Sipura Technology, Inc.

SPA Administration Guide

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Sipura Technology, Inc.
SPA Administration Guide

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1. Product Description

This guide describes basic administration and use of the Sipura Technology SPA phone adapter – an intelligent low-density Voice over IP (VoIP) gateway. The SPA enables carrier class residential and business IP Telephony services delivered over broadband or high-speed Internet connections. By intelligent, we mean the SPA maintains the states of all the calls it terminates. It is capable of making proper decisions in reaction to user input events (such as on/off hook or hook flash) with little or no involvement by a 'middle-man' server or media gateway controller.

Examples of proper reactions are: playing dial tone, collecting DTMF digits, comparing them against a dial plan and terminating a call. With intelligent endpoints at the edges of a network, performing the bulk of the call processing duties, the deployment of a large network with thousands of subscribers can scale quickly without the introduction of complicated, expensive servers. As described later in this section, the Session Initiation Protocol (SIP) is a good choice of call signaling protocol for the implementation of such a device in this type of network.

1.1. Introduction

The phenomenal growth of broadband Internet access (DSL, Cable, FTTH, etc.), has brought the realization of reliable packet switched IP Telephony Services with circuit switched toll-quality and subscriber feature transparency with that of the PSTN's CLASS feature-set. In addition to basic offerings comparable to traditional PSTN services, many service providers have integrated their IP Telephony offering with a large number of web-based productivity applications like unified messaging and call management features such as, remote call forward configuration via the web. Such advances over traditional phone services, with equal or better voice quality and lower per-minute prices, have made IP Telephony service a viable business. In fact, IP Telephony service providers in the US and abroad have seen their subscriber base growing at a rapid pace.

Important!! Please note: The information contained herein is not a warranty from Sipura Technology, Inc. Customers planning to use the SPA in a VoIP service deployment are warned to test all functionality they plan to support in conjunction with the SPA before putting the SPA in service. Some information in Section 1 of this guide is written for educational purposes and describes functionality not yet implemented in the SPA.

1.2. Large-Scale Deployment of VoIP Endpoints

The technical challenges in deploying and operating a residential IP Telephony service, however, are not small. One of the main challenges is to make the service transparent to subscribers: The subscribers shall expect to use their existing phones to make or receive calls in the same way as with the existing PSTN service. To enable this level of transparency, the IP Telephony solution has to be tightly integrated. A key element in this end-to-end IP Telephony solution is the provision of an endpoint device that sits at a subscriber's premises that serves as an IP Telephony gateway or telephone adapter. This phone adapter offers one or more standard telephone RJ-11 phone ports – identical to the phone wall jacks at home – where the subscriber can plug in their existing telephone equipment to access phone services. The IP Telephony gateway may connect to the IP network, like the Internet, through an uplink Ethernet connection.

1.2.1. Voice Quality Overview

Voice Quality perceived by the subscribers of the IP Telephony service should be indistinguishable from that of the PSTN. Voice Quality can be measured with such methods as Perceptual Speech Quality Measurement (PSQM) (1-5 – lower is better) and Mean Opinion Score (MOS) (1-5 – higher is better).

The table below displays speech quality metrics associated with various audio compression algorithms:

Algorithm	Bandwidth	Complexity	MOS Score
G.711	64 kbps	Very Low	4.5
G.726	16, 24, 32, 40 kbps	Low	4.1 (32 kbps)
G.729a	8 kbps	Low - Medium	4
G.729	8 kbps	Medium	4
G.723.1	6.3, 5.3 kbps	High	3.8

Please note: The SPA supports all the above voice coding algorithms.

Several factors that contribute to Voice Quality are described below.

Audio compression algorithm – Speech signals are sampled, quantized and compressed before they are packetized and transmitted to the other end. For IP Telephony, speech signals are usually sampled at 8000 samples per second with 12-16 bits per sample. The compression algorithm plays a large role in determining the Voice Quality of the reconstructed speech signal at the other end. The SPA supports the most popular audio compression algorithms for IP Telephony: G.711 a-law and µ-law, G.726, G.729a and G.723.1.

The encoder and decoder pair in a compression algorithm is known as a codec. The compression ratio of a codec is expressed in terms of the bit rate of the compressed speech. The lower the bit rate, the smaller the bandwidth required to transmit the audio packets. Voice Quality is usually lower with lower bit rate, however. But Voice Quality is usually higher as the complexity of the codec gets higher at the same bit rate.

Silence Suppression – The SPA applies silence suppression so that silence packets are not sent to the other end in order to conserve more transmission bandwidth; instead a noise level measurement can be sent periodically during silence suppressed intervals so that the other end can generate artificial comfort noise that mimics the noise at the other end (using a CNG or comfort noise generator).

Packet Loss – Audio packets are transported by UDP which does not guarantee the delivery of the packets. Packets may be lost or contain errors which can lead to audio sample drop-outs and distortions and lowers the perceived Voice Quality. The SPA applies an error concealment algorithm to alleviate the effect of packet loss.

Network Jitter – The IP network can induce varying delay of the received packets. The RTP receiver in the SPA keeps a reserve of samples in order to absorb the network jitter, instead of playing out all the samples as soon as they arrive. This reserve is known as a jitter buffer. The bigger the jitter buffer, the more jitter it can absorb, but this also introduces bigger delay. Therefore the jitter buffer size should be kept to a relatively small size whenever possible. If jitter buffer size is too small, then many late packets may be considered as lost and thus lowers the Voice Quality. The SPA can dynamically adjust the size of the jitter buffer according to the network conditions that exist during a call.

Echo – Impedance mismatch between the telephone and the IP Telephony gateway phone port can lead to near-end echo. The SPA has a near end echo canceller with at least 8 ms tail length to compensate for impedance match. The SPA also implements an echo suppressor with comfort noise generator (CNG) so that any residual echo will not be noticeable.

Hardware Noise – Certain levels of noise can be coupled into the conversational audio signals due to the hardware design. The source can be ambient noise or 60Hz noise from the power adaptor. The SPA hardware design minimizes noise coupling.

End-to-End Delay – End-to-end delay does not affect Voice Quality directly but is an important factor in determining whether subscribers can interact normally in a conversation taking place over an IP network. Reasonable delay figure should be about 50-100ms. End-to-end delay larger than 300ms is unacceptable to most callers. The SPA supports end-to-end delays well within acceptable thresholds.

1.3. The Session Initiation Protocol

1.3.1. Why SIP?

There are many excellent articles and books that discuss the advantages of SIP.[i] Here are some of the more popular details:

- SIP message constructs are very similar to those of HTTP which is well-known to be IP Network (Internet) friendly.
- SIP is transport agnostic – meaning it can be used over TCP/IP or UDP/IP, with or without security.
- SIP has a better chance of punching through NAT than other control protocols.
- SIP enables the implementation of intelligent endpoints to support scalable advanced services.

In a nutshell, SIP is a distributed signaling protocol (as opposed to a centralized protocol such as SS7, MGCP or MEGACO/H.248). With a distributive protocol, the intelligence does not necessarily reside on a central server, but can be built into the individual endpoints. By moving the intelligence to reside within the endpoints at the edge of the network, the processing load of the network application and associated call servers are significantly reduced, thus making the network a very scalable solution.

1.3.2. Components of a SIP Network

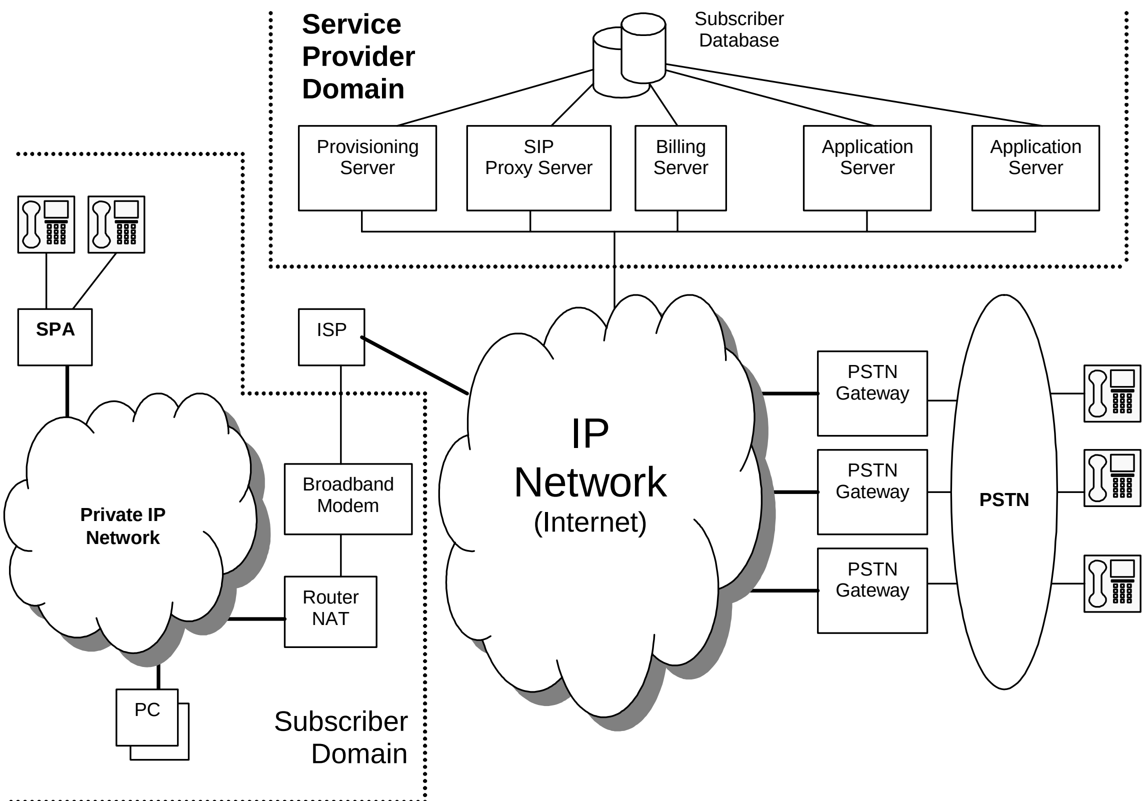

Figure 1 -- Components of a SIP IP Telephony Network

IP Telephony Gateway (SPA): The SPA is a small device that sits at the subscriber's premises. It converts between analog telephone signals and IP Telephony signals. It has up to two RJ-11 ports where standard analog telephones can be directly attached, and an RJ-45 interface for the Ethernet connection to the home or business LAN. Intelligence can be built into this device to provide a wide range of features to the subscribers in association with the other elements in the service. The SPA functions as a SIP User Agent (UA).

Home/SOHO Routers with NAT Functionality: A home/SOHO router is used for routing IP packets between the subscriber's private network and the ISP's public network. If the ISP provides only one public IP address to the subscriber, the devices attached to the private network will be assigned private IP addresses and the router will perform network address translation (NAT) on packets sent from the private network to the public network via the router. Home routers offer the following features:

- An R-J45 WAN interface for connection to the ISP's public network and one or more RJ-45 LAN interfaces for connection to the subscriber's private network. The router directs packets between the private network and the public network.
- A PPPoE client to connect with the ISP through a DSL modem.
- A DHCP client where the router will obtain an IP address, subnet mask, default router assignment, etc., for its WAN interface from a DHCP server on the public network.
- A DHCP server for auto-assignment of private IP addresses, subnet mask, and default router assignment to devices attached to the private network, i.e. computers, IP Telephony

gateways, etc. The default router in this case is the IP address of the LAN interface of the router itself.

- Performs NAT on packets sent from the private network to the public network. This is an important feature such that recipients of the private packets will perceive them as originated from a public IP address (the router's WAN interface) and will therefore return messages to the proper public IP address and port. Different routers may use different rules for allocating port numbers at the WAN interface to forward packets from a private IP address/port to a public IP address/port. The allocated port number is also used for routing packets from external IP addresses to a private address. Most routers will accept a number of static port mapping rules for forwarding packets received on a specific port at the WAN interface to a specific IP address/port in the private network.

PSTN - VoIP Gateways: These devices are required if user agents are expected to make calls to or receive calls from the PSTN. Many gateways may be deployed in order to service a wide area. Gateways also behave like SIP user agents. The proxy server can be configured with cost-saving rules based call routing information so that it may decide which gateway to use depending on the destination and the time of the call. The IP Telephony service provider will assign each subscriber an E164 telephone number so that it may be reached from the PSTN just like any other telephone.

Billing Servers: Billing servers are used to generate billing data per usage of the IP Telephony service. Typically, the service provider will charge a flat fee for unlimited calls between IP Telephony subscribers (on-net-to-on-net calls). Per use or minute chargers will be incurred only when the subscriber makes calls to PSTN numbers (on-net-to-off-net calls) through one of the PSTN gateways. CDR (call detail record) data are generated by the PSTN gateway and sent to the billing servers.

Provisioning Servers: Provisioning servers are used to provision the subscriber user agent devices, e.g. the SPA. When a subscriber signs up for IP Telephony service, he selects an appropriate service level and enters his personal information including billing information. This information is processed by the provisioning server and stored into the service provider's customer database. The provisioning server generates a device profile based on the subscriber's choice of options. The device profile, which is list of configuration parameters, is downloaded into the SPA from the provisioning server. The SPA can be configured to contact the provisioning server periodically to check for any update of the device profile, which may include a firmware upgrade or configuration modification to the SPA.

Application Servers: Application servers are used to provide value added services, such as call forwarding, outgoing or incoming call blocking

Voice Mail Servers: Specialized servers provide voice mail services to the IP Telephony service subscribers. When the subscriber is busy or the SPA is out of service for maintenance or other reason, incoming calls to the subscriber may be redirected to the voice mail servers where the caller can leave a voice mail. The voice mail server will then notify the subscriber's SPA of the availability of voice mail(s) in his mailbox. The subscriber can then contact the voice mail server to retrieve his voice mail(s). The SPA can indicate the message-waiting status to the subscriber through a number of methods such as stuttered dial tone heard through the telephone every time the subscriber lifts up the handset until the voice mail is retrieved.

1.3.3. Provisioning Overview

The SPA is configurable in many ways such that it can provide a wide range of customizable services and operate in many diverse environments with a variety different vendors' SIP Proxy Servers, VoIP Gateways, Voice Mail Servers, NAT applications, etc. Provisioning is the process by which the SPA obtains a set of configuration parameters in order for it to operate in the Service Provider's network.

The complete set of configuration parameters for an SPA corresponding to an individual subscriber is referred to as a configuration profile or simply a Profile. The Profile can be encoded as an XML file or a simple plain text file with a list of tag/value pairs. When the SPA unit is shipped from the factory, it contains a default common Profile and is considered Unprovisioned. To save costs and expedite delivery, however, it is very desirable that an Unprovisioned unit can be shipped directly from the factory to the subscriber's location without any preprocessing by the Service Provider.

The SPA contacts the Service Provider's provisioning server via the IP network or Internet when it is plugged into the subscriber's home or business Local Area Network (LAN) – assuming the provisioning server is reachable from the subscriber's home network – to pull the designated profile to be installed in that particular SPA unit. Furthermore, the SPA unit will periodically contact the provisioning server to download an updated profile. The protocol for downloading the configuration profile can be "clear text" TFTP or HTTP data or it can be encrypted TFTP, HTTP or HTTPS data if security is required. Security will be discussed in more details in a later section.

This type of autonomous remote provisioning, where the individual SPA unit pulls the profile from the provisioning server is very scalable and flexible. Using this provisioning method, a large number of SPA units can be provisioned simultaneously and updated periodically.

However, some basic information must be provided to the SPA before it can be provisioned in this fashion: a) the IP address or domain name of the provisioning server to contact, and b) an ID and/or a password to send to the provisioning server such that it can associate it with a specific subscriber and obtain the corresponding profile. This information can be sent out-of-band to the subscriber via secured email or in a letter inside a welcome kit, for example. The subscriber might need to punch in some numbers using a telephone connected to the SPA in order to enter this information into the unit. The SPA provides an easy-to-use interface with audio instructions to make this initial configuration process as painless as possible. An alternative is for the unit to be provisioned with this basic information by the Service Provider before the unit is shipped to the subscriber.

