyword, Java, ActiveX, Proxy, Cookie blocking
- ToD filter (Blocks Access by Time)
- Port Filtering, MAC and IP Address Filtering, DMZ Hosting, and NAT Technology
- Stateful Packet Inspection (SPI) firewall with Denial of Service (DoS) prevention
- Access Control Lists
- VPN Passthrough for IPSec, PPTP, and L2TP Protocols
- VPN termination – up to 5 IPSec tunnels
- Cisco QuickVPN
- Password Protected Admin and User Access Authority
- VoIP Provisioning/Configuration/Authentication:
HTTPS with Factory Installed Client Certificate
HTTP Digest - Encrypted Authentication via MD5 (RFC 1321)
Up to 256-bit AES Encryption

Provisioning and Management

- Web GUI for easy configuration from any web browser
- Remote administration and remote upgrades available over the Internet
- E-mail and Web-based Logging of Security Events

PRODUCT DATA

Model No. **AG310**

- Supports traffic and event logging
- TR-059 – support QoS-Enabled IP Services
- TR-069 – support for all required RPCs and parameters
- TR-064 (optional firmware)
- TR-068 – performance & interoperability
- Remote access to CLI via Telnet
- Firmware upgrade via HTTP, TFTP
- Configuration file backup and restore
- SNMP v1, v2, v3 - MIB II, DSL Forum ADSL MIB, 802.11 MIB
- Diagnostics – Self test for Flash/RAM memory, LAN port, Wireless LAN
- DSL Statistics
- ITU-T I.610 F4/F5 Loopback
- ATM Ping, Ping Traceroute
- Linksys Syslog on LAN
- Reset to factory defaults
- VoIP Provisioning, Administration & Maintenance
- Comprehensive Third Party SIP server and VoIP Gateway Interoperability Documentation:

Documentation

- Quick Installation, User, and Configuration Guides
- Administration Guide - Service Providers Only
- Provisioning Guide - Service Providers Only

Environmental

Dimensions	6.69" x 6.69" x 1.22" (170 mm x 170 mm x 31 mm) W x H x D
Unit Weight	14.39 oz. (0.41 kg)
Power	12VDC, 1.25A (Power supplies available for North America, EU, or UK)
Operating Temp.	32°~104°F (0°~40°C)
Storage Temp.	-4°~158°F (-20°~70°C)
Operating Humidity	10~85% Non-condensing
Storage Humidity	5~90% Non-Condensing

Package Contents

- AG310
- Power adapter/supply
- Installation CD
- 6 ft. RJ-45 Yellow cable
- 2 6-ft. RJ-11 Gray cables; Cabling Type CAT 5 UTP
- White box

Specifications

Physical Interfaces

Power input and on/off switch

Reset Button

ADSL – 1 RJ-11 (Annex A model), or RJ-45 (Annex B model) Port.

Ethernet - 4 Ports, RJ-45, IEEE 802.3u, 10/100 Auto MDI/MDI-X LAN switch

Phone – 1 RJ-11 FXS Phone Port, for Analog Circuit Telephone Device (Tip/Ring)

FXS - Ring Voltage: 40-55 V Configurable

Subscriber Line Interface Circuit (SLIC) -

Ring Frequency: 10 Hz - 40 Hz

Ring Waveform: Trapezoidal and Sinusoidal

Maximum Ringer Load: 3 REN

On-hook/off-hook Characteristics:

On-hook voltage (tip/ring): -50 V NOMINAL

Off-hook current: 25 mA min

Terminating Impedance: 8 Configurable Settings including

North America 600 ohms, European CTR21

Line - 1 RJ-11 FXO Phone Port, For a Telco or PBX Connection

LEDs - (per TR-068) Power (Red/Green), Ethernet (1-4) (Green), Phone (Green) Line (Green), DSL (Green), Internet (Red/Green), Voice Status (Red/Green)

DSL Standards

Annex A, Annex B, UR-2 models, G.992.1 (G.dmt) , G.992.2 (G.lite), T1.413i2 Issue 2, G.992.3 (ADSL2) (Annex B future), G.992.4 (G.lite.bis), G.992.5 (ADSL2+), G.994.1 (G.hs), Dying Gasp, S=1/2

ATM Specifications

8 PVCs

CoS – UBR, CBR, VBRrt, VBRnrt with Traffic Shaping

RFC2684 LLC Encapsulation and VC Multiplexing over AAL5

RFC2364 PPPoA

RFC2225 Classical IP over ATM (CLIP)

RFC2516 PPPoE

RFC1483 bridged, routed

OAM F5

VPI Range (0-255)

VCI Range (32-65535)

Bridging

802.1d transparent bridging, RFC2684/1483 bridged encapsulation, Supports bridging to Ethernet ports. Simultaneous bridging and routing

Routing/Networking Functions

NAT

PAT

NAPT

ALGs, including HTTP, H.323, AIM, MSGames, Diablo, RTP, RTSP, SIP, NetMeeting, MSN Messenger, FTP, Quick Time, mIRC, Real Player, CuSeeMe, VPN pass through, and others

DMZ

DNS Server/Relay

TFTP Client

Protocols : TCP/IP, UDP, ICMP, ARP, ARP Proxy, IPCP

IGMP Proxy v1, v2, & V3

IP QoS per TR-059:

US traffic forwarding based on:

destination IP address(es) with subnet mask

originating IP address(es) with subnet mask

source MAC address

protocol (TCP, UDP, ICMP)

source port

destination port

DiffServ codepoint (IETF RFC 3260)

IEEE 802.1Q VLAN identification and packet length.

