Access201N

VoIP Residential Gateway

Superior voice quality even with heavy data traffic

The Access201N, Residential VoIP Gateway, delivers excellent voice quality over existing broadband services (DSL, cable, etc.). This compact device was specifically designed for residential and small office, home office (SOHO) so they can continue to use their existing analog phones and faxes while enjoying the benefits Voice-over-IP (VoIP) offers — price and advanced features.

Access201N is a cost-efficient means for service providers to migrate their customers' traditional analog telephones and fax machines onto IP-based networks.

It can be used by subscribers who already have routers or broadband modems with integrated routers in their home or office networks. Service providers can give their customers' all the benefits of emerging IP applications, without replacing their existing analog infrastructures.

Access201N is based on standards-based VoIP technology that is deployed in networks globally. It uses standard tools for configuration and provisioning (TFTP and HTTP servers) and troubleshooting (Syslog).

This unique single-box solution supports T.38 for analog-to-IP fax conversion, G.729 as well as other complex compression algorithms on both voice ports, and it supports all current VoIP signaling standards (MGCP, SIP).

Access201N employs embedded support for IEEE 802.1q VLAN tagging and 802.1p QoS for basic traffic engineering. Each Access201N has two RJ-11 voice ports that support independent telephone numbers, and one 10/100BaseTX port. The voice ports interface directly with any analog telephone or fax machine. The 10/100BaseTX port provides connectivity to existing Ethernet LAN networks or broadband connections such as xDSL, cable modems, or fixed wireless solutions.

Access201N is one in a series of VoIP residential gateways that deliver up to 4 Ethernet ports for the LAN or video applications, integrated routing options, 10/100BaseTX copper, and 100BaseFX fiber uplink WAN interfaces.

Product Highlights

- Two independent phone lines (FXS) provide VoIP service to analog telephones and fax machines
- Complex CODECs on both ports as well as T.38 and G.711 fax
- Compact stackable, desk or wall-mounted solution is fast and simple to install and use
- Mass auto-configuration and auto-provisioning using HTTP or TFTP servers
- Integrated Web server — GUI interface simplifies provisioning
- Flexible multi-protocol support: MGCP, SIP
- Telephone line pulse metering, used as call-charge units for pay phone
- Interoperable with most major softswitches

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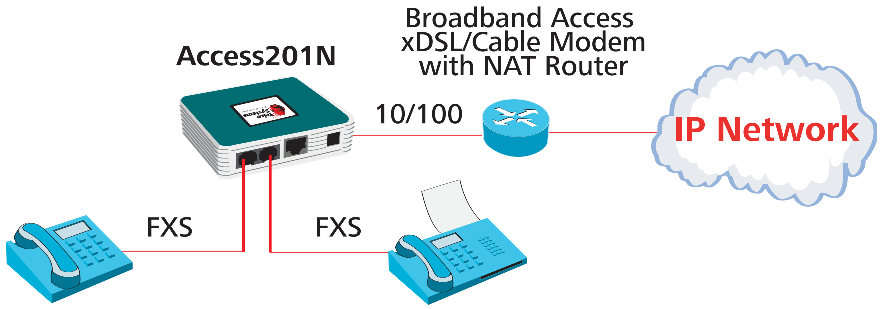

Specifications

Voice protocols

MGCP (RFC 2705, RFC 3660, NCS)
SIP (RFC 2543, RFC 3261)

CODECS (concurrently on both ports)

ITU G.711 (μ-law, A-law)
ITU G.723.1
ITU G.726
ITU G.729 A/B
ITU T.38 (SIP, MGCP)
ITU FXS

Voice features

Automatic fax/modem detection
Voice Activity Detection (VAD)
Comfort Noise Generation (CNG)
RTP voice packet encapsulation
DTMF detection and generation
Enhanced call features: Caller ID, call hold, attended/unattended call forward, 3 way conferencing, redial
Automatic tone generation: dial, busy, fast busy, ring back, stutter tone, distinctive ring

Switching and Provisioning

Dynamic Host Configuration Protocol (DHCP) client
(RFC 2131)
RFC 2068 HTTP
TFTP
Static and dynamic IP address assignment

Quality of Service

IEEE 802.1q
IEEE 802.1p
BOOTP

WAN Interface

10/100BaseTX auto sensing (IEEE 802.3u)

Connector:	RJ-45
Transmission:	Full/half duplex

WAN Cabling

10BaseT	UTP, Cat. 3, 4, 5, 100m.
100BaseTX	UTP, Cat. 5, 100m

Telephony Interface

2x Analog Connector RJ-11 (POTS), FXS
5 REN

Indicators

WAN	WAN activity
Phone1	Line status
Phone2	Line status
Ready	Unit and application indicator
Power	

Management

Internet - Web-based management for user and Admin Interfaces Telnet, Web, Remote auto configuration mass software download and auto configuration using TFTP or HTTP
IP Configuration DHCP or static
SW Download auto-upgrade via TFTP and HTTP

Security

Passwords, encrypted configuration file

General

Dimensions (H) 0.95" (24.1mm)
(W) 3.5" (88.9mm)
(D) 3.5" (88.9mm)
Power 5 VDC external power supply
Weight 0.88 lbs (0.4 kg)
Operating Temp. 0° to 45° C (32° to 113°F)
Operating Humidity 10% – 90%, non-condensing

Emission and Safety Regulations

FCC Class B, UL, CUL, CE

Features and Benefits

- Full SIP and MGCP protocol support
- Voice activity detection (VAD) saves bandwidth
- Complex CODEC schemes simultaneous on both ports minimizes bandwidth use
- Telephone line pulse metering, used as call-charge units for pay phone
- Interoperable with most major softswitches
- Integrated web server for simple end-user provisioning
- Fully managed solution and auto provisioning and automatic configuration with TFTP and HTTP to aid large installations
- Desk or wall mounted

Ordering Information

Part Number	Description
AC201N-X	VoIP gateway, 2 voice ports, 1 10/100BaseTX uplink port. X=S for SIP, M for MGCP

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