In addition to the batch mode of remote provisioning, the SPA allows an interactive mode of local provisioning. One way to offer this feature is through the use of an IVR system (accessed through an attached telephone set). The user can access a diagnostic or configuration menu to check the status of the device or to change some of the settings. This method of provisioning may be applied by an administrator when the device is at the Service Provider's office, or by the subscriber under the guidance of trained personnel during over-the-phone troubleshooting.

A third method of entering provisioning information into the SPA is by way of its integral web server via a browser on a PC. The subscriber has the option to set and adjust configuration parameters via an easy-to-use, password protected graphical user interface. This method of provisioning might be preferred by administrators who wish to access the SPA over a secure corporate/institutional LAN or by the residential subscriber who is a "power user."

1.3.4. Security Overview

Security may be applied at many levels in the context of the SPA. The following are examples of information that should be secured:

- The configuration profile pulled from the provisioning server – The downloading of the profile should be secured since it contains authentication (password/user name ID / number) information for accessing subscriber telephony services. It may also contain other passwords and/or encryption keys used for a variety of management and service operations.
- The administration password to the SPA unit – The unit must disallow access to administrative functions to unauthorized users. This access can be controlled with an administrator password. The administrator password can be one of the parameters in the SPA configuration profile.
- The SIP signaling messages – The SIP messages exchanged between the SIP proxy server and the SPA should be encrypted with a secret key. This can be achieved, for instance, by transporting SIP over TLS.

- RTP packets – The RTP payload exchanged between SIP user agents can be encrypted with a secret key to protect against eavesdropper. The secret key can be negotiated with proper SIP signaling messages. Hence the signaling path must be secured also.

1.3.4.1. Proxy Servers

Proxy servers handle two functions:

1. Accept registrations from the SIP user agents,
2. Proxy requests and responses between user agents.

Registration is the process by which a user agent tells the proxy who it is and at what IP address and port that it can be reached via SIP. Registration usually expires within a finite period (e.g., 60s or 3600s) and the UA shall renew their registration periodically before the last registration expires. When a user agent initiates a call, it sends a SIP INVITE request to the proxy server and indicates the target recipient of the call. The proxy server then consults a database to determine where to forward the request to the destination user agent. The proxy server can request authentication credentials from the user agent before granting the service. The credentials are computed by the user agent based on a pre-provisioned password and a challenge “nonce” dynamically generated by the proxy server per request. This mechanism prevents unauthorized user agents from getting IP Telephony services through the proxy server. SIP proxy servers are operated by the IP Telephony service provider and resides at the service provider’s domain. They may be implemented in many different ways. They can be stateless, stateful, or B2BUA. Stateless proxies do not maintain states of each call; they simply proxy the requests and responses between the user agents. Hence they are the simplest, most scalable, but provide the least types of services. Advanced IP Telephony services are possible with these proxies only with intelligent user agent devices that are capable of delivering these services without proxy intervention. Stateful proxies maintain the call state of each call and can provide more intelligent services at the expense of more processing load per call. B2BUA proxies process every request and response from the user agents and are capable of providing very advance services even with relatively simple user agent devices. Obviously B2BUA proxies have the highest processing load per call.

1.3.5. SIP Services

Today’s PSTN offers a large number of enhanced services in addition to basic phone services. Most of the services offered by the PSTN are accessed by the subscribers through their telephone sets. The subscribers provide their input by talking into the handset, pressing the keypad, the switch hook or flash button, while the PSTN presents instructions/information/confirmation to the subscribers through a variety of audio tones, beeps and/or announcements. The SPA supports a comparable range of services via a similar user interface in order to make the IP Telephony service transparent to subscribers.

The SPA is fully programmable and can be custom provisioned to emulate just about any traditional telephony service available today. This ability to transparently deliver legacy services over an IP network coupled with the availability of Internet connected devices (PCs. PDA, etc.) and browsers opens up a new world of potential offerings that a provider can use to differentiate their service and grow their business.

The following is a list of commonly supported phone services:

1.3.5.1. Basic Services

1.3.5.1.1. Making Calls to PSTN and IP Endpoints

This is the most basic service. When the user picks up the handset, the SPA provides dial tone and is ready to collect dialing information via DTMF digits from a touch tone telephone. While it is possible to support overlapped dialing within the context of SIP, the SPA collects a complete phone number and sends the full number in a SIP INVITE message to the proxy server for further call processing. In order to minimize dialing delay, the SPA maintains a dial plan and matches it against the cumulative number entered by the user. The SPA also detects invalid phone numbers not compatible with the dial plan and alerts the user via a configurable tone (reorder) or announcement.

1.3.5.1.2. Receiving Calls from PSTN and IP Endpoints

The SPA can receive calls from the PSTN or other IP Telephony subscribers. Each subscriber is assigned an E.164 phone number so that they may be reached from wired or wireless callers on the PSTN. The SPA supplies ring voltage to the attached telephone set to alert the user of incoming calls.

1.3.5.2. Enhanced Services

Enhanced Services are provided in addition to Basic calling services and accessed by way of a touchtone phone through a series of menus. Since the service enabled by the SPA are Internet in nature, these enhanced services can be made better by offering users a web browser based interface to control certain aspects of some or all services.

1.3.5.2.1. Caller ID

In between ringing bursts, the SPA can generate a Caller ID signal to the attached phone when the phone is on-hook.

Calling Line Identification Presentation (CLIP)

Some subscribers will elect to always block their Caller ID information, yet there may be a circumstance where sending Caller ID information for a particular call is desired, i.e. trying to reach a party that does not accept Caller ID blocked calls.

The subscriber activates this service to send his Caller ID when making an outgoing call. To activate the service, the subscriber enters the corresponding * or # code prior to making the call. This service is in effect only for the duration of the current call.

Calling Line Identification Restriction (CLIR) – Caller ID Blocking

The subscriber activates this service to hide his Caller ID when making an outgoing call. To activate the service, the subscriber enters the corresponding * or # code prior to making the call. This service is in effect only for the duration of the current call.

1.3.5.2.2. Call Waiting

The subscriber can accept a call from a 3rd party while engaging in an active call. The SPA shall alert the subscriber for the 2nd incoming call by playing a call waiting tone.

Disable or Cancel Call Waiting

By setting the corresponding configuration parameter on the SPA, the SPA supports disabling of call waiting permanently or on a per call basis.

Call-Waiting with Caller ID

In between call waiting tone bursts, the SPA can generate a Caller-ID signal to the attached phone when it is off hook.

1.3.5.2.3. Voice Mail

Message Waiting Indication

Service Providers may provide voice mail service to their subscribers. When voice mail is available for a subscriber, a notification message will be sent from the Voice Mail server to the SPA. The SPA indicates that a message is waiting by, playing stuttered dial tone (or other configurable tone) when the user picks up the handset.

Checking Voice Mail

The SPA allows the subscriber to connect to their voice mail box by dialing their personal phone number.

1.3.5.2.4. Call Transfer

Three parties are involved in Call Transfer: The transferor, transferee, and transfer target. There are 2 flavors of call transfer: Attended Transfer (Transfer with consultation) and Unattended Transfer ("Blind" Transfer).

Attendant Transfer

The transferor dials the number of the transfer target, then he hangs up (or enters some * or # code) when the transfer target answers or rings to complete the transfer.

Unattended or "Blind" Transfer

The transferor enters some * or # code and then dials the number of the transfer target to complete the transfer (without waiting for the target to ring or answer).

1.3.5.2.5. Call Hold

Call Hold lets you put a caller on hold for an unlimited period of time. It is especially useful on phones without the hold button. Unlike a hold button, this feature provides access to a dial tone while the call is being held.

1.3.5.2.6. Three-Way Calling

The subscriber can originate a call to a 3rd party while engaging in an active call.

1.3.5.2.7. Three-Way Ad-Hoc Conference Calling

The SPA can host a 3-way conference and perform 3-way audio mixing (without the need of an external conference bridge device or service).

1.3.5.2.8. Call Return

The SPA supports a service that allows the SPA to automatically dials the last caller's number.

1.3.5.2.9. Call Return on Busy

If the last called number is busy, the subscriber can order this service to monitor the called party and to receive a notification from the SPA (such as special phone ring) when that party becomes available.

1.3.5.2.10. Automatic Call Back

This feature allows the user to place a call to the last number they tried to reach whether the call was answered, unanswered or busy by dialing an activation code.

1.3.5.2.11. Call Forwarding

These services forward all the incoming calls to a static or dynamically configured destination number based on three different settings. These services may be offered by the SPA or by the SIP proxy server. They can be activated by entering certain * or # code, followed by entering a telephone number to forward calls to. The SPA provides audio instructions to prompt the user for a forwarding number and confirms that the requested service has been activated.

Call FWD – Unconditional

All calls are immediately forwarded to the designated forwarding number. The SPA will not ring or provide call waiting when Call FWD – Unconditional is activated.

Call FWD – Busy

Calls are forwarded to the designated forwarding number if the subscriber's line is busy because of the following; Primary line already in a call, primary and secondary line in a call or conference.

Call FWD - No Answer

Calls are forwarded to the designated forwarding number after a configurable time period elapses while the SPA is ringing and does not answer.

1.3.5.2.12. Anonymous Call Blocking

By setting the corresponding configuration parameter on the SPA, the subscriber has the option to block incoming calls that do not reveal the caller's Caller ID.

1.3.5.2.13. Distinctive / Priority Ringing

The SPA supports a number of ringing and call waiting tone patterns to be played when incoming calls arrive. The choice of alerting pattern to use is carried in the incoming SIP INVITE message inserted by the SIP Proxy Server (or other intermediate application server in the Service Provider's domain).

1.3.5.2.14. Speed Dialing

The SPA supports speed dialing of up to eight (8) phone numbers or IP addresses. To enter a telephone number speed dial using a touch tone telephone, the user dials a feature code (*74), followed by a number (2-9), then the destination speed dialed target number. When the user wishes to speed dial a target number, they press the corresponding speed dial assigned number followed by the "#" (pound) key.

Users may also enter/review speed dials from User1/User2 web-pages. This interface or similar is required to enter IP address targets.

1.3.5.3. PSTN Interworking

The SPA is designed to provide a transparent interworking relationship with the PSTN. Service providers can deploy the SPA in such a way that PSTN endpoints – wired or wireless – communicating with SPA endpoints do so without modification to their configuration or network settings.

The service provider may choose to deploy a multi-protocol VoIP network, much the same way the PSTN supports multiple signaling schemes today. Most telecommunication providers operate equipment that supports CAS or channel associated signaling, ISDN signaling and SS7 signaling. When VoIP is introduced or used in the telecommunications landscape, it is likely that the service provider will implement a signaling gateway that supports multiple IP Telephony protocols along with legacy PSTN protocols. The signaling gateway is commonly referred to as a Softswitch.

Architecture and functionality can vary greatly amongst the different softswitch vendors. The protocols used will depend on the types of connections that will be set-up across the service provider's network. If the provider is simply providing transport of calls to/from their network to another provider's network, but not originating or terminating calls with the endpoints, SIP will likely be used for softswitch to softswitch communication.

If the service provider is offering origination and/or termination on endpoint equipment then it is very likely that the softswitch chosen for network operations will support multiple PSTN and VoIP signaling protocols.

The table below lists the most commonly accepted, de-facto standards used when implementing a VoIP signaling scheme based on the type of gateway or endpoint equipment being deployed:

VoIP Equipment Type	Typical Port Density	De-Facto Signaling Standards
Trunking Gateways	Greater Than 500 Ports	H.248-Megaco / MGCP / IPDC
Access Gateways	Between five and 500 Ports	SIP / H.323

PBX/KTS Platforms	Between ten and 500 Ports	SIP / H.323 / SCCP
PBX/KTS Telephone Sets	One Port	SIP / MGCP / SCCP
Phone Adapters and IP Centrex Phones	Up to four Ports	SIP / MGCP

The SPA supports SIP today. It has the capability to communicate with a variety of endpoints and signaling entities via SIP messages.

1.4. Network Address Translation (NAT) Traversal

1.4.1. Why NAT?

A NAT allows multiple devices to share the same external IP address to access the resources on the external network. The NAT device is usually available as one of the functions performed by a router that routes packets between an external network and an internal (or private) one. A typical application of a NAT is to allow all the devices in a subscriber's home network to access the Internet through a router with a single public IP address assigned by the ISP. The IP header of the packets sent from the private network to the public network can be substituted by the NAT with the public IP address and a port selected by the router according to some algorithm. In other words, recipient of the packets on the public network will perceive the packets as coming from the external address instead of the private address of the device where the packets are originated.

In most Internet protocols, the source address of a packet is also used by the recipient as the destination to send back a response. If the source address of the packets sent from the private network to the public network is not modified by the router, the recipient may not be able to send back a response to the originator of the message since its private source IP address/port is not usable. When a packet is sent from a device on the private network to some address on the external network, the NAT selects a port at the external interface from which to send the packet to the destination address/port. The private address/port of the device, the external address/port selected by the NAT to send the packet, and the external destination address/port of the packet form a NAT *Mapping*.

The mapping is created when the device first sends a packet from the particular source address/port to the particular destination address/port and is remembered by the NAT for a short period of time. This period varies widely from vendor to vendor; it could be a few seconds, or a few minutes, or more, or less. While the mapping is in effect, packets sent from the same private source address/port to the same public destination address/port is reused by the NAT. The expiration time of a mapping is extended whenever a packet is sent from the corresponding source to the corresponding destination.

More importantly, packets sent from that public address/port to the external address/port of the NAT will be routed back to the private address/port of the mapping session that is in effect. Some NAT devices actually reuse the same mapping for the same private source address/port to any external IP address/port and/or will route packets sent to its external address/port of a mapping from any external address/port to the corresponding private source address/port. These characteristics of a NAT can be exploited by an SPA to let external entities send SIP messages and RTP packets to it when it is installed on a private network.

1.4.2. VoIP-NAT Interworking

In the case of SIP, the addresses where messages/data should be sent to an SPA are embedded in the SIP messages sent by the device. If the SPA is sitting behind a NAT, the private IP address assigned to it is not usable for communications with the SIP entities outside the private network. The SPA must substitute the private IP address information with the proper external IP address/port in the mapping chosen by the underlying NAT to communicate with a particular public peer address/port. For this the SPA needs to perform the following tasks:

- Discover the NAT mappings used to communicate with the peer. This could be done with the help of some external device. For example a server could be deployed on the external network such that the server will respond to a special NAT-Mapping-Discovery request by sending back a message to the source IP address/port of the request, where the message will contain the source IP address/port of the original request. The SPA can send such a request when it first attempts to communicate with a SIP entity in the public network and stores the mapping discovery results returned by the server.
- Communicate the NAT mapping information to the external SIP entities. If the entity is a SIP Registrar, the information should be carried in the Contact header that overwrites the private address/port information. If the entity is another SIP UA when establishing a call, the information should be carried in the Contact header as well as in the SDP embedded in SIP message bodies. The VIA header in outbound SIP requests might also need to be substituted with the public address if the UAS relies on it to route back responses.
- Extend the discovered NAT mappings by sending keep-alive packets. Since the mapping is only alive for short period, the SPA continues to send periodic keep-alive packets through the mapping to extend its validity as necessary.

1.5. SPA Hardware Overview

The SPA has one of the smallest form factors on the market. It can be installed in minutes as a table-top or wall mount CPE device. The images below show the SPA-2000. The SPA-1000 and SPA-3000 are similar to size and shape – the only difference being the color of the adapter.

Figures Figure 2, Figure 3, Figure 4 and Figure 5 show the front, rear, left side and right side of the SPA-2000, respectively.

Figure 2 – SPA-2000 Front

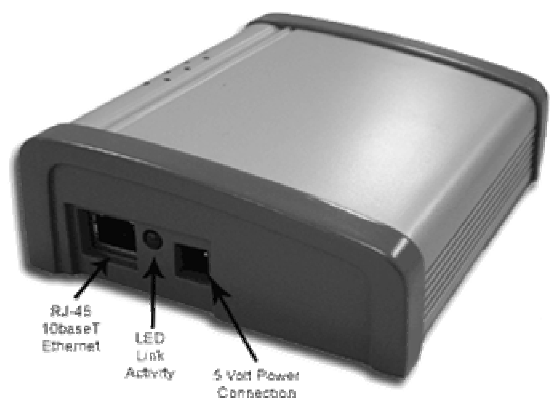

Figure 3 – SPA-2000 Left Side

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technology, inc.

