ProVu Communications
Linksys SPA-921

Affordable, single line, business class VoIP phone

Stylish and functional in design, the SPA-921 VoIP Phone is ideal for a both business and residential use.

Feature-rich

Features on the SPA-921 include a high resolution graphical display, speaker phone, and a 2.5 mm head-set port. The SPA-921 supports one line with two simultaneous calls. This line can be configured as a unique phone number (or extension), or can be configured to share a number that is assigned to multiple phones.

Ideal for ITSP's

The SPA-921 provides secure software upgrades, enabling internet service providers to deliver high quality support to their subscribers. Remote provisioning also saves service providers the hassle and expense of managing, pre-loading, and re-configuring customer premise equipment (CPE).

With plenty of features, the SPA-921 addresses the requirements of traditional business users whilst using the advantages of IP telephony.

Key Features

- 1 Voice line with two call appearances
- 128 x 64 monochrome LCD display
- Menu driven user interface
- Hands free option Call hold, Call waiting, Call transfer, Call conferencing, Do not disturb.
- Call logs- made, answered, missed calls, with time- 60 entries each
- Caller ID
- Multiple ring tones with selectable default ring tone per line
- Personal address book with auto dial- 100 entries
- Speed dialling
- Automated provisioning up to 256 byte encryption
- Multiple ring tones
- Built in web server for administration and configuration

A Division of Cisco Systems, Inc.

ProVu Communications Ltd
Savile Mill, Savile Street, Milnsbridge, Huddersfield, HD3 4PG
Tel: 01484 840048 Fax: 01484 651742
Email: contact@provu.co.uk
www.provu.co.uk
Registration No: 3748320
VAT Registration No: 732754332

ProVu Communications

Linksys SPA-921

Technical Data

Telephony Features:

- One Voice Line with Two Call Appearances
- Backlit Pixel Based Display: 128x64 Monochrome Graphical Liquid Crystal Display (LCD)
- Line Status - Active Line Indication, Name and Number
- Menu Driven User Interface
- Shared Line Appearance **
- Call Hold, Call Waiting, Music On Hold** , Speaker phone
- Caller ID Name and Number and Outbound Caller ID Blocking
- Outbound Caller ID Blocking
- Call Transfer - Attended and Blind, Call Timer
- Three Way Call Conferencing with Local Mixing
- Connects to External Conference Bridge for Multi-party Conferencing
- Automatic Redial of Last Calling and Last CalledNumbers
- Call Pick Up - Selective and Group **
- Call Park and UnPark **
- Call Back on Busy, Call Swap
- Call Blocking - Anonymous and Selective
- Call Forwarding - Unconditional, No Answer, On Busy
- Hot Line and Warm Line Automatic Calling
- Call Logs (60 entries each): Made, Answered, and Missed Calls
- Redial from Call Logs, On Hook Dialling
- Personal Directory with Auto-dial (100 entries)
- Do Not Disturb (callers hear line busy tone)
- Digits Dialled with Number Auto-Completion
- Anonymous Caller Blocking
- On Hook Default Audio Configuration (Speaker phone and Headset)
- Multiple Ring Tones with Selectable Ring Tone per Line
- Called Number with Directory Name Matching
- Call Number using Name - Directory Matching or via Caller ID
- Subsequent Incoming Calls with Calling Name and Number
- Date and Time with Intelligent Daylight Savings Support
- Call Duration and Start Time Stored in Call Logs
- Name and Identity (Text) Displayed at Start Up
- Distinctive Ringing Based on Calling and Called Number
- Ten User Downloadable Ring Tones
- Speed Dialling, Eight Entries
- Configurable Dial/Numbering Plan Support
- Intercom **
- Group Paging **
- NAT Traversal, including STUN Support
- Syslog, Debug, Report Generation, and Event Logging
- Secure Call Encrypted Voice Communication Support
- Built-in Web Server for Administration and Configuration with Multiple Security Levels
- Optionally Require Admin Password to Reset Unit to Factory Defaults

Hardware Features:

- Pixel Based Display: 128x64 Monochrome LCD Screen
- Audio Mute On/Off Button
- Headset On/Off Button
- Speaker phone On/Off Button
- Four Soft Key Buttons
- Voice Mail Message Waiting Indicator Light+Retrieval Button
- Dedicated Hold Button
- Volume Control
- High Quality Handset and Cradle
- Built-In High Quality Microphone and Speaker
- Headset Jack
- 5 volt DC Universal (100-240 Volt) Switching Power Adaptor
- LED Test Function

Data Networking:

- 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)
- 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)
- VLAN Tagging 802.1p/q - Layer 2 QoS
- SNTP - Simple Network Time Protocol (RFC 2030)

Voice Gateway:

- SIPv2 - Session Initiation Protocol Version 2 (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 (including STUN)
- SIPFrag (RFC 3420)
- Secure (Encrypted) Calling via Pre-Standard Implementation of Secure RTP
- Codec Name Assignment
- Voice Algorithms:
- - G.711 (A-law and Î¼-law)
- - G.726 (16/24/32/40 kbps)
- - G.729 A
- - G.723.1 (6.3 kbps, 5.3 kbps)
- Dynamic Payload Support
- Adjustable Audio Frames Per Packet
- DTMF: In-band and Out-of-Band (RFC 2833) (SIP INFO)
- Flexible Dial Plan Support with Inter-Digit Timers
- IP Address / URI Dialing Support
- Call Progress Tone Generation
- Jitter Buffer - Adaptive
- Frame Loss Concealment
- VAD - Voice Activity Detection with Silence Suppression
- Attenuation / Gain Adjustments
- MWI - Message Waiting Indicator Tones
- VMWI - Voice Mail Waiting Indicator - Via NOTIFY, SUBSCRIBE
- Caller ID Support (Name and Number)
- Third Party Call Control (RFC 3725)

Provisioning, administration and maintenance:

- Integrated Web Server Provides Web Based Administration and Configuration
- Telephone Key Pad Configuration via Display Menu / Navigation
- Automated Provisioning and Upgrade via HTTPS, HTTP, TFTP
- Asynchronous Notification of Upgrade Availability via NOTIFY
- Non-intrusive, In-Service Upgrades
- Report Generation and Event Logging
- Statistics Transmitted in BYE Message
- Syslog and Debug Server Records - Configurable Per Line

Physical Interfaces:

- 1 10baseT RJ-45 Ethernet Port (IEEE 802.3)
- Handset: RJ-7 Connector
- Built-in Speaker phone and Microphone
- Headset 2.5 mm Port
- ** Feature requires support by call server

ProVu Communications Ltd
Savile Mill, Savile Street, Milnsbridge, Huddersfield, HD3 4PG
Tel: 01484 840048 Fax: 01484 651742
Email: contact@provu.co.uk
www.provu.co.uk
Registration No: 3748320
VAT Registration No: 732754332

Revision: 44 - 06/02/2007