One Best Effort (BE) queue, one Expedited Forwarding (EF) queue, four Assured Forwarding (AF) queues.

IP ToS (RFC 1349) for upstream queues

UPnP, UPnP NAT Traversal

MAC address filtering

Static and Dynamic Routing (RIP1 and 2)

DHCP Server/Client/Relay

NTP (Network Time Protocol) for Synchronizing with a Real-Time Server

PVC to port mapping (Ethernet and Phone ports)

Voice Gateway

SIPv2: Session Initiation Protocol v2 (RFC 3261, 3262, 3263, 3264)

SIP Proxy Redundancy - Dynamic via DNS SRV, A Records

Re-registration with Primary SIP Proxy Server

SIP Support in Network Address Translation Networks - NAT (incl. STUN)

Secure (Encrypted) Calling via Pre-Standard Implementation of Secure RTP

Codec Name Assignment

Voice Algorithms

G.711 (A-law and μ -law)
G.726 (16/24/32/40 kbps)
G.729 A
G.723.1 (6.3 kbps, 5.3 kbps decrypt)
Dynamic Payload
Adjustable Audio Frames per Packet

Fax Capability

Fax Tone Detection and Pass-Through (Using G.711)
Fax Pass-Through - Using G.711
DTMF: In-band & Out-of-band (RFC 2833) (SIP Info)
Flexible Dial Plan Support with Interdigit Timers and IP Dialing
Call Progress Tone Generation
Jitter Buffer - Adaptive
Frame Loss Concealment
Full Duplex Audio
Echo Cancellation (G.165/G.168)
VAD - Voice Activity Detection with Silence Suppression
Attenuation / Gain Adjustments
Flash Hook Timer
MWI - Message Waiting Indicator Tones
VMWI - Visual Message Waiting Indicator via FSK
Polarity Control
Hook Flash Event Signaling
Caller ID Generation (Name & Number) - Bellcore, DTMF, ETSI
Music on Hold Client
Streaming Audio Server - up to 10 sessions

VoIP Provisioning, Web Browser Administration & Configuration via Integrated Web Server Administration & Maintenance:

Telephone Key Pad Configuration with Interactive Voice Prompts
Automated Provisioning & Upgrade via HTTP, TFTP
Asynchronous Notification of Upgrade Availability via SIP NOTIFY
Non-intrusive, In-Service Upgrades
Report Generation & Event Logging
Stats in BYE Message
Syslog & Debug Server Records - Per Line Configurable
Per Line and Purpose Configurable Syslog and Debug Options

Certifications

FCC Part 15B Subpart B Class B, FCC Part 15C Subpart B, FCC Part 68
Safety -UL 60950,
UL 1950
A-Tick
UPnP able/cert Able
CE certification pending
Comply to RoHS

The Linksys ADSL2/2+ Gateway with integrated FXS and FXO Phone Ports are part of a complete portfolio of products for data, voice, and video services delivery. The gateway is an integrated solution that enables the service provider to offer not just broadband connectivity, but also value-added services like home networking and VoIP. The gateway combines a high speed ADSL2/2+ interface, 4 port Ethernet switch, and analog telephone adapter (ATA) into one compact and cost-effective design. Market leading features have been included in each aspect of the solution.

The high speed ADSL2/2+ interface meets the needs of a variety of deployment architectures with support for up to 8 PVCs and port mapping. Remote management and provisioning via DSL Forum TR-069 protocol reduces service provider operational expenses and improves the end user experience. With an integrated router, the gateway provides all the key features that residential users have come to expect, as well as those that the service providers will require for future services and upgrades. Security features like an SPI firewall, filtering, and VPN support protect the end user's home network, while advanced features like QoS and IGMP help create the framework for upgrading to triple play services.

The design incorporates one FXS port for simple connection of standard analog telephones or fax machines. The gateway supports VoIP using SIP with an extensive and flexible set of configurable parameters. Service providers can deliver toll quality IP phones with advanced features, such as call waiting, caller ID, and conferencing over their DSL broadband service. The AG310 supports one PSTN FXO port to connect to a Telco or PBX circuit. The AG310 FXS and FXO lines can be independently configured via software controlled by the service provider or the end user.

A user calling from a mobile phone or land line will be able to reduce and even eliminate international and long distance telephone charges by first calling their AG310 via a local telephone number. The advanced authentication and call routing intelligence programmed into the AG310 will route the call via the Internet to the far end destination. In addition, when using the AG310 at the far end, VoIP calls placed to that location can be either answered or further processed and routed on as a local call to any legacy land line or mobile phone.

The residential gateway is the all-in-one solution for launching advanced services today, like voice, as well as a manageable, upgradeable platform for future offerings. As part of a family of home networking products that cover personal computing, communications, entertainment, and home monitoring and management, the Linksys ADSL2/2+ Gateway with Phone Ports provides the complete answer for both the Service Provider and their customers.

Linksys

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<http://www.linksys.com/international>

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