Figure 4 – SPA-2000 Rear

Figure 5 – SPA-2000 Right Side

The SPA has the following interfaces for networking, power and visual status indication:

1. Two (2) RJ-11 Type Analog Telephone Jack Interfaces (Figure 5, above):

These interfaces accept standard RJ-11 telephone connectors. An Analog touchtone telephone or fax machine may be connected to either interface. If the service supports only one incoming line, the analog telephone or fax machine should be connected to port one (1) of the SPA. Port one (1) is the outermost telephone port on the SPA and is labeled “Phone 1.”

The SPA-3000 has an RJ-11 interface labeled “Line” which can be used to connect the adapter to a PSTN analog telephone circuit.

2. One LED for Unit Status (Figure 5, above):

This LED indicates status via the following behaviors:

ON – LED remains solid on

OFF – LED remains solid off

LONG (Long On) – 3.0s on, 1s off continuously

FAST – 0.1s on, 0.1s off continuously

SLOW – 0.5s on, 0.5s off continuously

VSLO (Very Slow) – 1.0s on, 1.0s off continuously

HB (Heart Beat) – 0.1s on, 0.1s off, 0.1s on, 1s off continuously

HB2 (Heart Beat 2) - 0.1s on, 0.1s off, 0.1s on, 0.1s off, 0.1s on, 1.2s off continuously

ERR0(Error 0) - 0.5s on, 0.3s off, 0.1s on, 0.1s off, 0.1s on, 2s off continuously

ERR1(Error 1) – 0.1s on, 0.1s off, 0.1s on, 0.1s off, 0.5s on, 2s off continuously

ERR2(Error 2) – 0.1s on, 0.1s off, 0.1s on, 0.1s off, 0.5s on, 0.2s off, 0.5s on, 2s off continuously

3. One Ethernet 10baseT RJ-45 Jack Interface (

Figure 3, above):

This interface accepts a standard or crossover Ethernet cable with standard RJ-45 connector. For optimum performance, Sipura Technology recommends that a Category 5 cable or greater be used in conjunction with the SPA.

4. One LED for Data Link and Activity (

Figure 3, above):

This LED indicates status via the following behaviors:

ON – LED remains solid on

OFF – LED remains solid off

FAST – 0.125s on, 0.125s off continuously

SLOW – 0.5s on, 0.5s off continuously

Variable Blink – LED blinks according to packet traffic activity

5. One 5 Volt Power Adapter Interface (

Figure 3, above)

This interface accepts the SPA power adapter that came with the unit. Sipura Technology does not support the use of any other power adapters other then the power adapter that was shipped with the SPA unit.

2. Installation Overview

Please check to make sure that you have the following package contents:1. Sipura Phone Adapter Unit2. Ethernet Cable
3. RJ-11 Phone Cable (SPA-3000 Only)4. SPA Quickstart Guide5. 5 Volt Power AdapterYou will also need:1. One or Two Analog Touch Tone Telephones (or Fax Machine)2. Access to an IP Network via an Ethernet Connection
3. Access to a PSTN network connection – SPA-3000 only.

Please observe the following steps to install the SPA.From the Left Side of the SPA:1. Insert a standard RJ-45 Ethernet cable (included) into the LAN port.2. Insert the power adapter cable into the 5V power adapter cable receptacle. Ensure that the power adapter jack is snugly attached to the SPA.From the Right Side of the SPA:1. Insert a standard RJ-11 telephone cable into the Phone 1 port.2. Connect the other end of the cable to an analog telephone or fax machine.3. Insert a standard RJ-11 telephone cable into the Phone 2 port (Optional).4. Connect the other end of the cable to an analog telephone or fax machine.

Note: Do not connect RJ-11 telephone cable from the SPA-1000 or SPA-2000 to the wall jack to prevent any chance of connection to the circuit switched telco network.You may now insert the plug end of the power adapter into a live power outlet which will power up the SPA.

3. Software Configuration

3.1. Provisioning

Please refer to the Sipura SPA Provisioning Guide document for information pertaining to the implementation of HTTPS provisioning features available with Sipura release 2.0. This document also contains a great deal of information regarding the steps that a typical service provider should take when setting up a provisioning system for large numbers of Sipura analog telephone adapters.

3.1.1. Provisioning Capabilities

The SPA provides for secure provisioning and remote upgrade. Provisioning is achieved through configuration profiles transferred to the device via TFTP, HTTP or HTTPS. The SPA can be configured to resync its internal configuration state to a remote profile periodically and on power up.

Firmware release 2.0 provides 256-bit symmetric key encryption of profiles. In addition, an unprovisioned SPA can receive an encrypted profile specifically targeted for that device without requiring an explicit key. Version 2.0 supports a secure first-time provisioning mechanism using SSL functionality. This functionality is explained later in this section.

Remote upgrade is achieved via TFTP, HTTP or HTTPS. The SPA upgrade logic is capable of automating multi-stage upgrades, in case intermediate upgrades are ever required to reach a future upgrade state from an older release.

General purpose parameters are provided as an additional aid to service providers in managing the provisioning process.

All profile resyncs are attempted only when the SPA is idle, since they may trigger a software reboot.

User intervention is not required to initiate or complete a profile update or firmware upgrade.

3.1.2. Configuration Profile

The SPA configuration profile is a binary file with encoded SPA parameter values and user access permissions for those parameters. By convention, the profile is named with the extension ".cfg" (e.g. spa2000.cfg). The Sipura Profile Compiler tool (SPC) is provided for compiling a plain-text file containing parameter-value pairs into a properly formatted and encrypted CFG file. The spc tool is available from Sipura for the Win32 environment (spc.exe), Linux-i386-elf environment (spc-linux-i386-static) and for the OpenBSD environment.

The syntax of the plain-text file accepted by the release 2.0 profile compiler is a series of parameter-value pairs, with the value in double quotes. Each parameter-value pair is followed by a semicolon, e.g. parameter_name "parameter_value";. If no quoted value is specified for a parameter (or if a parameter specification is missing entirely from the plain-text file) the value of the parameter will remain unchanged in the SPA.

The syntax also controls the parameter's user-level access when using the built-in web interface to the SPA. An optional exclamation point or question mark, immediately following the parameter name, indicates the parameter should be user read-write or read-only, respectively. If neither mark is present, the parameter is made inaccessible to the user from the web interface. Note that this syntax has no effect on the admin-level access to the parameters.

In this way, a service provider is given full control over which parameters become inaccessible, read-only, or read-write following provisioning of the SPA.

If the parameter specification is missing entirely from the plain-text file, the user-level access to the parameter will remain unchanged in the SPA.

If the plain-text file contains multiple occurrences of the same parameter-value specification, the last such occurrence overrides any earlier ones.

Parameter names in the plain-text file must match the corresponding names appearing in the SPA web interface, with the following modifications:

- Inter-word spaces are replaced by underscores '_' (e.g. Multi_Word_Parameter).
- For the SPA, line and user specific parameters use bracketed index syntax to identify which line or user they refer to (e.g. Line_Enable[1] and Line_Enable[2]).

Comments are delimited by a '#' character up to the end-of-line. Blank lines can be used for readability.

```
Parameter_name [ '?' | '!' ] ["quoted_parameter_value_string"] ';'
```

Boolean parameter values are asserted by any one of the values {Yes | yes | Enable | enable | 1}. They are deasserted by any one of the values {No | no | Disable | disable | 0}.

Example of plain-text file entries:

```
# These parameters are for illustration only

Feature_Enable      ! "Enable" ;   # user read-write
Another_Parameter   ? "3600"   ;   # user read-only
Hidden_Parameter      "abc123" ;   # user not-accessible

Some_Entry          ! ;   # user read-write, leave value unchanged
```

Multiple plain text files can be spliced together to generate the source for each CFG file. This is accomplished by the "import" directive: the literal string "import" (placed at the start of a new line) followed by one or more spaces and the file name to splice into the stream of parameter-value pairs. The following example illustrates. File splicing can be nested several files deep.

```
# base.txt contains . . .
Param1 "base value 1" ;
Param2 "base value 2" ;
. . .


# spa1234.txt contains . . .
import base.txt
Param1 "new value overrides base" ;
Param7 "particular value 7" ;
. . .


# The spa1234.txt file above is equivalent to . . .
Param1 "base value 1" ;
Param2 "base value 2" ;
. . .
Param1 "new value overrides base" ;
Param7 "particular value 7" ;
. . .
```

A sample plain-text file, containing default parameter-value and access settings for the SPA can be obtained from the profile compiler tool, using the following command-line arguments.

```
spc –-sample-profile defaults.txt
```

Once a plain-text file has been generated with the desired parameter settings, it needs to be compiled into a binary CFG file. The profile compiler can generate a generic unencrypted CFG file, a targeted CFG file (encrypted for a unique SPA), a generic key encrypted CFG file, or a targeted and key encrypted CFG file.

A generic CFG file (non-targeted) is accepted as valid by any SPA device. A targeted CFG file is only accepted as valid by the SPA device bearing the target MAC address. Targeted CFG files are encrypted with a 128-bit algorithmically generated key, and therefore do not require a key to be issued explicitly. Targeted CFG files provide a basic level of security for remotely locking an otherwise unprovisioned SPA.

Firmware version 2.0 uses symmetric key encryption. Using HTTPS, an SSL channel can be used for initial secure remote provisioning using asymmetric key encryption.

Firmware 2.0 supports RC4 and AES symmetric key algorithms, with keys of up to 256 bits. The key can be specified explicitly as a hex-string, or it can be generated from a password or a quoted pass-phrase. In the case of passwords and pass-phrases, the internally generated key is 128 bits in length.

The following command-line syntax generates a generic and unencrypted CFG file:

```
spc spa2000.txt spa2000.cfg
```

A targeted CFG file (with basic encryption) is specified by supplying the MAC address of the target device:

```
spc –-target 000e08aaa010 spa2000.txt spa2000.cfg
```

An encrypted CFG file requires either a password (or quoted pass-phrase) or a hex-string. The following lines illustrate command-line invocations for various combinations of keys and algorithms.

```
spc –-rc4 –-ascii-key apple4sale spa2000.txt spa2000.cfg
spc –-aes –-ascii-key lucky777 spa2000.txt spa2000.cfg
spc –-aes –-ascii-key “my secret phrase” spa2000.txt spa2000.cfg
spc –-aes –-hex-key 8d23fe7...a5c29 spa2000.txt spa2000.cfg
```

A CFG file can be both targeted and key encrypted, as suggested by the following example:

```
spc –-target 000e08aaa010 –-aes –-hex-key 9a20...eb47 a.txt a.cfg
```

The status messages printed by spc can be suppressed with the “--quiet” command line option. Or they can be redirected to a file, with the “--log file_name” command line option. In the latter case, the spc command line invocation itself is also printed in the log file, preceded by a timestamp.

```
spc –-quiet . . .
spc –-log prov.log . . .
```

3.1.3. Provisioning Parameters

Provisioning is controlled by the following parameters (firmware upgrades are discussed in a later section).

- Provision_Enable
- Resync_On_Reset
- Resync_Random_Delay
- Resync_Periodic
- Resync_Error_Retry_Delay
- Resync_From_SIP
- Profile_Rule
- Log_Resync_Request_Msg
- Log_Resync_Success_Msg
- Log_Resync_Failure_Msg
- GPP_A
- GPP_B
- GPP_C
- GPP_D
- GPP_SA
- GPP_SB
- GPP_SC
- GPP_SD

Provision Enable:

```
ParName:    Provision_Enable

Default:    Enable
```

The CFG profile must be requested by the SPA, and cannot be pushed from a provisioning server (although a service provider can effectively push a profile by triggering the request operation remotely via SIP NOTIFY). The functionality is controlled by the Provision_Enable parameter. The parameter enables the functionality encompassed by the remaining provisioning parameters.

In addition, Provision_Enable also gates the ability to issue an explicit resync command from the web interface (discussed in a later section of this document).

Resync on Reset:

```
ParName:    Resync_On_Reset

Default:    Enable
```

Resync_On_Reset determines whether the SPA will attempt to resync with the provisioning server on power-up and following explicit reboot requests.

Resync Random Delay:

```
ParName:    Resync_Random_Delay
Default:    2
```

Resync_Random_Delay helps to scatter resync requests from multiple devices uniformly over a period of time, whose duration (in seconds) is indicated by this parameter. Hence, if a number of SPA devices were to power-up at the same time, their resync requests would be distributed over time, lessening the impact on the provisioning servers.

Resync Periodic:

```
ParName:    Resync_Periodic
Default:    3600
```

The SPA attempts to resync with the provisioning server periodically, provided the Resync_Periodic parameter is configured with a non-zero value. The value (in seconds) indicates the interval between resync attempts,

Resync Error Retry Delay:

```
ParName:    Resync_Error_Retry_Delay
Default:    3600
```

If a resync attempt fails, the SPA will retry with a delay indicated by the Resync_Error_Retry_Delay parameter, specified in seconds. If the value is zero, the SPA treats resync failures as though they were successful, and simply waits for the next periodic event to resync.

Resync From SIP:

```
ParName:    Resync_From_SIP
Default:    Enable
```

Resync_From_SIP gates the ability of a service provider to trigger a profile resync via a SIP NOTIFY message to the SPA.

Profile Rule:

```
ParName:    Profile_Rule
Default:    /spa$PSN.cfg
```

The Profile_Rule parameter is a script that identifies the provisioning server to contact when performing a profile resync. The string supports one level of macro expansion, using a small set of variables. Following macro substitution, the rule is evaluated to obtain the URL of the CFG file to be requested from the provisioning server.

The URL can be partially specified, in which case default values are assumed for the unspecified terms. The filepath portion of the URL must always be specified.

The Profile_Rule supports additional syntax that allows the URL to be a function of the firmware release currently running in the SPA. This mechanism can aid the service provider's firmware upgrade sequence, by allowing them to define different configuration profiles for different stages of an upgrade sequence.

The conditional syntax consists of a sequence of condition-url pairs, separated by the '|' character. The condition component tests the current firmware version number against a specified value. If the last url in the sequence does not have an associated condition, it will be attempted unconditionally.

The sequence of conditions is evaluated until one is satisfied. The URL associated with that condition is then used to resync the SPA. No additional URLs in the rule are considered.

Optional qualifiers can be specified in brackets, preceding each URL. As of release 1.0, the only supported qualifier is the key used to encrypt the CFG file, if key-based encryption is used.

To ease testing and development, the script syntax also supports using '#' as a comment delimiter (until end-of-parameter). This allows a potentially long script to be temporarily "commented out".

The syntax for the rule is as follows (with standard conventions for URLs):

```
rule = term [ '|' term [ '|' term . . . ] ]
term = '(' relop version ')' '?' [options] url
relop = '<' | '>' | '==' | '!=' | '!'
version = major ['.' minor ['.' build ['(' features ')'] ] ]
options = '[' --key key-string ']'
key-string = password | quoted-pass-phrase | hex-string
url = [method://][server[:port]]/filepath
method(*) = tftp | http | https
server(**) = empty | ipquad | FQDN
```

(*) Version 2.0 supports TFTP, HTTP and HTTPS.
(**) If unspecified, the TFTP server name provided by the LAN's DHCP server is used instead. Also, an FQDN with multiple DNS entries is multiply resolved by the SPA.

The variables available for macro substitution (with example values) are as follows:

```
PN          SPA-2000               Product Name
PSN         2000                   Product Series Number
MA          000e08aaa010           MAC Address
MAU         000E08AAA010           MAC Address (upper case)
MAC         00:0e:08:aa:a0:10      MAC Addr with Colon separators
SN          88012BAAA10102         Serial Number
SWVER       1.0.2                  Firmware Version Number
```

```
HWVER        1.0.1                   Hardware Version Number
UPGCOND      1.0.2<1.1               Upgrade(*) Condition
SCHEME         tftp                  Access Scheme
SERV         http.phoneme.com        Server Name
SERVIP       10.2.3.200              Server IP Address
PORT         69                      TCP/IP Request Port
PATH         /guest/spa2000.cfg      File path
ERR          corrupt file            Error/Info(**) message
A to D       some-value              Contents of GPP_A to GPP_D
SA to SD     some-value              Contents of GPP_SA to GPP_SD
```

(*) Note that the UPGCOND term is particularly useful in the Upgrade_Rule (discussed later in this document), but applies equally as a resync condition. It shows which term of the rule triggered the operation.
(**) Upon successful firmware upgrade, the ERR variable carries the version of the newly installed load.

