ZOOM®

Zoom Voice over IP Telephone Adapter

Economical solution provides Voice over the Internet using conventional telephones

Uses industry-standard Session Initiation Protocol (SIP)

Clear, natural-sounding voice quality

Supports remote delivery of firmware updates

Supports automatic provisioning by service providers

Supports Quality of Service (QoS)

Supports STUN (Simple Traversal of UDP over NATs)

Supports advanced telephone services including CLASS features such as Call Waiting, Caller Identification, Call Transfer, Call Hold, Call Forwarding, Distinctive Ring, and Voice Message Waiting Indication

Proven compatibility with SIP-standard servers from Asterisk™ Cisco, NetCentrex™, Quintum and more.

Model 5806 with G.711, iLBC and G.729 Codecs
Model 5807 with G.711 and iLBC Codecs

The economical Zoom Model 5806 and Model 5807 Voice over IP Telephone Adapters allow standard analog telephones to make and receive calls over a broadband Internet connection. Simply plug a conventional or cordless phone into the adapter and then plug the telephone adapter into a broadband-connected router or Internet gateway. Up to five and often more attached phones can be rung simultaneously using the built-in ring generator.

The Model 5806 and Model 5807 support industry-standard Session Initiation Protocol (SIPv2) and work with a wide range of service providers and SIP-based VoIP equipment.

Voice over IP Service providers can deliver a rich variety of advanced telephone services through the Models 5806, or 5807 including CLASS features such as Call Waiting, Caller Identification, Call Transfer, Call Hold, Call Forwarding, Distinctive Ringing, and Voice Message Waiting Indication.

The VoIP terminal adapters can be configured remotely using a TFTP or HTTP download from the service provider and updates to the firmware can be automatically delivered. Local configuration is done with a browser-based graphical user interface.

The Model 5806 supports G.711, iLBC and G.729 voice codecs. The Model 5807 supports G.711 and iLBC.

Both units are built and supported by Zoom Technologies, a publicly-traded company (NASDAQ:ZOOM) with over 28 years of experience in telephony and data communications.

Specifications

Analog Telephone Ports	• One FXS type Loop-start interface with RJ-11 • Up to 5 REN (Ringer Equivalence Number), up to 5 and often more typical telephones • Programmable Ring Patterns • Call progress tones supported: Initial dial tone, Secondary dial tone, Stuttered dial tone, Message waiting dial tone, Call forward dial tone, Pre-ringback dial tone, Ring back tone, Call waiting tone, Call holding tone, Call disconnect tone, Call conference tone, Busy tone, Reorder tone (network busy/fast busy), Off hook warning.
Status Indicators	• Power, LAN link, VoIP ready, VoIP call in progress, voice message waiting
Voice over IP (VoIP) Protocols	• SIPv2 - Session Initiation Protocol (RFC 3261, 3262, 3263, 3264) • SDP - Session Description Protocol (RFC 2327) • RTP - Real Time Protocol (RFC 1889, 1890) • RTCP - Real-Time Control Protocol (RFC 1889) • X-NSE - Tone Events for SIP/RTP (RFC 2833) • AVT - Tone Events for SIP/RTP (RFC 2833) • Power-on Auto Registration • Re-registration with SIP Proxy Server • SIP over UDP • SIP authentication (HHP Digest with MD5)

Specifications (continued)

Network Protocols	• IPv4 - Internet Protocol Version 4 (RFC 791)
	• TCP - Transmission Control Protocol (RFC 793)
	• UDP - User Datagram Protocol (RC 768)
	• ICMP - Internet Control Message Protocol. (RFC 792)
	• RARP - Reverse Address Resolution Protocol (RFC 903)
	• ARP - Address Resolution Protocol (RFC 826)
	• DNS - Domain Name Server
	• DHCP Client - Dynamic Host Control Protocol (RFC 2131)
	• NTP - Network Time Protocol (RFC 1305)
	• SNTP - Simple Network Time Protocol (RFC 2030)
	• STUN - Simple Traversal of UDP over NATs (RFC 3789)
	• HTTP - HyperText Transfer Protocol
	• TFTP - Trivial File Transfer Protocol (RFC 1350)
Voice Codecs	• G.711 - Pulse Code Modulation
	• iLBC (Internet Low Bitrate Codec)
	• G.729 (Model 5806 only)
Telephony	• Q.24 DTMF generation and detection
	• Configurable tone frequency and on/off cadence generation
	• Caller ID Generation and Detection (Type I and II)
	• 3-way conference calling with local mixing
	• Message waiting indicator light
	• G.711 Fax Pass-through
	• CLASS feature support
	• G.165, G.168 compliant line echo cancellation
	• Nonlinear echo cancellation
	• Double talk detection
Quality of Service Support	• Layer 2 Class-of-Service (CoS) Tagging (802.1P)
	• Layer 2 (802.1Q VLAN)
	• Layer 3 Type-of -Service (ToS) Tagging (RFC 791/1349)
	• Layer 3 DIFFServ (RFC 2475)i 0fo
Security	• Provisioning/Configuration/Authentication
	• Password-protected, Web based administration
	• ARC4 Encryption for TFTP Configuration Profiles
	• Authentication (Digest using MD5)
Size	• 14.6 cm X 11.2 cm X 2.8 cm (5.7 inches X 4.4 inches X 1.1 inches)
Minimum Requirements	• High speed Internet connection (typically a DSL or data-over cable connection)
	• A router or gateway to share the broadband Internet connection
	• A Touchtone telephone (conventional analog phone) or a fax machine
	• CD ROM drive in a personal computer which supports a Web browser (Windows, Macintosh, Linux or other) and is connected to the router

Package contents:

- VoIP telephone adapter
- Power adapter
- Quick Start Guide
- CD ROM
- Ethernet cable
- Phone cord

International Headquarters
Zoom Technologies, Inc.
207 South Street
Boston, MA 02111
USA
Tel: 617 423-1072
Fax: 617 423-3923
email: sales@zoom.com
www.zoom.com

Nasdaq: ZOOM

European Sales
Zoom Technologies, Inc.
Central House
2 Kings Road
Fleet, Hants
GU51 3AD   UK
Tel: +44 (0) 1252 582000
Fax +44 (0) 1252 617920