

ProVu Communications

Linksys SPA-2102



A Smart, Compact VoIP Phone Adapter

The SPA-2102 features 2 FXS ports which can be connected to either; existing analogue phones, fax machines or to a PBX system. The SPA-2102 includes 2 100BaseT RJ45 Ethernet interfaces (LAN-WAN) to connect to a home or office LAN, as well as an Ethernet connection to a broadband modem or router (WAN).

Easy to Install

The SPA-2102 is easy to install by the end user and can be remotely provisioned, configured and maintained by the service provider. Each SPA-2102 converts voice traffic into data packets for secure transmission over an IP network. The SPA-2102 supports many valuable call features such as Music on hold, speed dial and call forwarding.

Suitable for both Business and home environments

Compact in design, the SPA-2102 is suitable for both consumers and business VoIP service offerings including a full-featured IP Centrex environment. The SPA-2102 uses international standards for voice and data networking for reliable voice and fax operation.

Key Features

- Two Voice Ports (RJ11) for analogue phones or FAX machines
- Impedance agnostics - 8 Configurable Settings
- Call Waiting, Cancel call waiting, Call Waiting Caller ID, Call Return
- Caller ID with Name/Number
- Call Forwarding: No Answer, Busy, All
- Music on Hold, Do not Disturb, Call return, Call back on Busy, Call Blocking with Toll restriction.
- Message Waiting Indication – Visual and Tone based
- Three way Conference Calling with Local Mixing
- Speed Dialling of 8 Numbers/Addresses
- Fax Capability
- Ethernet Port
- Remotely Configurable, ideal for ITSPS
- Hot Line and Warm line calling

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Technical Data

Connectors:

- 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)
- DHCP Server - 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)
- Upload Data Rate Limiting - Static and Automatic
- QoS - Voice Packet Prioritization over Other Packet Types
- Router or Bridge Mode of Operation
- MAC Address Cloning
- Port Forwarding

User interface:

- 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

Web server:

- 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 Inter digit Timers and Ip Dialling
- Jitter Buffer – Adaptive
- Call Progress Tone Generation
- 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 Signalling
- Caller ID Generation(name, number) – Bell core, DTMF, ESTI
- 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 and 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
- Per Line and Purpose Configurable Syslog and Debug Options

SIP:

- 2 100baseT RJ-45 Ethernet Port (IEEE 802.3)
- 2 RJ-11 FXS Phone Ports - For Analogue Circuit Telephone Device (Tip/Ring)

Subscriber Interface circuit (SLIC):

- Ring Voltage: 40-55 VRMS Configurable
- Rng Frequency: 10 Hz - 40 Hz
- Ring Waveform: Trapezoidal and Sinusoidal
- Maximum Ringer Load: 3 REN
- On-hook/off-hook Characteristics:L
- On-hook voltage (tip/ring): -50 V NOMINA
- Off-hook current: 25 mA min.
- Terminating Impedance: 8 Configurable Settings

Regulatory Compliance Power Supply:

- North America 600 ohms, European CTR21
- FCC (Part 15, Class B) , CE, ICES-003
- Switching Type (100-240v) Automatic
- DC Input Voltage: +5 VDC at 2.0 A Max.
- Power Consumption: 5 Watts
- Power Adapter: 100-240v - 50-60Hz (26-34VA) AC Input, 1.8m cord

Indicator Lights/LED:

- Power, Ethernet (WAN), Phone 1, Phone 2

Documentation:

- Quick Installation, User, and configuration guides are downloadable from www.linksys.com
- Administration guide – Service providers only
- Provisioning guide – Service providers only

Dimensions/Weight:

- 3.98 x 3.98 x 1.10in (101 x 101 x 28mm) W x H x D
- 5.29oz (0.15kg)