In addition, the contents of the general purpose parameters, GPP_A, GPP_B, GPP_C, and GPP_D, are available as macro variables A, B, C, and D, respectively.

A secondary set of general purpose parameters is also available for macro substitution, GPP_SA, GPP_SB, GPP_SC, GPP_SD, using the respective expressions SA, SB, SC, and SD. These parameters are not accessible through the web interface, and can only be set via a configuration profile.

The macro variables are invoked by preceeding the name with a '$' character (e.g. $MAC). The substitution works even within a quoted string, without requiring additional escapes. If the name is immediately followed by an alphanumeric character, enclose the name in parentheses (e.g. $(MAC)).

To include a dollar sign in the rule, escape it with another dollar sign. That is $$ maps to $.

Profile_Rule syntax examples (each line is a separate example):

```
/spa2000.cfg
pserv.myvoice.com:42000/sip/$MA/spa2000.cfg
[--key 6e4f2a8733ba7c90aa13250bde4f6927]ur.well.com/Gj2fLx3Nqbg/a.cfg
(<1.0)?/pre-rel.cfg | /curr.cfg
```

Profile Example Scenarios:

Enterprise LAN with DHCP Supplied TFTP Server Name:
The DHCP server automatically advertises a TFTP server name to service the local network. Each SPA in the network is supplied a unique CFG file based on its MAC address. The TFTP server would also contain a generic spa2000.cfg in its tftp-root directory that contains the Profile_Rule indicated below. It would additionally carry individualized CFG files, one per device, within a tree below the tftp-root node. Each of these files would then individualize the devices.

```
/profiles/$MA/spa2000.cfg
```

When first powered-on, unprovisioned devices would download the /spa2000.cfg file from the TFTP server indicated by DHCP, (following their manufacturing default setting for the Profile_Rule parameter). The downloaded file would then direct the SPA to resync to the server and fetch the individualized CFG file, as per the rule above, which completes the provisioning sequence.

VoIP Service Provider:
Conceptually, a service provider solution would follow the steps as in the above example. In addition, it would then proceed to enable stronger encryption by implementing one more provisioning step, with one more level of redirection, involving a random CFG file path and encryption key. Hence, each of the “first-stage” CFG files above would point to a “second-stage” CFG file, with entries such as the following:

```
Profile_Rule “[--key $B] ps.global.com/profiles/active/$A/spa2000.cfg”;
GPP_A “Dz3P2q9sVgx7LmWbvu”;
GPP_B
“83c1e792bc6a824c0d18f429bea52d8483f2a24b32d75bc965d05e38c163d5ef”;
```

In practice, the first provisioning stage (which individualizes each SPA into fetching a unique CFG file) could be preconfigured during manufacturing.

For added security, the second stage, which introduces strong encryption, may be performed in-house, prior to shipping an SPA to each end-user.

Release 2.0 supports SSL-based key exchanges, alleviating the need for this in-house step, while preserving strong security for the provisioning process.

A provisioning flow chart, from the point of view of the SPA endpoint is presented in a later section.

Log Resync Request Message:

```
ParName:     Log_Resync_Request_Msg

Default:     $PN $MAC –- Requesting resync $SCHEME://$SERVIP:$PORT$PATH
```

The Log_Resync_Request_Msg is a script that defines the message sent to the configured Syslog server whenever the SPA attempts to resync with the provisioning server. The string supports one level of macro substitution, with the same variables as for the Profile_Rule above. An empty string does not generate a syslog message.

Log Resync Success Message:

```
ParName:     Log_Resync_Success_Msg

Default:     $PN $MAC –- Successful resync $SCHEME://$SERVIP:$PORT$PATH
```

The Log_Resync_Success_Msg is a script that defines the message sent to the configured Syslog server whenever the SPA successfully completes a resync with the provisioning server. The string supports one level of macro substitution, with the same variables as for the Profile_Rule above. An empty string does not generate a syslog message.

Log Resync Failure Message:

ParName:	Log_Resync_Failure_Msg
Default:	$PN $MAC –– Resync failed: $ERR

The Log_Resync_Failure_Msg is a script that defines the message sent to the configured Syslog server whenever the SPA fails to complete a resync with the provisioning server. The string supports one level of macro substitution, with the same variables as for the Profile_Rule above. An empty string does not generate a syslog message.

General Purpose Parameters:

ParName:	GPP_A
Default:	empty

GPP_A is one of 4 General Purpose Parameters, usable by both the provisioning and the upgrade logic. The parameter can be configured to hold any string value. Such a value can then be incorporated in other scripted parameters.

ParName:	GPP_B
Default:	empty

GPP_B is one of 4 General Purpose Parameters, usable by both the provisioning and the upgrade logic. The parameter can be configured to hold any string value. Such a value can then be incorporated in other scripted parameters.

ParName:	GPP_C
Default:	empty

GPP_C is one of 4 General Purpose Parameters, usable by both the provisioning and the upgrade logic. The parameter can be configured to hold any string value. Such a value can then be incorporated in other scripted parameters.

ParName:	GPP_D
Default:	empty

GPP_D is one of 4 General Purpose Parameters, usable by both the provisioning and the upgrade logic. The parameter can be configured to hold any string value. Such a value can then be incorporated in other scripted parameters.

General Purpose Secure Parameters:

```
ParName:     GPP_SA
Default:     empty
```

GPP_SA is one of 4 General Purpose Parameters, usable by both the provisioning and the upgrade logic. The parameter can be configured to hold any string value. Such a value can then be incorporated in other scripted parameters. This parameter is not accessible through the SPA web interface, and can only be set via a configuration profile. Also, the parameter cannot be incorporated as part of a syslog message.

```
ParName:     GPP_SB
Default:     Empty
```

GPP_SB is one of 4 General Purpose Parameters, usable by both the provisioning and the upgrade logic. The parameter can be configured to hold any string value. Such a value can then be incorporated in other scripted parameters. This parameter is not accessible through the SPA web interface, and can only be set via a configuration profile. Also, the parameter cannot be incorporated as part of a syslog message.

```
ParName:     GPP_SC
Default:     Empty
```

GPP_SC is one of 4 General Purpose Parameters, usable by both the provisioning and the upgrade logic. The parameter can be configured to hold any string value. Such a value can then be incorporated in other scripted parameters. This parameter is not accessible through the SPA web interface, and can only be set via a configuration profile. Also, the parameter cannot be incorporated as part of a syslog message.

```
ParName:     GPP_SD
Default:     Empty
```

GPP_SD is one of 4 General Purpose Parameters, usable by both the provisioning and the upgrade logic. The parameter can be configured to hold any string value. Such a value can then be incorporated in other scripted parameters. This parameter is not accessible through the SPA web interface, and can only be set via a configuration profile. Also, the parameter cannot be incorporated as part of a syslog message.

3.1.3.1. Firmware Upgrade

The SPA is firmware upgradeable via TFTP and HTTP. Firmware loads are released as single binary files, which contain all the modules pertaining to any one release version. By convention, the firmware loads are named with the extension ".bin" (e.g. spa.bin)

The SPA can be configured to upgrade to a specific version, possibly staging through intermediate releases, if necessary. This process can be automated for a pool of devices through configuration profile parameters.

Alternatively, an individual SPA can be directed to perform an upgrade to a specific firmware load via its built-in web server interface (this mechanism is discussed in section 3.4.4.1 of this document).

Firmware upgrades are attempted only when the SPA is idle, since they trigger a software reboot.

3.1.4. Upgrade Parameters

Firmware upgrades are controlled by the following parameters (which operate in a manner similar to but independent of the provisioning parameters).

- Upgrade_Enable
- Upgrade_Error_Retry_Delay
- Upgrade_Rule
- Log_Upgrade_Request_Msg
- Log_Upgrade_Success_Msg
- Log_Upgrade_Failure_Msg

Upgrade Enable:

```
ParName:     Upgrade_Enable

Default:     Enable
```

The firmware file must be requested by the SPA and cannot be pushed from an upgrade server (although a service provider can effectively push a new firmware load by triggering the request operation remotely via the CFG file). The functionality is controlled by the Upgrade_Enable parameter. The parameter enables the functionality encompassed by the remaining upgrade parameters.

In addition, Upgrade_Enable also gates the ability to issue an explicit upgrade command from the web interface (discussed in section 3.4.4.1 of this document).

Upgrade Error Retry Delay:

```
ParName:     Upgrade_Error_Retry_Delay

Default:     3600
```

If an upgrade attempt fails, the SPA will retry with a delay indicated by the Upgrade_Error_Retry_Delay parameter, specified in seconds. If the value is zero, the SPA treats upgrade failures as though they were successful, and will not retry to upgrade unless some event triggers a reboot.

Upgrade Rule:

```
ParName:     Upgrade_Rule

Default:     Empty
```

The Upgrade_Rule parameter is a script that identifies the upgrade server to contact during a firmware upgrade. The string supports one level of macro expansion, using a small set of variables. Following macro substitution, the rule is evaluated to obtain a URL of the firmware file to request from an upgrade server.

The URL can be partially specified, in which case default values are assumed for the unspecified terms. The filepath portion of the URL must be specified.

The Upgrade_Rule supports additional syntax that allows the URL to be a function of the firmware release currently running in the SPA. This mechanism can aid service providers sequence through a firmware upgrade, by allowing them to automatically stage the upgrade sequence, if so required by the firmware.

The conditional syntax consists of a sequence of condition-url pairs, separated by the '|' character. The condition component tests the current firmware version number against a specified value.

The sequence of conditions is evaluated until one is satisfied. The URL associated with that condition is then used to upgrade the SPA. No additional URLs in the rule are considered.

The upgrade will fail if the new firmware load does not satisfy the upgrade rule condition that suggested the URL. This alleviates the possibility of infinite upgrade loops, in case the device has been misconfigured.

The rule syntax is the same as for the Profile_Rule described in a previous section, except that there are no supported optional qualifiers for upgrades at this time. (That is, the bracketed options preceding the URL are not supported in the Upgrade_Rule). Also, all upgrade entries require a condition term to be specified.

Upgrade Rule Syntax Examples (each line is a separate example):

```
(! 1.0.2)? /spa2000/1-00-02/spa.bin
(<1.0)? tftp://pserv.myvoice.com:42001/upg/spa2000/1.0.2/spa.bin
(<0.99.52)?/spa09952.bin | (<1.0.2)?/spa10002.bin
```

In addition to TFTP, future firmware releases will support other upgrade methods.

Log Upgrade Request Message:

```
ParName:     Log_Upgrade_Request_Msg

Default:     $PN $MAC –- Requesting upgrade
             $SCHEME://$SERVIP:$PORT$PATH
```

The Log_Upgrade_Request_Msg is a script that defines the message sent to the configured Syslog server whenever the SPA attempts an upgrade from the upgrade server. The string supports one level of macro substitution, with the same variables as for the Upgrade_Rule above. An empty string does not generate a syslog message.

Log Upgrade Success Message:

```
ParName:    Log_Upgrade_Success_Msg

Default:    $PN $MAC –- Successful upgrade
            $SCHEME://$SERVIP:$PORT$PATH -- $ERR
```

The Log_Upgrade_Success_Msg is a script that defines the message sent to the configured Syslog server whenever the SPA successfully completes an upgrade from the upgrade server. The string supports one level of macro substitution, with the same variables as for the Upgrade_Rule above. An empty string does not generate a syslog message.

Log Upgrade Failure Message:

```
ParName:    Log_Upgrade_Failure_Msg

Default:    $PN $MAC –- Upgrade failed: $ERR
```

The Log_Upgrade_Failure_Msg is a script that defines the message sent to the configured Syslog server whenever the SPA fails to complete an upgrade from the upgrade server. The string supports one level of macro substitution, with the same variables as for the Upgrade_Rule above. An empty string does not generate a syslog message.

3.2. Configuration Update

Each SPA can be configured to periodically contact a Normal Provisioning Server (NPS). The NPS can be accessed with a less secure protocol since the updated profile is encrypted by a shared secret key. The NPS can be a standard TFTP, HTTP or HTTPS server.

3.2.1. Provisioning Server Redundancy

The Provisioning Server (PS) may be specified as an IP address or a FQDN. PS redundancy is not available in the former case. For the latter, SPA shall attempt to resolve the IP address of the PS via DNS SRV, then DNS A Record. In either case, the DNS server may return a number of IP addresses with priority (priority can be indicated in the case of SRV record; for A records, all IP addresses have the same priority). The SPA then contacts the IP address with the highest priority. If that fails, the SPA shall contact the next available IP address. The SPA shall continue the process until one of the PS responds. If all PS fail to respond, the SPA shall log an error to the Syslog server.

3.2.2. SPA Provisioning Flow

Firmware release 1.0 provides basic features in support of secure provisioning. This section describes the high-level provisioning flow supported by release 1.0 in the context of a service provider application.

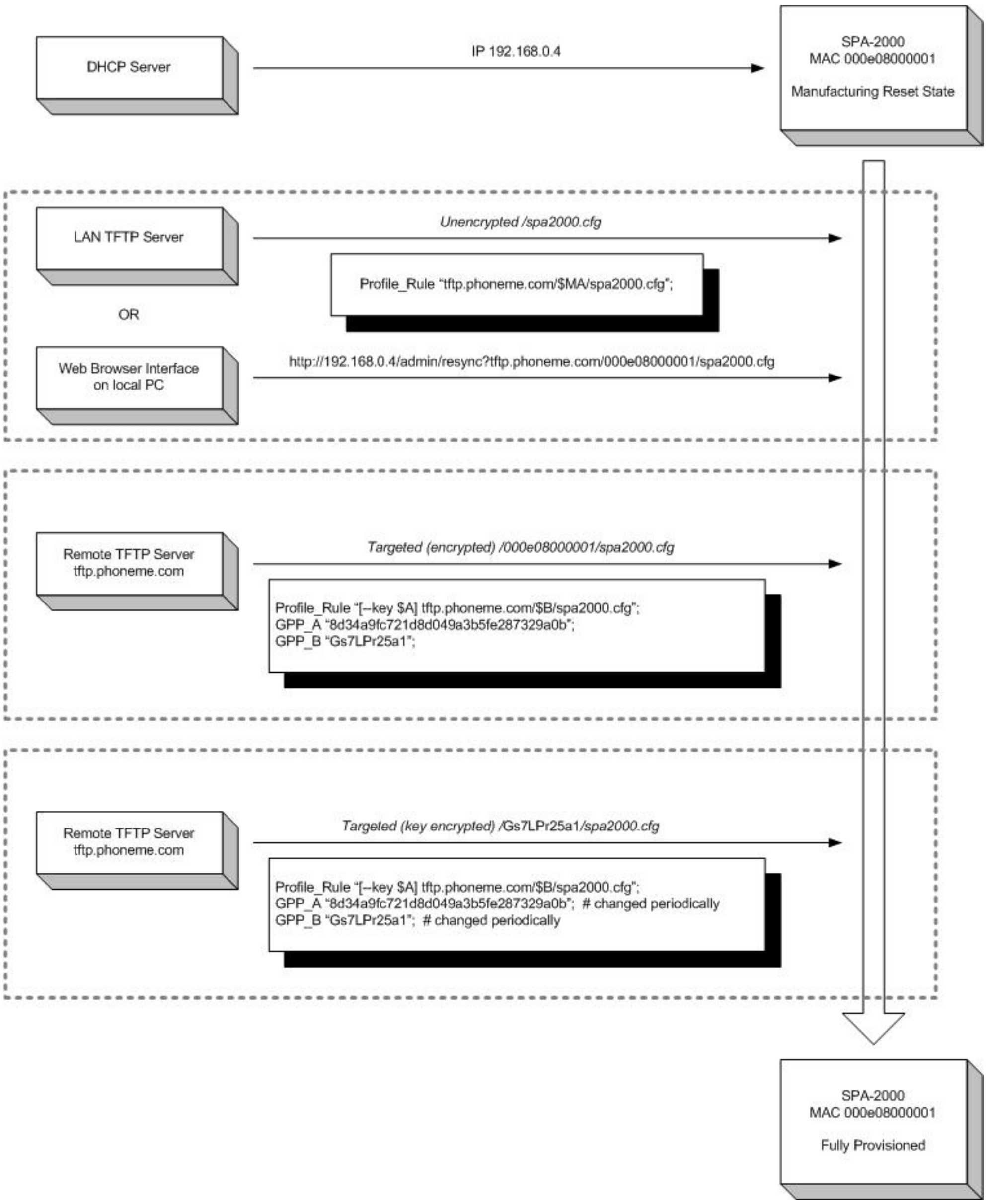
Future firmware releases will provide stronger secure provisioning mechanisms – in particular, the SPA will support asymmetric key encryption.

