LINKSYS®
A Division of Cisco Systems, Inc.

CISCO SYSTEMS

Internet Phone Adapter with 2 Ports for Voice-over-IP

PRODUCT DATA

Feature-Rich VoIP Service through your High-Speed Internet Connection

Enables feature-rich telephone service over your broadband Internet connection

Two standard telephone ports for analog phones or use one of the ports for a fax machine, each with an independent phone number

High quality, clear sounding voice service simultaneous with Internet use

Compatible with all common telephone features: Caller ID, Call Waiting, Voicemail, etc.

Model No. **PAP2T**

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Features

Telephony

- Two voice ports (RJ-11) for analog phones or Fax machines
- Impedance Agnostics - 8 Configurable Settings
- Call Waiting, Cancel Call Waiting, Call Waiting Caller ID
- Caller ID with Name/Number (Multi-national Variants)
- Caller ID Blocking
- Call Forwarding: 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
- Music on Hold

Package Contents

- PAP2T Phone Adapter Unit
- Power Adapter
- RJ-45 Ethernet Cable
- Quick Installation Guide

Specifications

Model	PAP2T * Note: Many specifications are programmable within a defined range or list of options. Please see the PAP2T Administration Guide for details. The configuration profile is uploaded to the PAP2T at the time of provisioning.
Data Networking	MAC Address (IEEE 802.3) IPv4 - Internet Protocol v4 (RFC 791) upgradeable to v6 (RFC 1883) ARP - Address Resolution Protocol DNS - A Record (RFC 1706), SRV Record (RFC 2782) DHCP Client - Dynamic Host Configuration Protocol (RFC 2131) ICMP - Internet Control Message Protocol (RFC792) TCP - Transmission Control Protocol (RFC793) UDP - User Datagram Protocol (RFC768) RTP - Real Time Protocol (RFC 1889) (RFC 1890) RTCP - Real Time Control Protocol (RFC 1889) DiffServ (RFC 2475), Type of Service - TOS (RFC 791/1349) SNTP - Simple Network Time Protocol (RFC 2030)
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) Dynamic Payload Adjustable Audio Frames per Packet
Fax Capability	Fax Tone Detection Pass-Through Fax Pass-Though - 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

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Specifications

	Attenuation / Gain Adjustments
	Flash Hook Timer
	MWI - Message Waiting Indicator Tones
	VMWI - 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
Security	Password Protected System Reset to Factory Default
	Password Protected Admin and User Access Authority
	Provisioning/Configuration/Authentication:
	HTTPS with Factory Installed Client Certificate
	HTTP Digest - Encrypted Authentication via MD5 (RFC 1321)
	Up to 256-bit AES Encryption
Provisioning, Administration & Maintenance:	Web Browser Administration & Configuration via Integral Web Server
	Telephone Key Pad Configuration with Interactive Voice Prompts
	Automated Provisioning & Upgrade via HTTPS, HTTP, TFTP
	Asynchronous Notification of Upgrade Availability via NOTIFY
	Non-intrusive, In-Service Upgrades
	Report Generation & Event Logging
	Stats in BYE Message
	Syslog & Debug Server Records - Per Line Configurable
Physical Interfaces:	1 10baseT RJ-45 Ethernet Port (IEEE 802.3)
	2 RJ-11 FXS Phone Ports - For Analog Circuit Telephone Device (Tip/Ring)
Subscriber Line Interface Circuit (SLIC):	Ring Voltage: 40-55 VRMS Configurable *
	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
Regulatory Compliance:	FCC (Part 15, Class B), cUL, CE, IC-003, A-Tick
Power Supply:	DC Input Voltage: +5 VDC at 2.0 A Max.
	Power Consumption: 5 Watts
	Switching Type (100-240v) Automatic
	Power Adapter: 100-240v - 50-60Hz (26-34VA) AC Input, 1.8m cord
Indicator Lights/LED:	Power, Ethernet, Phone1, Phone2
Documentation:	Quick-Start Installation and Configuration Guide
	User Guide
	Administration Guide - Service Providers Only
	Provisioning Guide - Service Providers Only

Environmental

Dimensions	3.98” x 3.98” x 1.10” (101mm x 101mm x 28mm) W x H x D
Unit Weight	5.40 oz. (0.153kg)
Operating Temp.	32º to 113ºF (0º to 45ºC)
Storage Temp.	-77º to 158ºF (-25º to 70ºC)
Operating Humidity	10 to 90% Non-condensing
Storage Humidity	10 to 90% Non-Condensing

The Linksys Internet Phone Adapter enables high-quality feature-rich VoIP (voice over IP) service through your broadband Internet connection. Just plug it into your home Router or Gateway and use the two standard telephone ports to connect analog phones or use one of the ports for a fax machine. Each phone port operates independently, with separate phone service and phone numbers, like having two telephone lines. You'll get clear reception and a reliable fax connection, even while using the Internet at the same time.

With Internet telephony, along with low domestic and international phone rates, an impressive array of special telephone features are available. Choose your preferred free local dialing area code, regardless of where you live. Or add a virtual telephone number in any area code, forwarded to your Internet phone. You can even add a toll-free number. The Linksys Internet Phone Adapter is compatible with these and all of the other special telephone features that are available from your Internet telephony service provider, such as Caller ID, Call Waiting, Voicemail, Call Forwarding, Distinctive Ring, and much more.

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Linksys products are available in more than 50 countries, supported by 12 Linksys Regional Offices throughout the world. For a complete list of local Linksys Sales and Technical Support contacts, visit our Worldwide Web Site at www.linksys.com.

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