Linksys SPA921 IP Phone

Product Name: Linksys SPA921 IP Phone

Manufacturer: -

Model Number: SPA-921-R

Availability: Discontinued
Please note that the Linksys SPA-921 has been discontinued. We recommend the Cisco SPA-303 as a good alternative.
 Stylish and functional in design, the SPA-921 IP Phone is ideal for a residence & business using a hosted IP telephony service, an IP PBX, or a large scale IP Centrex deployment.

• The SPA921 leverages industry leading VoIP technology from Linksys to deliver an upgradeable High Quality IP phone that is Unparalleled in features, value, and support.
• Standard features on the SPA921 include a high resolution graphical display, speakerphone, and a 2.5 mm head-set port.
• The SPA-921 supports one line with two call appearances and provides support for three way conferencing, attended call transfer, and placing a call on hold to answer an incoming call.
• The 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.

Features:

• Full featured one-line business class IP phone
• Speakerphone. Caller ID. Call Hold, Transfer, Conferencing, and more
• Easy to install with secure remote provisioning. Menu based and web based configuration.
• Comprehensive Interoperability and SIP Based Feature Set
• Connect directly to an Internet Telephone Service Provider or connect to an IP PBX

Based on the SIP standard, the SPA921 has been tested to ensure comprehensive interoperability with equipment from VoIP infrastructure leaders enabling service providers to quickly roll-out competitive, feature rich services to their customers. With hundreds of features and configurable sevice parameters, the SPA-921 addresses the requirements of traditional business users while leveraging the advantages of IP telephony. Features such as easy station moves, presence,and shared line appearances (across local and geographically dispersed locations) are just some of the many advantages of the SPA921.
Carrier-Grade Security, Provisioning, and Management
The SPA921 uses standard encryption protocols to provide secure remote in-service software upgrades. Linksys secure remote provisioning tools include detailed performance measurement and troubleshooting features, enabling network providers to deliver high quality support to their subscribers. Remote provisioning also saves service providers the hassle and expense of managing, preloading, and re-configuring customer premise equipment (CPE).
Package Contents

• SPA-921 IP Phone, Handset, and Stand
• Handset Cord - 56 cm (26 in)
• RJ45 Ethernet Cable - 1.8 m (6 ft) Cord
• Quick Installation Guide
• 5v Power Adapter - 1.8 m (6 ft) Cord

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 **
• Speakerphone
• Call Hold
• Music on Hold **
• Call Waiting
• Caller ID Name and Number and Outbound Caller ID Blocking
• Outbound Caller ID Blocking
• Call Transfer - Attended and Blind
• Three Way Call Conferencing with Local Mixing
• Connects to External Conference Bridge for Multi-party Conferencing
• Automatic Redial of Last Calling and Last Called Numbers
• On-Hook Dialing
• Call Pick Up - Selective and Group **
• Call Park and UnPark **
• Call Swap
• Call Back on Busy
• 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
• Distinctive Ringing Based on Calling and Called Number
• Ten User Downloadable Ring Tones - Ring Tone Generator Free from www.linksys.com
• Speed Dialing, Eight Entries
• Configurable Dial/Numbering Plan Support
• Intercom **
• Group Paging **
• Personal Directory with Auto-dial (100 entries)
• Do Not Disturb (callers hear line busy tone)
• Digits Dialed with Number Auto-Completion
• Anonymous Caller Blocking
• URI (IP) Dialing Support (Vanity Numbers)
• On Hook Default Audio Configuration (Speakerphone 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
• Call Timer
• Name and Identity (Text) Displayed at Start Up
• NAT Traversal, including STUN Support
• DNS SRV and Multiple A Records for Proxy Lookup and Proxy Redundancy
• 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
• Automated Remote Provisioning, Multiple Methods. Up to 256 Bit Encryption: (HTTP, HTTPS, TFTP)
• Optionally Require Admin Password to Reset Unit to Factory Defaults

** Feature requires support by call server
Hardware Features

• Pixel Based Display: 128x64 Monochrome LCD Graphical Display
• Dedicated Illuminated Buttons for:

• Audio Mute On/Off
• Ethernet LAN - 10BaseT RJ-45
• 5 volt DC Universal (100-240 Volt) Switching Power Adaptor
• LED Test Function
• Speakerphone On/Off
• Four Soft Key Buttons
• Four Way Rocking Directional Knob for Menu Navigation
• Voice Mail Message Waiting Indicator Light
• Voice Mail Message Retrieval Button
• Dedicated Hold Button
• Settings Button for Access to Feature, Set-up, and Configuration Menus
• Headset On/Off
• Volume Control Rocking Up/Down Knob Controls Handset, Headset, Speaker, Ringer
• Standard 12-Button Dialing Pad
• High Quality Handset and Cradle
• Built-In High Quality Microphone and Speaker
• Headset Jack - 2.5 millimeter

Data Networking

• MAC Address (IEEE 802.3)
• 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)
• VLAN Tagging 802.1p/q - Layer 2 QoS
• SNTP - Simple Network Time Protocol (RFC 2030)
• IPv4 - Internet Protocol v4 (RFC 791) upgradeable to v6 (RFC 1883)
• ARP - Address Resolution Protocol
• DNS - A Record (RFC 1706), SRV Record (RFC 2782)

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)
• 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)
• 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) vo - 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)

Provisioning, Administration & Maintenance

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

Physical Interfaces

• 1 10baseT RJ-45 Ethernet Port (IEEE 802.3)
• Handset: RJ-7 Connector
• Built-in Speakerphone and Microphone
• Headset 2.5 mm Port

Power Supply

• Switching Type (100-240v) Automatic
• DC Input Voltage: +5 Volts DC at 2.0 Amps Maximum
• Power Consumption: 5 Watts
• Power Adapter: 100-240v - 50-60Hz (26-34VA) AC Input, 1.8m (6 ft) cord

Indicator Lights/LED

• Four (4) Call Appearance/Line Buttons with Associated Tricolor LED
• Message Waiting Indicator LED
• Voicemail Message Retrieval Button
• Hold Button
• LED Test Function
• Headset On/Off Button with LED
• Mute Button with LED
• Line LED State Indication: Active, Idle, On Hold, Unregistered Speakerphone On/Off Button with LED

Linksys SPA Series VoIP Phone (SIP) Comparison Table

SPA Model

Voice Lines

Ethernet Ports

High Resolution Display

Power Over Ethernet

Mains Power Supply

SPA901

1

1

N

N

Y

SPA921

1

1

Y

N

Y

SPA922

1

2

Y

Y

N

SPA941

2-4

1

Y

N

Y

SPA942

2-4

2

Y

Y

N

SPA962

6

2

Y Colour

Y

N

Price: £64.90