At a high level, the provisioning process involves four (4) provisioning states described in the below table.

Flow Step	**Step Description**
MFG-RESET	Manufacturing Reset Performing manufacturing reset on the SPA returns the device to a fully unprovisioned state. All configurable parameters regain their manufacturing default values. Manufacturing reset can be performed from any state through the IVR sequence ****RESET#1# Allowing the end user to perform manufacturing reset guarantees that the device can always be returned to an accessible state.
SP-CUST	Service Provider Customization The provisioning parameters are customized for a particular service provider network. The Profile_Rule parameter must be configured in this step to point to a device specific configuration profile, using a service provider specific provisioning server. The step can be accomplished in one of 3 ways: 1. Auto-configuration via local DHCP server. A TFTP server name or IPv4 address is specified by DHCP on he local network. The indicated TFTP server carries the desired Profile_Rule entry in the CFG file /spa2000.cfg 2. Enter a resync URL. An end-user opens a browser onto the SPA's web server, explicitly requesting a resync to a specific TFTP server, using this URL syntax: http://x.x.x.x/admin/resync?prvserv/spa2000.cfg where x.x.x.x is the IP address of the specific SPA and prvserv is the target TFTP server, followed by a profile path. 3. Edit Profile_Rule parameter. Open the provisioning pane on the SPA web interface, and enter the TFTP URL in the Profile_Rule parameter: e.g. prserv/spa2000.cfg The spa2000.cfg file modifies the Profile_Rule to contact a specific TFTP server, and request a MAC-address specific CFG file. For example, the following entry will contact a specific provisioning server, requesting a new profile unique to this unit: Profile_Rule "tftp.callme.com/profile/$MA/spa2000.cfg";

Flow Step	Step Description
SEC-PRV-1	Secure Provisioning – Initial Configuration The initial device-unique CFG file should be targeted to each SPA by compiling the CFG file with the spc "--target" option. This provides an initial level of encryption which does not require the exchange of keys. The initial device-unique CFG file should reconfigure the profile parameters to enable stronger encryption, by programming a 256-bit encryption key, and pointing to a randomly generated TFTP directory. For example, the CFG file might contain: Profile_Rule "[--key $A] tftp.callme.com/profile/$B/spa2000.cfg"; GPP_A "8e4ca259…"; # 256 bit key GPP_B "Gp3sqLn…"; # random CFG file path directory
SEC-PRV-2	Secure Provisioning – Full Configuration The subsequent profile resync operations retrieve 256-bit encrypted CFG files, which maintain the SPA in a state synchronized to the provisioning server. All remaining SPA parameters are configured and maintained through this strongly encrypted profile. The encryption key and random directory location can be changed periodically for extra security.

The SPA provisioning flow is diagramed in the following figure:



3.3. IVR Interface

Administrators and/or users can check (read) and set (write) basic network configuration settings via a touchtone telephone connected to one of the RJ-11 phone ports of the SPA.

Please Note:

Service Providers offering service using the SPA may restrict, protect or turn off certain aspects of the unit's IVR and web configuration capabilities.

The Interactive Voice Response (IVR) capabilities of the SPA are designed to give the administrator and/or user basic read/write capabilities such that the unit can attain basic IP network connectivity and the more advanced browser-based configuration menu may be accessed.

1. The SPA IVR uses the following conventions: By factory default there is no password and no password authentication is prompted for all the IVR settings. If administrator password is set, password authentication will be prompted for certain IVR settings. See 3.4.2 for detailed information about administrator password.

To input the password using the phone keypad, the following translation convention applies:

- To input: A, B, C, a, b, c -- press '**2**'
- To input: D, E, F, d, e, f -- press '**3**'
- To input: G, H, I, g, h, i -- press '**4**'
- To input: J, K, L, j, k, l -- press '**5**'
- To input: M, N, O, m, n, o -- press '**6**'
- To input: P, Q, R, S, p, q, r, s -- press '**7**'
- To input: T, U, V, t, u, v -- press '**8**'
- To input: W, X, Y, Z, w, x, y, z -- press '**9**'
- To input all other characters in the administrator password, press '**0**'

Note: This translation convention only applies to the password input.

For example: to input password "**test#@1234**" by phone keypad, you need to press the following sequence of digits: **8378001234**.

2. After entering a value, press the **#** (pound) key to indicate end of input.

- To Save value, press '**1**'
- To Review the value, press '**2**'
- To Re-enter the value, press '**3**'
- To Cancel the value entry and return to the main configuration menu, press '*****' (star)

Notes:

- The final '#' key won't be counted into value.
- Saved settings will take effect when the telephone is hung-up and if necessary, the SPA will automatically reboot.

3. After one minute of inactivity, the unit times out. The user will need to re-enter the configuration menu from the beginning by pressing *** * * ***.

4. If, while entering a value (like an IP address) and you decide to exit without entering any changes, you may do so by pressing the ***** (star) key **twice within a half second** window of time. Otherwise, the entry of the ***** (star) key will be treated as a dot (decimal point).

Example: To enter IP address, use numbers 0 – 9 on the telephone key pad and use the * (star) key to enter a decimal point.

To enter the following IP address value: 192.168.2.215

A. Use the touchtone key pad to enter: **192*168*2*215#**

B. When prompted, enter **1** to save setting to configuration.

C. Hang-up the phone to cause setting to take effect.

- or -

D. Enter the value of the next setting category to modify . . .

5. Hang-up the phone to cause all settings to take effect.

SPA Interactive Voice Response (IVR) Menu:

IVR Action	IVR Menu Choice	Parameter(s)	Notes:
Enter IVR Menu	*** * * ***	None	Ignore SIT or other tones until you hear, “Sipura configuration menu. Please enter option followed by the pound key or hang-up to exit.”
Exit IVR Menu	**3948**	None	
Check DHCP	**100**	None	IVR will announce if DHCP in enabled or disabled.
Enable/Disable DHCP	**101**	Enter 1 to enable Enter 0 to disable	**Requires Password**
Check IP Address	**110**	None	IVR will announce the current IP address of SPA.
Set Static IP Address	**111**	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	DHCP must be “Disabled” otherwise you will hear, “Invalid Option,” if you try to set this value. **Requires Password**
Check Network Mask	**120**	None	IVR will announce the current network mask of SPA.
Set Network Mask	**121**	Enter value using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	DHCP must be “Disabled” otherwise you will hear, “Invalid Option,” if you try to set this value. **Requires Password**
Check Static Gateway IP Address	**130**	None	IVR will announce the current gateway IP address of SPA.
Set Static Gateway IP Address	**131**	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	DHCP must be “Disabled” otherwise you will hear, “Invalid Option,” if you try to set this value. **Requires Password**

Check MAC Address	**140**	None	IVR will announce the MAC address of SPA in hex string format.
Check Firmware Version	**150**	None	IVR will announce the version of the firmware running on the SPA.
Check Primary DNS Server Setting	**160**	None	**IVR will announce the current setting in the Primary DNS field.**
Set Primary DNS Server	**161**	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	**Requires Password**
Check SPA's Web Server Port	**170**	None	**IVR will announce the port that the web server is listening on. (Default is 80)**
Enable/Disable Web Server of SPA	**7932**	Enter 1 to enable Enter 0 to disable	**Requires Password**
Manual Reboot of Unit	**732668**	None	After you hear "Option Successful," hang-up. Unit will reboot automatically.
User Factory Reset of Unit **WARNING:** **ALL "User-Changeable" NON-DEFAULT SETTINGS WILL BE LOST!** **This might include network and service provider data.**	**877778**	Enter 1 to confirm Enter *(star) to cancel operation	SPA will prompt for confirmation. After confirming, you will hear "Option Successful." Hang-up. Unit will reboot and all "User Changeable" configuration parameters will be reset to factory default values.
Factory Reset of Unit **WARNING:** **ALL NON-DEFAULT SETTINGS WILL BE LOST!** **This includes network and service provider data.**	**73738**	Enter 1 to confirm Enter * (star) to cancel operation	SPA will prompt for confirmation. After confirming, you will hear "Option Successful." Hang-up. Unit will reboot and all configuration parameters will be reset to factory default values.

Note: If the Administrator password is not set or the user is allowed to change it, the items marked with "Requires Password" will not require a password.

3.4. Web Interface

The SPA provides a built-in web server. Configuration and administration can be performed through this convenient web interface.

3.4.1. Web Interface Conventions

The SPA uses the following conventions with the web administration capabilities:

- The SPA web administration supports two privilege levels: Administrator and User. To use the User privilege, simply point a web browser at the IP address of the SPA; to use the administrator privilege, use URL http://IP_Address_Of_SPA/admin/. See 3.4.2 for more information about administration privileges.
- Version 1.0 of the SPA supports Internet Explorer 5.5 and above and Netscape 7.0 and above.
- The web configuration pages can be password protected. See 3.4.2 for more information about password protect.
- The user name of web Administrator is : **admin**
- The user name of web User is : **user**
- **Note**: The user names for both administrator and User are fixed and cannot be changed.
- After making changes to SPA configuration parameters, pressing "**Submit All Changes**" button will apply all the changes and if necessary, automatically reboot the device. Multiple changes may be made on multiple page tabs of the web interface at the same time. Pressing "**Submit All Changes**" will apply all the modifications.

  Important Note: switching between page tabs won't apply the changes to SPA, The only way to apply the changes is to press the "**Submit All Changes**" button.
- If the "**Undo All Changes**" button is clicked, any modifications to profile parameters *on any and all pages* will be reset back to their original values before modification.

  NOTE: Pressing the "**Undo All Changes**" has no effect on the SPA; it will only reset the values on the web page.

3.4.2. Administration Privileges

The SPA supports two levels of administration privileges: Administrator and User, both privileges can be password protected. **Important note**: by factory default, there are no passwords assigned for both Administrator and User.

The Administrator has the privilege to modify all the web profile parameters and can also modify the passwords of both Administrator and User. A User only has the privilege to access part of the web profile parameters; the parameter group that User can access is specified by the Administrator, which can only be done through provisioning.

To access the Administrator level privilege, use URL: http://IP_Address_Of_SPA/admin/. If the password has been set for Administrator, the browser will prompt for authentication. The username for Administrator is "admin" and cannot be changed.

To access the User level privilege, use URL: http://IP_Address_Of_SPA/. If the password has been set for User, the browser will prompt for User authentication. The username for User is "user" and cannot be changed.

When browsing Administrator pages, one can switch to User privileges by click the link "User Login". (**Note**: if User password was set, the browser will prompt for User authentication when you click "User Login" link). On the other side, from the User pages you can switch to Administrator privilege by clicking the link "Admin Login." Authentication is needed if Administrator password has been set.

Warning: Switching between the User and Administrator will discard the uncommitted changes that have already been made on the web pages.

3.4.3. Basic and Advanced Views

The web configuration interface provides a Basic and an Advanced view from which the various configuration parameters can be accessed. The SPA Provisioning tab is only visible from the Advanced Administrator view of the web interface.

Warning: Switching between the basic and advanced view will discard the uncommitted changes that have already been made on the web pages.

3.4.4. Functional URLs

The web interface of the SPA supports several functions through special URLs: Upgrade, Reboot and Resync. Administrator privilege is needed for these functions.

3.4.4.1. Upgrade URL

Through upgrade URL you can upgrade the SPA to a firmware specified by the URL. Note: If the value of "upgrade enable" parameter in Provisioning tab is no, you cannot upgrade the SPA even if the web page tells you that the upgrade will be done when it is not in use. See 3.1.3.1 to get more information on firmware upgrade.

The syntax of Upgrade URL is:

http://<spa-ip-addr>/upgrade?[protocol://][server-name[:port]][/firmware-pathname] or

http://<spa-ip-addr>/admin/upgrade?[protocol://][server-name[:port]][/firmware-pathname]

If no protocol is specified, TFTP is assumed. Note: Only TFTP is supported in the current release.

If no server-name is specified, the host that requests the URL is used as server-name.

If no port specified, default port of the protocol is used. (69 for TFTP)

The "firmware-pathname" is typically the file name of the SPA binary located in the root directory of the TFTP server. If no firmware-pathname is specified, "/spa.bin" is assumed.

For example: **http://192.168.2.217/upgrade?tftp://192.168.2.251/spa.bin**

3.4.4.2. Resync URL

Through Resync URL you can force the SPA to do a resync to a profile specified in the URL.

Note: The SPA will resync only when it is idle.

The syntax of Resync URL is:

http://<spa-ip-addr>/resync?[[protocol://][server-name[:port]]/profile-pathname]

If no parameter follows "/resync?", the profile rule setting in provisioning is used. See 3.1.3 for detailed information about profile rule in provisioning

If no protocol is specified, TFTP protocol is assumed. Note: Only TFTP is supported in the current release.

If no server-name is specified, the host that requests the URL is used as server-name.

If no port specified, default port of the protocol is used – 69 for TFTP.

The profile-path is the path to the new profile to resync with.

For example: http://192.168.2.217/upgrade?tftp://192.168.2.251/spaconf.scf

3.4.4.3. Reboot URL

Through the Reboot URL, you can reboot the SPA.

Note: Upon request, the SPA will reboot only when it is idle.

The Reboot URL is: http://<spa-ip-addr>/admin/reboot

3.5. Configuration Parameters

3.5.1. Configuration Profile Compiler

The SPA accepts configuration profiles in binary format. A translation tool (spc.exe) translates a human editable format into the binary format understood by the SPA.

The current spc.exe tool expects a semicolon, ";", to separate each parameter definition. If a parameter is not defined in the configuration profile, the current value for that parameter is retained by the SPA.

Below, is an example of a typical SPA-2000 configuration text file. The SPA-1000 is similar. Please note that the SPA-3000 has a number of parameters unique to its platform.

