Full Featured IP PBX System for the Small Business and Home Office

The SPA9000 marries the rich feature set of high-end PBX telephone systems with the convenience and cost advantages of Voice over IP. It has common voice system features such as an auto-attendant, shared line appearances, three way call conferencing, intercom, music on hold, call-forwarding and much more. The SPA9000 opens up access to the benefits of VoIP, including low cost long distance service, telephone number portability, and one network for both voice and data.

The SPA9000 is so easy to configure that a fully working system can be set up in minutes. New telephones are automatically detected and registered when they are connected to the SPA9000. The SPA9000 has an integrated web server that allow features to be configured using a web browser. The web server has multiple levels of password protected access to user and service level features. Service level settings may be locked by the Internet Telephone Service Provider to ensure they are not inadvertently corrupted. The Internet Telephone Service Provider also can remotely update the software and settings through a secure encrypted connection.

With its integrated router, the SPA9000 can be either connected directly to the internet connection or to another router on your network. The SPA9000 has separate WAN and LAN Ethernet ports. The WAN connection can be connect through DHCP or a fixed IP address. The LAN port can assign IP addresses to IP telephones and computers using NAT and DHCP.

While the SPA9000 will work with any SIP compatible IP telephone, it is the ideal host for Linksys business telephones, such as the SPA901, SPA921, SPA922, SPA941, and SPA942. Powerful configuration capabilities enable the SPA9000 to support a greater set of advanced features with these telephones, such as shared line appearances, hunt groups, call transfer, call parking lot, and group paging. With its two FXS ports, the SPA9000 can support traditional analog devices such as telephones, answering machines, FAX machines, and media adapters.
Features

- SIP Application Server, Proxy, Registrar and Location Server (RFC3261)
- Multiple Service Provider Lines / SIP Account Support (4)
- Shared Line Appearance (SLA)
- Automated Attendant (AA)
- Configurable AA Answer Delay
- Interactive Voice Response (IVR)
- Recordable IVR Prompts
- Automatic Call Distribution (ACD)
- Configurable Call Routing
  - Least Cost Routing
  - Multiple DID Numbers Per VoIP Line
  - Call Routing to Multiple Extensions or Targeted User
  - Call Hunting - Sequential, Round Robin, Random
- Phone Configuration and Management Server
  - Discovery and Configuration of IP Phones
  - Assignment of Extension
  - Assignment of Dial plan
  - Proxy Logging of SIP Messages
  - Phone Firmware Upgrade Management
- Corporate Directory with Automatic Update
- Configuration and Maintenance via Web Interface (Local or Remote)
  - Status Display of All Connections
- Remote Configuration via
  - HTTPS with XML Formatted Files
  - HTTP or TFTP with 256-Bit Encrypted Binary Files
- Call Park - User Definable Parking Space Number
- Call Unpark
- Call Transfer
- Call Forward
- Group Paging
- Intercom
- Directed Call Pick Up
- Group Call Pick Up
- Music / Information via Streaming Audio Server (SAS) for Calls:
  - On Hold
  - Parked in the Parking Lot
  - Being Transferred
- Simultaneous Ringing (Find Me Service)
- Do Not Disturb
- Voice Mail Integration - Service Provider Based
  - Voice Mail Notification via SUBSCRIBE / NOTIFY
  - Forward Call Directly to Voice mail
- Integrated Media Proxy or Direct RTP Routing to ITSP
- Differentiated Services (DiffServ) / Type of Service (TOS) Support
- Two FXS Ports for Phones, Fax machines, Media Adapters
- Voice encoding according to G.711 (64kbit/s)
- Fax Support using G.711 Pass-Through or T.38
- Echo Cancellation (G.165)

Additional Features when used with SPA Phones

- Line Status - Active Line Indication, Name/Number
- Digits Dialed with Number Auto-Completion
- Call Hold
- Call Waiting
- Call Transfer - Attended and Blind
- Call Conferencing
- Automatic Redial
- Call Pick Up - Selective and Group **
- Call Swap
- Call Forwarding - Unconditional, No Answer, On Busy
PRODUCT DATA

Model No.  SPA9000

- Hot Line and Warm Line Automatic Calling
- Call Log (60 entries each): Made, Answered, Missed Calls
- Personal Directory with Auto-dial (100 entries)
- Do Not Disturb
- URI (IP) Dialing Support (Vanity Numbers)
- On Hook Default Audio Configuration (Hands Free/Headset)
- Multiple Ring Tones with Selectable Default Ring Tone per Line
- Called Number with Directory Name Matching
- Calling Number with 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 with Call Time Stamp Stored in Call Logs
- Name/Identity (Text) Display at Start Up
- Distinctive Ringing Based on Calling and Called Number
- User Downloadable Ring Tones and Ring Tone Generator (Free from www.linksys.com)
- Download on Demand Ring Tones - 10
- Speed Dial Support
- Configurable Dial/Numbering Plan Support - per Line
- 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 Admin and Config with Multiple Security Levels
- Automated Provisioning, Multiple Schemes-Up to 256 Bit Encryption: (HTTP, HTTPS, TFTP)
- Require Admin Password to Reset Unit to factory Defaults Option

** Service feature availability is call feature server platform dependent.

- FCC (Part 15 Class B), CE, A-Tick, ICES-003
- Password Protected System Reset to Factory Default
- Password Protected Admin and User Access Authority
- HTTPS with Factory Installed Client Certificate
- HTTP Digest - Encrypted Authentication via MD5 (RFC 1321)
- Up to 256-bit AES Encryption

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


- 1 - SPA9000 System
- 1 - 5 Volt Power Adapter
- 1 - RJ45 Ethernet Cable
- 1 - Quick Installation

Compliance

Security

LEDs

Documentation

Package Contents

Environmental

Dimensions  3.98 x 3.98 x 1.1 in (101 x 101 x 28 mm) W x H x D
Unit Weight  5.3 oz (0.15 kg)
Power  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
Operating Temp.  32°~113°F (0°~45°C)
Storage Temp.  -13°~185°F (-25°~85°C)
Operating Humidity  10~90% Non-condensing
Storage Humidity  10~90% Non-Condensing
### Specifications

#### 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)
- **DHCP Server** - Dynamic Host Configuration Protocol (RFC 2131)
- **PPoE Client** - Point to Point Protocol over Ethernet (RFC 2516)
- **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
- **SNTP** - Simple Network Time Protocol (RFC 2030)
- **PPPoE** Client - Point to Point Protocol over Ethernet (RFC 2516)

#### Voice Gateway
- **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 Support**
- **Adjustable Audio Frames Per Packet**
- **DTMF:** In-band & 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 w/ Silence Suppression
- **Attenuation / Gain Adjustments**
- **MWI** - Message Waiting Indicator Tones
- **VMWI** - Via NOTIFY, SUBSCRIBE
- **Caller ID Support** (Name & Number)
- **Web Browser Administration & Configuration via Integral Web Server**
- **Telephone Key Pad Configuration** of Select Networking Parameters via IVR
- **Web Browser 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:
- **2 10/100BaseT RJ-45 Ethernet Port (IEEE 802.3) -- 1 WAN, 1 LAN**
- **2 RJ-11 FXS Phone Ports** - For Analog Circuit Telephone Device (Tip/Ring)
- **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 Switching Type (100-240v) Automatic