

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

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

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-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

Package Contents

Specifications

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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
Physical Interfaces:	Syslog & Debug Server Records - Per Line Configurable
	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: Documentation:	Power, Ethernet, Phone1, Phone2
	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.	32° to 113°F (0° to 45°C)
Storage Temp.	-77° to 158°F (-25° to 70°C)
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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