The below profile print out can be obtained by using the SPC for

```
# ***
# *** Sipura SPA Series Configuration Parameters
# ***


# *** System Configuration

Restricted_Access_Domains          "" ;
Enable_Web_Server                  "Yes" ;
Web_Server_Port                    "80" ;
Enable_Web_Admin_Access            "Yes" ;
Admin_Passwd                       "" ;
User_Password                    ! "" ;

# *** Internet Connection Type

DHCP                             ! "Yes" ;
Static_IP                        ! "" ;
NetMask                          ! "" ;
Gateway                          ! "" ;

# *** Optional Network Configuration

HostName                         ! "" ;
Domain                           ! "" ;
Primary_DNS                      ! "" ;
Secondary_DNS                    ! "" ;
DNS_Server_Order                   "Manual" ;  # options: Manual/Manual,DHCP/DHCP,Manual
DNS_Query_Mode                     "Parallel" ;  # options: Parallel/Sequential
Syslog_Server                      "" ;
Debug_Server                       "" ;
Debug_Level                        "0" ;  # options: 0/1/2/3
Primary_NTP_Server                 "" ;
Secondary_NTP_Server               "" ;

# *** Configuration Profile

Provision_Enable                   "Yes" ;
Resync_On_Reset                    "Yes" ;
Resync_Random_Delay                "2" ;
```

```
Resync_Periodic                  "3600" ;
Resync_Error_Retry_Delay         "3600" ;
Forced_Resync_Delay              "14400" ;
Resync_From_SIP                  "Yes" ;
Resync_After_Upgrade_Attempt     "Yes" ;
Resync_Trigger_1                 "" ;
Resync_Trigger_2                 "" ;
Profile_Rule                     "/spa$PSN.cfg" ;
Profile_Rule_B                   "" ;
Profile_Rule_C                   "" ;
Profile_Rule_D                   "" ;
Log_Resync_Request_Msg           "$PN $MAC -- Requesting resync
$SCHEME://$SERVIP:$PORT$PATH" ;
Log_Resync_Success_Msg           "$PN $MAC -- Successful resync
$SCHEME://$SERVIP:$PORT$PATH" ;
Log_Resync_Failure_Msg           "$PN $MAC -- Resync failed: $ERR" ;

# *** Firmware Upgrade

Upgrade_Enable                   "Yes" ;
Upgrade_Error_Retry_Delay        "3600" ;
Downgrade_Rev_Limit              "" ;
Upgrade_Rule                     "" ;
Log_Upgrade_Request_Msg          "$PN $MAC -- Requesting upgrade
$SCHEME://$SERVIP:$PORT$PATH" ;
Log_Upgrade_Success_Msg          "$PN $MAC -- Successful upgrade
$SCHEME://$SERVIP:$PORT$PATH -- $ERR" ;
Log_Upgrade_Failure_Msg          "$PN $MAC -- Upgrade failed: $ERR" ;

# *** General Purpose Parameters

GPP_A                            "" ;
GPP_B                            "" ;
GPP_C                            "" ;
GPP_D                            "" ;
GPP_E                            "" ;
GPP_F                            "" ;
GPP_G                            "" ;
GPP_H                            "" ;
GPP_I                            "" ;
GPP_J                            "" ;
GPP_K                            "" ;
GPP_L                            "" ;
GPP_M                            "" ;
GPP_N                            "" ;
GPP_O                            "" ;
GPP_P                            "" ;
GPP_SA                           "" ;
GPP_SB                           "" ;
GPP_SC                           "" ;
GPP_SD                           "" ;

# *** SIP Parameters

Max_Forward                      "70" ;
Max_Redirection                  "5" ;
Max_Auth                         "2" ;
SIP_User_Agent_Name              "$VERSION" ;
SIP_Server_Name                  "$VERSION" ;
SIP_Accept_Language              "" ;
DTMF_Relay_MIME_Type             "application/dtmf-relay" ;
Hook_Flash_MIME_Type             "application/hook-flash" ;
Remove_Last_Reg                  "No" ;
Use_Compact_Header               "No" ;

# *** SIP Timer Values (sec)

SIP_T1                           ".5" ;
SIP_T2                           "4" ;
```

```
SIP_T4                            "5" ;
SIP_Timer_B                       "32" ;
SIP_Timer_F                       "32" ;
SIP_Timer_H                       "32" ;
SIP_Timer_D                       "32" ;
SIP_Timer_J                       "32" ;
INVITE_Expires                    "240" ;
ReINVITE_Expires                  "30" ;
Reg_Min_Expires                   "1" ;
Reg_Max_Expires                   "7200" ;
Reg_Retry_Intvl                   "30" ;
Reg_Retry_Long_Intvl              "1200" ;

# *** Response Status Code Handling

SIT1_RSC                          "" ;
SIT2_RSC                          "" ;
SIT3_RSC                          "" ;
SIT4_RSC                          "" ;
Try_Backup_RSC                    "" ;
Retry_Reg_RSC                     "" ;

# *** RTP Parameters

RTP_Port_Min                      "16384" ;
RTP_Port_Max                      "16482" ;
RTP_Packet_Size                   "0.030" ;
Max_RTP_ICMP_Err                  "0" ;
RTCP_Tx_Interval                  "0" ;

# *** SDP Payload Types

NSE_Dynamic_Payload               "100" ;
AVT_Dynamic_Payload               "101" ;
G726r16_Dynamic_Payload           "98" ;
G726r24_Dynamic_Payload           "97" ;
G726r40_Dynamic_Payload           "96" ;
G729b_Dynamic_Payload             "99" ;
NSE_Codec_Name                    "NSE" ;
AVT_Codec_Name                    "telephone-event" ;
G711u_Codec_Name                  "PCMU" ;
G711a_Codec_Name                  "PCMA" ;
G726r16_Codec_Name                "G726-16" ;
G726r24_Codec_Name                "G726-24" ;
G726r32_Codec_Name                "G726-32" ;
G726r40_Codec_Name                "G726-40" ;
G729a_Codec_Name                  "G729a" ;
G729b_Codec_Name                  "G729ab" ;
G723_Codec_Name                   "G723" ;

# *** NAT Support Parameters

Handle_VIA_received               "No" ;
Handle_VIA_rport                  "No" ;
Insert_VIA_received               "No" ;
Insert_VIA_rport                  "No" ;
Substitute_VIA_Addr               "No" ;
Send_Resp_To_Src_Port             "No" ;
STUN_Enable                       "No" ;
STUN_Test_Enable                  "No" ;
STUN_Server                       "" ;
EXT_IP                            "" ;
EXT_RTP_Port_Min                  "" ;
NAT_Keep_Alive_Intvl              "15" ;

# ***

Line_Enable[1]                    "Yes" ;
SAS_Enable[1]                     "No" ;
```

```
MOH_Server[1]                     "" ;
SAS_DLG_Refresh_Intvl[1]          "30" ;
NAT_Mapping_Enable[1]             "No" ;
SAS_Inbound_RTP_Sink[1]           "" ;
SIP_Port[1]                       "5060" ;
NAT_Keep_Alive_Enable[1]          "No" ;
EXT_SIP_Port[1]                   "" ;
NAT_Keep_Alive_Msg[1]             "$NOTIFY" ;
SIP_TOS/DiffServ_Value[1]         "0x68" ;
NAT_Keep_Alive_Dest[1]            "$PROXY" ;
RTP_TOS/DiffServ_Value[1]         "0xb8" ;
SIP_Debug_Option[1]               "none" ;  # options: none/1-line/1-line excl. OPT/1-line
excl. NTFY/1-line excl. REG/1-line excl. OPT|NTFY|REG/full/full excl. OPT/full excl.
NTFY/full excl. REG/full excl. OPT|NTFY|REG
Network_Jitter_Level[1]           "high" ;  # options: low/medium/high/very high
SIP_100REL_Enable[1]              "No" ;
Blind_Attn-Xfer_Enable[1]         "No" ;
Auth_Resync-Reboot[1]             "Yes" ;
SIP_Remote-Party-ID[1]            "No" ;

# *** Proxy and Registration

Proxy[1]                          "" ;
Use_Outbound_Proxy[1]             "No" ;
Outbound_Proxy[1]                 "" ;
Use_OB_Proxy_In_Dialog[1]         "Yes" ;
Register[1]                       "Yes" ;
Make_Call_Without_Reg[1]          "No" ;
Register_Expires[1]               "3600" ;
Ans_Call_Without_Reg[1]           "No" ;
Use_DNS_SRV[1]                    "No" ;
DNS_SRV_Auto_Prefix[1]            "No" ;
Proxy_Fallback_Intvl[1]           "3600" ;

# *** Subscriber Information

Display_Name[1]                   "" ;
User_ID[1]                        "" ;
Password[1]                       "" ;
Use_Auth_ID[1]                    "No" ;
Auth_ID[1]                        "" ;
Mini_Certificate[1]               "" ;
SRTP_Private_Key[1]               "" ;

# *** Supplementary Service Subscription

Call_Waiting_Serv[1]              "Yes" ;
Block_CID_Serv[1]                 "Yes" ;
Block_ANC_Serv[1]                 "Yes" ;
Dist_Ring_Serv[1]                 "Yes" ;
Cfwd_All_Serv[1]                  "Yes" ;
Cfwd_Busy_Serv[1]                 "Yes" ;
Cfwd_No_Ans_Serv[1]               "Yes" ;
Cfwd_Sel_Serv[1]                  "Yes" ;
Cfwd_Last_Serv[1]                 "Yes" ;
Block_Last_Serv[1]                "Yes" ;
Accept_Last_Serv[1]               "Yes" ;
DND_Serv[1]                       "Yes" ;
CID_Serv[1]                       "Yes" ;
CWCID_Serv[1]                     "Yes" ;
Call_Return_Serv[1]               "Yes" ;
Call_Back_Serv[1]                 "Yes" ;
Three_Way_Call_Serv[1]            "Yes" ;
Three_Way_Conf_Serv[1]            "Yes" ;
Attn_Transfer_Serv[1]             "Yes" ;
Unattn_Transfer_Serv[1]           "Yes" ;
MWI_Serv[1]                       "Yes" ;
VMWI_Serv[1]                      "Yes" ;
Speed_Dial_Serv[1]                "Yes" ;
```

```
Secure_Call_Serv[1]                "Yes" ;
Referral_Serv[1]                   "Yes" ;
Feature_Dial_Serv[1]               "Yes" ;

# *** Audio Configuration

Preferred_Codec[1]                 "G711u" ;  # options: G711u/G711a/G726-16/G726-24/G726-
32/G726-40/G729a/G723
Silence_Supp_Enable[1]             "No" ;
Use_Pref_Codec_Only[1]             "No" ;
Echo_Canc_Enable[1]                "Yes" ;
G729a_Enable[1]                    "Yes" ;
Echo_Canc_Adapt_Enable[1]          "Yes" ;
G723_Enable[1]                     "Yes" ;
Echo_Supp_Enable[1]                "Yes" ;
G726-16_Enable[1]                  "Yes" ;
FAX_CED_Detect_Enable[1]           "Yes" ;
G726-24_Enable[1]                  "Yes" ;
FAX_CNG_Detect_Enable[1]           "Yes" ;
G726-32_Enable[1]                  "Yes" ;
FAX_Passthru_Codec[1]              "G711u" ;  # options: G711u/G711a
G726-40_Enable[1]                  "Yes" ;
FAX_Codec_Symmetric[1]             "Yes" ;
DTMF_Tx_Method[1]                  "Auto" ;  # options: InBand/AVT/INFO/Auto
FAX_Passthru_Method[1]             "NSE" ;  # options: None/NSE/ReINVITE
Hook_Flash_Tx_Method[1]            "None" ;  # options: None/AVT/INFO
FAX_Process_NSE[1]                 "Yes" ;
Release_Unused_Codec[1]            "Yes" ;

# *** Dial Plan

Dial_Plan[1]                       "(*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-
9]xxxxxxS0|xxxxxxxxxxxx.)" ;
Enable_IP_Dialing[1]               "No" ;

# *** FXS Port Polarity Configuration

Idle_Polarity[1]                   "Forward" ;  # options: Forward/Reverse
Caller_Conn_Polarity[1]            "Forward" ;  # options: Forward/Reverse
Callee_Conn_Polarity[1]            "Forward" ;  # options: Forward/Reverse

# *** Call Forward Settings

Cfwd_All_Dest[1]                 ! "" ;
Cfwd_Busy_Dest[1]                ! "" ;
Cfwd_No_Ans_Dest[1]              ! "" ;
Cfwd_No_Ans_Delay[1]             ! "20" ;

# *** Selective Call Forward Settings

Cfwd_Sel1_Caller[1]              ! "" ;
Cfwd_Sel1_Dest[1]                ! "" ;
Cfwd_Sel2_Caller[1]              ! "" ;
Cfwd_Sel2_Dest[1]                ! "" ;
Cfwd_Sel3_Caller[1]              ! "" ;
Cfwd_Sel3_Dest[1]                ! "" ;
Cfwd_Sel4_Caller[1]              ! "" ;
Cfwd_Sel4_Dest[1]                ! "" ;
Cfwd_Sel5_Caller[1]              ! "" ;
Cfwd_Sel5_Dest[1]                ! "" ;
Cfwd_Sel6_Caller[1]              ! "" ;
Cfwd_Sel6_Dest[1]                ! "" ;
Cfwd_Sel7_Caller[1]              ! "" ;
Cfwd_Sel7_Dest[1]                ! "" ;
Cfwd_Sel8_Caller[1]              ! "" ;
Cfwd_Sel8_Dest[1]                ! "" ;
Cfwd_Last_Caller[1]              ! "" ;
Cfwd_Last_Dest[1]                ! "" ;
Block_Last_Caller[1]             ! "" ;
```

```
Accept_Last_Caller[1]           ! "" ;

# *** Speed Dial Settings

Speed_Dial_2[1]                 ! "" ;
Speed_Dial_3[1]                 ! "" ;
Speed_Dial_4[1]                 ! "" ;
Speed_Dial_5[1]                 ! "" ;
Speed_Dial_6[1]                 ! "" ;
Speed_Dial_7[1]                 ! "" ;
Speed_Dial_8[1]                 ! "" ;
Speed_Dial_9[1]                 ! "" ;

# *** Supplementary Service Settings

CW_Setting[1]                   ! "Yes" ;
Block_CID_Setting[1]            ! "No" ;
Block_ANC_Setting[1]            ! "No" ;
DND_Setting[1]                  ! "No" ;
CID_Setting[1]                  ! "Yes" ;
CWCID_Setting[1]                ! "Yes" ;
Dist_Ring_Setting[1]            ! "Yes" ;
Secure_Call_Setting[1]            "No" ;

# *** Distinctive Ring Settings

Ring1_Caller[1]                 ! "" ;
Ring2_Caller[1]                 ! "" ;
Ring3_Caller[1]                 ! "" ;
Ring4_Caller[1]                 ! "" ;
Ring5_Caller[1]                 ! "" ;
Ring6_Caller[1]                 ! "" ;
Ring7_Caller[1]                 ! "" ;
Ring8_Caller[1]                 ! "" ;

# *** Ring Settings

Default_Ring[1]                 ! "1" ;  # options: 1/2/3/4/5/6/7/8
Default_CWT[1]                  ! "1" ;  # options: 1/2/3/4/5/6/7/8
Hold_Reminder_Ring[1]           ! "8" ;  # options: 1/2/3/4/5/6/7/8/none
Call_Back_Ring[1]               ! "7" ;  # options: 1/2/3/4/5/6/7/8
Cfwd_Ring_Splash_Len[1]         ! "0" ;
Cblk_Ring_Splash_Len[1]         ! "0" ;
VMWI_Ring_Splash_Len[1]         ! ".5" ;
VMWI_Ring_Policy[1]               "New VM Available" ;  # options: New VM Available/New VM
Becomes Available/New VM Arrives
Ring_On_No_New_VM[1]              "No" ;

# ***

Line_Enable[2]                    "Yes" ;
SAS_Enable[2]                     "No" ;
MOH_Server[2]                     "" ;
SAS_DLG_Refresh_Intvl[2]          "30" ;
NAT_Mapping_Enable[2]             "No" ;
SAS_Inbound_RTP_Sink[2]           "" ;
SIP_Port[2]                       "5061" ;
NAT_Keep_Alive_Enable[2]          "No" ;
EXT_SIP_Port[2]                   "" ;
NAT_Keep_Alive_Msg[2]             "$NOTIFY" ;
SIP_TOS/DiffServ_Value[2]         "0x68" ;
NAT_Keep_Alive_Dest[2]            "$PROXY" ;
RTP_TOS/DiffServ_Value[2]         "0xb8" ;
SIP_Debug_Option[2]               "none" ;  # options: none/1-line/1-line excl. OPT/1-line
excl. NTFY/1-line excl. REG/1-line excl. OPT|NTFY|REG/full/full excl. OPT/full excl.
NTFY/full excl. REG/full excl. OPT|NTFY|REG
Network_Jitter_Level[2]           "high" ;  # options: low/medium/high/very high
SIP_100REL_Enable[2]              "No" ;
Blind_Attn-Xfer_Enable[2]         "No" ;
```

```
Auth_Resync-Reboot[2]             "Yes" ;
SIP_Remote-Party-ID[2]            "No" ;

# *** Proxy and Registration

Proxy[2]                          "" ;
Use_Outbound_Proxy[2]             "No" ;
Outbound_Proxy[2]                 "" ;
Use_OB_Proxy_In_Dialog[2]         "Yes" ;
Register[2]                       "Yes" ;
Make_Call_Without_Reg[2]          "No" ;
Register_Expires[2]               "3600" ;
Ans_Call_Without_Reg[2]           "No" ;
Use_DNS_SRV[2]                    "No" ;
DNS_SRV_Auto_Prefix[2]            "No" ;
Proxy_Fallback_Intvl[2]           "3600" ;

# *** Subscriber Information

Display_Name[2]                   "" ;
User_ID[2]                        "" ;
Password[2]                       "" ;
Use_Auth_ID[2]                    "No" ;
Auth_ID[2]                        "" ;
Mini_Certificate[2]               "" ;
SRTP_Private_Key[2]               "" ;

# *** Supplementary Service Subscription

Call_Waiting_Serv[2]              "Yes" ;
Block_CID_Serv[2]                 "Yes" ;
Block_ANC_Serv[2]                 "Yes" ;
Dist_Ring_Serv[2]                 "Yes" ;
Cfwd_All_Serv[2]                  "Yes" ;
Cfwd_Busy_Serv[2]                 "Yes" ;
Cfwd_No_Ans_Serv[2]               "Yes" ;
Cfwd_Sel_Serv[2]                  "Yes" ;
Cfwd_Last_Serv[2]                 "Yes" ;
Block_Last_Serv[2]                "Yes" ;
Accept_Last_Serv[2]               "Yes" ;
DND_Serv[2]                       "Yes" ;
CID_Serv[2]                       "Yes" ;
CWCID_Serv[2]                     "Yes" ;
Call_Return_Serv[2]               "Yes" ;
Call_Back_Serv[2]                 "Yes" ;
Three_Way_Call_Serv[2]            "Yes" ;
Three_Way_Conf_Serv[2]            "Yes" ;
Attn_Transfer_Serv[2]             "Yes" ;
Unattn_Transfer_Serv[2]           "Yes" ;
MWI_Serv[2]                       "Yes" ;
VMWI_Serv[2]                      "Yes" ;
Speed_Dial_Serv[2]                "Yes" ;
Secure_Call_Serv[2]               "Yes" ;
Referral_Serv[2]                  "Yes" ;
Feature_Dial_Serv[2]              "Yes" ;

# *** Audio Configuration

Preferred_Codec[2]                "G711u" ;  # options: G711u/G711a/G726-16/G726-24/G726-
32/G726-40/G729a/G723
Silence_Supp_Enable[2]            "No" ;
Use_Pref_Codec_Only[2]            "No" ;
Echo_Canc_Enable[2]               "Yes" ;
G729a_Enable[2]                   "Yes" ;
Echo_Canc_Adapt_Enable[2]         "Yes" ;
G723_Enable[2]                    "Yes" ;
Echo_Supp_Enable[2]               "Yes" ;
G726-16_Enable[2]                 "Yes" ;
FAX_CED_Detect_Enable[2]          "Yes" ;
```

```
G726-24_Enable[2]                  "Yes" ;
FAX_CNG_Detect_Enable[2]           "Yes" ;
G726-32_Enable[2]                  "Yes" ;
FAX_Passthru_Codec[2]              "G711u" ;  # options: G711u/G711a
G726-40_Enable[2]                  "Yes" ;
FAX_Codec_Symmetric[2]             "Yes" ;
DTMF_Tx_Method[2]                  "Auto" ;  # options: InBand/AVT/INFO/Auto
FAX_Passthru_Method[2]             "NSE" ;  # options: None/NSE/ReINVITE
Hook_Flash_Tx_Method[2]            "None" ;  # options: None/AVT/INFO
FAX_Process_NSE[2]                 "Yes" ;
Release_Unused_Codec[2]            "Yes" ;

# *** Dial Plan

Dial_Plan[2]                       "(*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-
9]xxxxxxS0|xxxxxxxxxxxx.)" ;
Enable_IP_Dialing[2]               "No" ;

# *** FXS Port Polarity Configuration

Idle_Polarity[2]                   "Forward" ;  # options: Forward/Reverse
Caller_Conn_Polarity[2]            "Forward" ;  # options: Forward/Reverse
Callee_Conn_Polarity[2]            "Forward" ;  # options: Forward/Reverse

# *** Call Forward Settings

Cfwd_All_Dest[2]                 ! "" ;
Cfwd_Busy_Dest[2]                ! "" ;
Cfwd_No_Ans_Dest[2]              ! "" ;
Cfwd_No_Ans_Delay[2]             ! "20" ;

# *** Selective Call Forward Settings

Cfwd_Sel1_Caller[2]              ! "" ;
Cfwd_Sel1_Dest[2]                ! "" ;
Cfwd_Sel2_Caller[2]              ! "" ;
Cfwd_Sel2_Dest[2]                ! "" ;
Cfwd_Sel3_Caller[2]              ! "" ;
Cfwd_Sel3_Dest[2]                ! "" ;
Cfwd_Sel4_Caller[2]              ! "" ;
Cfwd_Sel4_Dest[2]                ! "" ;
Cfwd_Sel5_Caller[2]              ! "" ;
Cfwd_Sel5_Dest[2]                ! "" ;
Cfwd_Sel6_Caller[2]              ! "" ;
Cfwd_Sel6_Dest[2]                ! "" ;
Cfwd_Sel7_Caller[2]              ! "" ;
Cfwd_Sel7_Dest[2]                ! "" ;
Cfwd_Sel8_Caller[2]              ! "" ;
Cfwd_Sel8_Dest[2]                ! "" ;
Cfwd_Last_Caller[2]              ! "" ;
Cfwd_Last_Dest[2]                ! "" ;
Block_Last_Caller[2]             ! "" ;
Accept_Last_Caller[2]            ! "" ;

# *** Speed Dial Settings

Speed_Dial_2[2]                  ! "" ;
Speed_Dial_3[2]                  ! "" ;
Speed_Dial_4[2]                  ! "" ;
Speed_Dial_5[2]                  ! "" ;
Speed_Dial_6[2]                  ! "" ;
Speed_Dial_7[2]                  ! "" ;
Speed_Dial_8[2]                  ! "" ;
Speed_Dial_9[2]                  ! "" ;

# *** Supplementary Service Settings

CW_Setting[2]                    ! "Yes" ;
Block_CID_Setting[2]             ! "No" ;
```

```
Block_ANC_Setting[2]             ! "No" ;
DND_Setting[2]                   ! "No" ;
CID_Setting[2]                   ! "Yes" ;
CWCID_Setting[2]                 ! "Yes" ;
Dist_Ring_Setting[2]             ! "Yes" ;
Secure_Call_Setting[2]             "No" ;

# *** Distinctive Ring Settings

Ring1_Caller[2]                  ! "" ;
Ring2_Caller[2]                  ! "" ;
Ring3_Caller[2]                  ! "" ;
Ring4_Caller[2]                  ! "" ;
Ring5_Caller[2]                  ! "" ;
Ring6_Caller[2]                  ! "" ;
Ring7_Caller[2]                  ! "" ;
Ring8_Caller[2]                  ! "" ;

# *** Ring Settings

Default_Ring[2]                  ! "1" ;  # options: 1/2/3/4/5/6/7/8
Default_CWT[2]                   ! "1" ;  # options: 1/2/3/4/5/6/7/8
Hold_Reminder_Ring[2]            ! "8" ;  # options: 1/2/3/4/5/6/7/8/none
Call_Back_Ring[2]                ! "7" ;  # options: 1/2/3/4/5/6/7/8
Cfwd_Ring_Splash_Len[2]          ! "0" ;
Cblk_Ring_Splash_Len[2]          ! "0" ;
VMWI_Ring_Splash_Len[2]          ! ".5" ;
VMWI_Ring_Policy[2]                "New VM Available" ;  # options: New VM Available/New VM
Becomes Available/New VM Arrives
Ring_On_No_New_VM[2]               "No" ;

# *** Call Progress Tones

Dial_Tone                          "350@-19,440@-19;10(*/0/1+2)" ;
Second_Dial_Tone                   "420@-19,520@-19;10(*/0/1+2)" ;
Outside_Dial_Tone                  "420@-16;10(*/0/1)" ;
Prompt_Tone                        "520@-19,620@-19;10(*/0/1+2)" ;
Busy_Tone                          "480@-19,620@-19;10(.5/.5/1+2)" ;
Reorder_Tone                       "480@-19,620@-19;10(.25/.25/1+2)" ;
Off_Hook_Warning_Tone              "480@-10,620@0;10(.125/.125/1+2)" ;
Ring_Back_Tone                     "440@-19,480@-19;*(2/4/1+2)" ;
Confirm_Tone                       "600@-16;1(.25/.25/1)" ;
SIT1_Tone                          "985@-16,1428@-16,1777@-
16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)" ;
SIT2_Tone                          "914@-16,1371@-16,1777@-
16;20(.274/0/1,.274/0/2,.380/0/3,0/4/0)" ;
SIT3_Tone                          "914@-16,1371@-16,1777@-
16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)" ;
SIT4_Tone                          "985@-16,1371@-16,1777@-
16;20(.380/0/1,.274/0/2,.380/0/3,0/4/0)" ;
MWI_Dial_Tone                      "350@-19,440@-19;2(.1/.1/1+2);10(*/0/1+2)" ;
Cfwd_Dial_Tone                     "350@-19,440@-19;2(.2/.2/1+2);10(*/0/1+2)" ;
Holding_Tone                       "600@-19;*(.1/.1/1,.1/.1/1,.1/9.5/1)" ;
Conference_Tone                    "350@-19;20(.1/.1/1,.1/9.7/1)" ;
Secure_Call_Indication_Tone        "397@-19,507@-19;15(0/2/0,.2/.1/1,.1/2.1/2)" ;

# *** Distinctive Ring Patterns

Ring1_Cadence                      "60(2/4)" ;
Ring2_Cadence                      "60(.3/.2,1/.2,.3/4)" ;
Ring3_Cadence                      "60(.8/.4,.8/4)" ;
Ring4_Cadence                      "60(.4/.2,.3/.2,.8/4)" ;
Ring5_Cadence                      "60(.2/.2,.2/.2,.2/.2,1/4)" ;
Ring6_Cadence                      "60(.2/.4,.2/.4,.2/4)" ;
Ring7_Cadence                      "60(.4/.2,.4/.2,.4/4)" ;
Ring8_Cadence                      "60(0.25/9.75)" ;

# *** Distinctive Call Waiting Tone Patterns
```

```
CWT1_Cadence                      "30(.3/9.7)" ;
CWT2_Cadence                      "30(.1/.1, .1/9.7)" ;
CWT3_Cadence                      "30(.1/.1, .3/.1, .1/9.3)" ;
CWT4_Cadence                      "30(.1/.1,.1/.1,.1/9.5)" ;
CWT5_Cadence                      "30(.3/.1,.1/.1,.3/9.1)" ;
CWT6_Cadence                      "30(.1/.1,.3/.2,.3/9.1)" ;
CWT7_Cadence                      "30(.3/.1,.3/.1,.1/9.1)" ;
CWT8_Cadence                      "2.3(.3/2)" ;

# *** Distinctive Ring/CWT Pattern Names

Ring1_Name                        "Bellcore-r1" ;
Ring2_Name                        "Bellcore-r2" ;
Ring3_Name                        "Bellcore-r3" ;
Ring4_Name                        "Bellcore-r4" ;
Ring5_Name                        "Bellcore-r5" ;
Ring6_Name                        "Bellcore-r6" ;
Ring7_Name                        "Bellcore-r7" ;
Ring8_Name                        "Bellcore-r8" ;

# *** Ring and Call Waiting Tone Spec

Ring_Waveform                     "Sinusoid" ;  # options: Sinusoid/Trapezoid
Ring_Frequency                    "25" ;
Ring_Voltage                      "70" ;
CWT_Frequency                     "440@-10" ;

# *** Control Timer Values (sec)

Hook_Flash_Timer_Min              ".1" ;
Hook_Flash_Timer_Max              ".9" ;
Callee_On_Hook_Delay              "0" ;
Reorder_Delay                     "5" ;
Call_Back_Expires                 "1800" ;
Call_Back_Retry_Intvl             "30" ;
Call_Back_Delay                   ".5" ;
VMWI_Refresh_Intvl                "30" ;
Interdigit_Long_Timer             "10" ;
Interdigit_Short_Timer            "3" ;
CPC_Delay                         "2" ;
CPC_Duration                      "0" ;

# *** Vertical Service Activation Codes

Call_Return_Code                  "*69" ;
Blind_Transfer_Code               "*98" ;
Call_Back_Act_Code                "*66" ;
Call_Back_Deact_Code              "*86" ;
Cfwd_All_Act_Code                 "*72" ;
Cfwd_All_Deact_Code               "*73" ;
Cfwd_Busy_Act_Code                "*90" ;
Cfwd_Busy_Deact_Code              "*91" ;
Cfwd_No_Ans_Act_Code              "*92" ;
Cfwd_No_Ans_Deact_Code            "*93" ;
Cfwd_Last_Act_Code                "*63" ;
Cfwd_Last_Deact_Code              "*83" ;
Block_Last_Act_Code               "*60" ;
Block_Last_Deact_Code             "*80" ;
Accept_Last_Act_Code              "*64" ;
Accept_Last_Deact_Code            "*84" ;
CW_Act_Code                       "*56" ;
CW_Deact_Code                     "*57" ;
CW_Per_Call_Act_Code              "*71" ;
CW_Per_Call_Deact_Code            "*70" ;
Block_CID_Act_Code                "*67" ;
Block_CID_Deact_Code              "*68" ;
Block_CID_Per_Call_Act_Code       "*81" ;
Block_CID_Per_Call_Deact_Code     "*82" ;
Block_ANC_Act_Code                "*77" ;
```

```
Block_ANC_Deact_Code              "*87" ;
DND_Act_Code                      "*78" ;
DND_Deact_Code                    "*79" ;
CID_Act_Code                      "*65" ;
CID_Deact_Code                    "*85" ;
CWCID_Act_Code                    "*25" ;
CWCID_Deact_Code                  "*45" ;
Dist_Ring_Act_Code                "*26" ;
Dist_Ring_Deact_Code              "*46" ;
Speed_Dial_Act_Code               "*74" ;
Secure_All_Call_Act_Code          "*16" ;
Secure_No_Call_Act_Code           "*17" ;
Secure_One_Call_Act_Code          "*18" ;
Secure_One_Call_Deact_Code        "*19" ;
Referral_Services_Codes           "" ;
Feature_Dial_Services_Codes       "" ;

# *** Outbound Call Codec Selection Codes

Prefer_G711u_Code                 "*017110" ;
Force_G711u_Code                  "*027110" ;
Prefer_G711a_Code                 "*017111" ;
Force_G711a_Code                  "*027111" ;
Prefer_G723_Code                  "*01723" ;
Force_G723_Code                   "*02723" ;
Prefer_G726r16_Code               "*0172616" ;
Force_G726r16_Code                "*0272616" ;
Prefer_G726r24_Code               "*0172624" ;
Force_G726r24_Code                "*0272624" ;
Prefer_G726r32_Code               "*0172632" ;
Force_G726r32_Code                "*0272632" ;
Prefer_G726r40_Code               "*0172640" ;
Force_G726r40_Code                "*0272640" ;
Prefer_G729a_Code                 "*01729" ;
Force_G729a_Code                  "*02729" ;

# *** Miscellaneous

Set_Local_Date_(mm/dd)            "" ;
Set_Local_Time_(HH/mm)            "" ;
Time_Zone                         "GMT-07:00" ;  # options: GMT-12:00/GMT-11:00/GMT-
10:00/GMT-09:00/GMT-08:00/GMT-07:00/GMT-06:00/GMT-05:00/GMT-04:00/GMT-03:30/GMT-03:00/GMT-
02:00/GMT-
01:00/GMT/GMT+01:00/GMT+02:00/GMT+03:00/GMT+03:30/GMT+04:00/GMT+05:00/GMT+05:30/GMT+05:45/GMT
+06:00/GMT+06:30/GMT+07:00/GMT+08:00/GMT+09:00/GMT+09:30/GMT+10:00/GMT+11:00/GMT+12:00/GMT+13
:00
FXS_Port_Impedance                "600" ;  # options:
600/900/600+2.16uF/900+2.16uF/270+750||150nF/220+820||120nF/220+820||115nF/370+620||310nF
FXS_Port_Input_Gain               "-3" ;
FXS_Port_Output_Gain              "-3" ;
DTMF_Playback_Level               "-16" ;
DTMF_Playback_Length              ".1" ;
Detect_ABCD                       "Yes" ;
Playback_ABCD                     "Yes" ;
Caller_ID_Method                  "Bellcore(N.Amer,China)" ;  # options:
Bellcore(N.Amer,China)/DTMF(Finland,Sweden)/DTMF(Denmark)/ETSI DTMF/ETSI DTMF With PR/ETSI
DTMF After Ring/ETSI FSK/ETSI FSK With PR(UK)
FXS_Port_Power_Limit              "3" ;  # options: 1/2/3/4/5/6/7/8
Protect_IVR_FactoryReset          "No" ;
```

3.5.1.1.1. Internal Error Codes

The SPA defines a number of internal error codes (X00–X99) to facilitate configuration in providing finer control over the behavior of the unit under certain error conditions. They can be viewed as extensions to the SIP response codes 100–699. The definitions are shown below

Error Code	Description
X00	Transport layer (or ICMP) error when sending a SIP request
X20	SIP request times out while waiting for a response
X40	General SIP Protocol Error (e.g., unacceptable codec in SDP in 200 and ACK messages, or times out while waiting for ACK)
X60	Dialed number invalid according to given dial plan

3.5.1.1.2. Data Types

- **Uns<n>** – Unsigned n-bit value, where n = 8, 16, or 32. It can be specified in decimal or hex format such as 12 or 0x18 as long as the value can fit into n bits.
- **Sig<n>** – Signed n-bit value. It can be specified in decimal or hex format. Negative values must be preceded by a "-" sign. A '+' sign before positive value is optional
- **Str<n>** – A generic string with up to n non-reserved characters.
- **Float<n>** – A floating point value with up to n decimal places.
- **Time<n>** – Time duration in seconds, with up to n decimal places. Extra decimal places specified are ignored.
- **PwrLevel** – Power level expressed in dBm with 1 decimal place, such as –13.5 or 1.5 (dBm)
- **Bool**: Boolean value of either "yes" or "no"
- **{a,b,c,…}** – A choice among a, b, c, …
- **IP** – IP Address in the form of x.x.x.x, where x between 0 and 255. For example 10.1.2.100
- **Port** – TCP/UDP Port number (0-65535). It can be specified in decimal of hex format.
- **UserID** – User ID as appeared in a URL; up to 63 characters
- **FQDN** – Fully Qualified Domain Name, such as "sip.sipura.com:5060", or "109.12.14.12:12345". It can contain up to 63 characters
- **Phone** – A phone number string, such as 14081234567, *69, *72, 345678, or a generic URL such as 1234@10.10.10.100:5068, or jsmith@sipura.com. It can contain up to 39 characters.
- **ActCode** – Activation code for a supplementary service, such as *69. It can contain up to 7 characters.
- **PhTmplt** – A phone number template. Each template may contain 1 or more patterns separated by a ",". White space at the beginning of each pattern is ignored. "?" and "*" represent wildcard characters. It can contain up to 39 characters. Examples: "1408*, 1510*", "1408123????, 555?1".
- **RscTmplt** – A template of SIP Response Status Code, such as "404, 5*", "61?", "407, 408, 487, 481". It can contain up to 39 characters.
- **CadScript** – A mini-script that specifies the cadence parameters of a signal. Up to 127 characters. Syntax: $S_1[;S_2]$, where $S_i=D_i(on_{i,1}/off_{i,1}[,on_{i,2}/off_{i,2}[,on_{i,3}/off_{i,3}[,on_{i,4}/off_{i,4}[,on_{i,5}/off_{i,5}[,on_{i,6}/off_{i,6}]]]]])$ and is known as a *section*, $on_{i,j}$ and $off_{i,j}$ are the on/off duration in seconds of a *segment* and i = 1 or 2, and j = 1 to 6. D_i is the total duration of the section in seconds. All durations can have up to 3 decimal places to provide 1 ms resolution. The wildcard character "*" stands for infinite duration. The segments within a section are played in order and repeated until the total duration is played. Examples:

```
Example 1: Normal Ring

     60(2/4)

  Number of Cadence Sections = 1
     Cadence Section 1: Section Length = 60 s
        Number of Segments = 1
           Segment 1: On=2s, Off=4s

  Total Ring Length = 60s
```

```
Example 2: Distinctive Ring (short,short,short,long)

     60(.2/.2,.2/.2,.2/.2,1/4)

  Number of Cadence Sections = 1
     Cadence Section 1: Section Length = 60s
        Number of Segments = 4
           Segment 1: On=0.2s, Off=0.2s
           Segment 2: On=0.2s, Off=0.2s
           Segment 3: On=0.2s, Off=0.2s
           Segment 4: On=1.0s, Off=4.0s

  Total Ring Length = 60s
```

- **FreqScript** – A mini-script that specifics the frequency and level parameters of a tone. Up to 127 characters. Syntax: $F_1@L_1[,F_2@_{L2}[,F_3@L_3[,F_4@L_4[,F_5@L_5[,F_6@L_6]]]]]$, where F_1–F_6 are frequency in Hz (unsigned integers only) and L_1–L_6 are corresponding levels in dBm (with up to 1 decimal places). White spaces before and after the comma are allowed (but not recommended)

```
Example 1: Call Waiting Tone

     440@-10

  Number of Frequencies = 1
     Frequency 2 = 440 Hz at –10 dBm


Example 2: Dial Tone

     350@-19,440@-19

  Number of Frequencies = 2
     Frequency 1 = 350 Hz at –19 dBm
     Frequency 2 = 440 Hz at –19 dBm
```

- **ToneScript** – A mini-script that specifies the frequency, level and cadence parameters of a call progress tone. May contain up to 127 characters. Syntax: FreqScript;$Z_1[;Z_2]$. The section Zi is similar to the S_i section in a CadScript except that each on/off segment is followed by a frequency components parameter: $Z_i = D_i(on_{i,1}/off_{i,1}/f_{i,1}[,on_{i,2}/off_{i,2}/f_{i,2}\ [,on_{i,3}/off_{i,3}/f_{i,3}\ [,on_{i,4}/off_{i,4}/f_{i,4}\ [,on_{i,5}/off_{i,5}/f_{i,5}\ [,on_{i,6}/off_{i,6}/f_{i,6}]]]]])$, where fi,j = $n_1[+n_2]+n_3[+n_4[+n_5[+n_6]]]]$ and $1 < n_k < 6$ indicates which of the frequency components given in the FreqScript shall be used in that segment; if more than one frequency component is used in a segment, the components are summed together.

```
Example 1: Dial Tone

     350@-19,440@-19;10(*/0/1+2)

  Number of Frequencies = 2
     Frequency 1 = 350 Hz at –19 dBm
     Frequency 2 = 440 Hz at –19 dBm
  Number of Cadence Sections = 1
     Cadence Section 1: Section Length = 10 s
```

```
        Number of Segments = 1
           Segment 1: On=forever, with Frequencies 1 and 2

  Total Tone Length = 10s


Example 2: Stutter Tone

     350@-19,440@-19;2(.1/.1/1+2);10(*/0/1+2)

  Number of Frequencies = 2
     Frequency 1 = 350 Hz at –19 dBm
     Frequency 2 = 440 Hz at –19 dBm
  Number of Cadence Sections = 2
     Cadence Section 1: Section Length = 2s
        Number of Segments = 1
           Segment 1: On=0.1s, Off=0.1s with Frequencies 1 and 2
     Cadence Section 2: Section Length = 10s
        Number of Segments = 1
           Segment 1: On=forever, with Frequencies 1 and 2

  Total Tone Length = 12s


Example 3: SIT Tone

     985@-16,1428@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)

  Number of Frequencies = 3
     Frequency 1 = 985 Hz at –16 dBm
     Frequency 2 = 1428 Hz at –16 dBm
     Frequency 3 = 1777 Hz at –16 dBm
  Number of Cadence Sections = 1
     Cadence Section 1: Section Length = 20s
        Number of Segments = 4
           Segment 1: On=0.38s, Off=0s, with Frequency 1
           Segment 2: On=0.38s, Off=0s, with Frequency 2
           Segment 3: On=0.38s, Off=0s, with Frequency 3
           Segment 4: On=0s, Off=4s, with no frequency components

  Total Tone Length = 20s
```

- **ProvisioningRuleSyntax** – Scripting syntax used to define configuration resync and firmware upgrade rules. Refer to the provisioning discussion for an explanation of the syntax.
- **DialPlanScript** – Scripting syntax used to specify line 1 and line 2 dial plans. Refer to the dial plan section of this document for an explanation.

3.5.1.1.3. Notations

- **<Par Name>** represents a configuration parameter name. In a profile, the corresponding tag is formed by replacing the space with an underscore "_", such as **Par_Name**.
- An empty default value field implies an empty string < **""** >.
- The SPA shall continue to use the last configured values for tags that are not present in a given profile.
- Templates are compared in the order given. The first, **not the closest**, match is selected. The parameter name must match exactly.

- If more than one definition for a parameter is given in a configuration file, the last such definition in the file is the one that will take effect in the SPA.
- A parameter specification with an empty parameter value forces the parameter back to its default value. To specify an empty string instead, use the empty string **“”** as the parameter value.

3.5.2. Dial Plan

The SPA allows each line to be configured with a distinct dial plan. The dial plan specifies how to interpret digit sequences dialed by the user, and how to convert those sequences into an outbound dial string.

The SPA syntax for the dial plan closely resembles the corresponding syntax specified by MGCP and MEGACO. Some extensions are added that are useful in an end-point.

The dial plan functionality is regulated by the following configurable parameters:

- Interdigit_Long_Timer
- Interdigit_Short_Timer
- Dial_Plan ([1] and [2])

Other timers are configurable via parameters, but do not directly pertain to the dial plan itself. They are discussed elsewhere in this document.

Interdigit Long Timer:

```
ParName:    Interdigit_Long_Timer
Default:    10
```

The Interdigit_Long_Timer specifies the default maximum time (in seconds) allowed between dialed digits, when no candidate digit sequence is as yet complete (see discussion of Dial_Plan parameter for an explanation of candidate digit sequences).

Interdigit Short Timer:

```
ParName:    Interdigit_Short_Timer
Default:    3
```

The Interdigit_Short_Timer specifies the default maximum time (in seconds) allowed between dialed digits, when at least one candidate digit sequence is complete as dialed (see discussion of Dial_Plan parameter for an explanation of candidate digit sequences).

Dial Plan[1] and Dial Plan[2]:

```
ParName:    Dial_Plan[1] and Dial_Plan[2]
```

```
Default:     ( *xx | [3469]11 | 0 | 00 | <:1408>[2-9]xxxxxx |
               1[2-9]xx[2-9]xxxxxx | 011x. )
```

The Dial_Plan parameters contain the actual dial plan scripts for each of lines 1 and 2.

Dial Plan Digit Sequences:

The plans contain a series of digit sequences, separated by the '|' character. The collection of sequences is enclosed in parentheses, '(' and ')'.

When a user dials a series of digits, each sequence in the dial plan is tested as a possible match. The matching sequences form a set of candidate digit sequences. As more digits are entered by the user, the set of candidates diminishes until only one or none are valid.

Any one of a set of terminating events triggers the SPA to either accept the user-dialed sequence, and transmit it to initiate a call, or else reject it as invalid. The terminating events are:

- No candidate sequences remain: the number is rejected.
- Only one candidate sequence remains, and it has been matched completely: the number is accepted and transmitted after any transformations indicated by the dial plan, unless the sequence is barred by the dial plan (barring is discussed later), in which case the number is rejected.
- A timeout occurs: the digit sequence is accepted and transmitted as dialed if incomplete, or transformed as per the dial plan if complete.
- An explicit 'send' (user presses the '#' key): the digit sequence is accepted and transmitted as dialed if incomplete, or transformed as per the dial plan if complete.

The timeout duration depends on the matching state. If no candidate sequences are as yet complete (as dialed), the Interdigit_Long_Timeout applies. If a candidate sequence is complete, but there exists one or more incomplete candidates, then the Interdigit_Short_Timeout applies.

White space is ignored, and may be used for readability.

Digit Sequence Syntax:

Each digit sequence within the dial plan consists of a series of elements, which are individually matched to the keys pressed by the user. Elements can be one of the following:

- Individual keys '0', '1', '2' . . . '9', '*', '#'.
- The letter 'x' matches any one numeric digit ('0' .. '9')
- A subset of keys within brackets (allows ranges): '[' set ']' (e.g. [389] means '3' or '8' or '9')
    - Numeric ranges are allowed within the brackets: digit '-' digit (e.g. [2-9] means '2' or '3' or … or '9')
    - Ranges can be combined with other keys: e.g. [235-8*] means '2' or '3' or '5' or '6' or '7' or '8' or '*'.

Element repetition:

Any element can be repeated zero or more times by appending a period ('.' character) to the element. Hence, "01." matches "0", "01", "011", "0111", … etc.

Subsequence Substitution:

A subsequence of keys (possibly empty) can be automatically replaced with a different subsequence using an angle bracket notation: ‘<’ dialed-subsequence ‘:’ transmitted-subsequence ‘>’. So, for example, “<8:1650>xxxxxxx” would match “85551212” and transmit “16505551212”.

Intersequence Tones:

An “outside line” dial tone can be generated within a sequence by appending a ‘,’ character between digits. Thus, the sequence “9, 1xxxxxxxxxx” sounds an “outside line” dial tone after the user presses ‘9’, until the ‘1’ is pressed.

Number Barring:

A sequence can be barred (rejected) by placing a ‘!’ character at the end of the sequence. Thus, “1900xxxxxxx!” automatically rejects all 900 area code numbers from being dialed.

Interdigit Timer Master Override:

The long and short interdigit timers can be changed in the dial plan (affecting a specific line) by preceding the entire plan with the following syntax:

- Long interdigit timer: ‘L’ ‘:’ delay-value ‘,’
- Short interdigit timer: ‘S’ ‘:’ delay-value ‘,’

Thus, “L=8,(. . .)” would set the interdigit long timeout to 8 seconds for the line associated with this dial plan. And, “L:8,S:4,(. . .)” would override both the long and the short timeout values.

Local Timer Overrides:

The long and short timeout values can be changed for a particular sequence starting at a particular point in the sequence. The syntax for long timer override is: ‘L’ delay-value ‘ ‘. Note the terminating space character. The specified delay-value is measured in seconds. Similarly, to change the short timer override, use: ‘S’ delay-value <space>.

Pause:

A sequence may require an explicit pause of some duration before continuing to dial digits, in order for the sequence to match. The syntax for this is similar to the timer override syntax: ‘P’ delay-value <space>. The delay-value is measured in seconds.

This syntax allows for the implementation of Hot-Line and Warm-Line services. To achieve this, one sequence in the plan must start with a pause, with a 0 delay for a Hot Line, and a non-zero delay for a Warm Line.

Implicit sequences:

The SPA implicitly appends the vertical code sequences entered in the Regional parameter settings to the end of the dial plan for both line 1 and line 2. Likewise, if Enable_IP_Dialing is enabled, then ip dialing is also accepted on the associated line.

Examples:

The following dial plan accepts only US-style 1 + area-code + local-number, with no restrictions on the area code and number.

```
( 1 xxx xxxxxxx )
```

The following also allows 7-digit US-style dialing, and automatically inserts a 1 + 212 (local area code) in the transmitted number.

```
( 1 xxx xxxxxxx | <:1212> xxxxxxx )
```

For an office environment, the following plan requires a user to dial 8 as a prefix for local calls and 9 as a prefix for long distance. In either case, an “outside line” tone is played after the initial 8 or 9, and neither prefix is transmitted when initiating the call.

```
( <9,:> 1 xxx xxxxxxx | <8,:1212> xxxxxxx )
```

The following allows only placing international calls (011 call), with an arbitrary number of digits past a required 5 digit minimum, and also allows calling an international call operator (00). In addition, it lengthens the default short interdigit timeout to 4 seconds.

```
S:4, ( 00 | 011 xxxxx x. )
```

The following allows only US-style 1 + area-code + local-number, but disallows area codes and local numbers starting with 0 or 1. It also allows 411, 911, and operator calls (0).

```
( 0 | [49]11 | 1 [2-9]xx [2-9]xxxxxx )
```

The following allows US-style long distance, but blocks 9xx area codes.

```
( 1 [2-8]xx [2-9]xxxxxx )
```

The following allows arbitrary long distance dialing, but explicitly blocks the 947 area code.

```
( 1 947 xxxxxxx ! | 1 xxx xxxxxxx )
```

The following implements a Hot Line phone, which automatically calls 1 212 5551234.

```
( S0 <:12125551234> )
```

The following provides a Warm Line to a local office operator (1000) after 5 seconds, unless a 4 digit extension is dialed by the user.

```
( P5 <:1000> | xxxx )
```

3.5.3. System Parameters

System Configuration

Parameter Name	Description	Type	Default
Restricted Access Domains	This feature is used when implementing software customization.	Str127	
Enable Web Server	Enable/disable web server of SPA **This feature should only be used on firmware version 1.0.9 or later.**	Bool	Yes
Enable Web Admin Access	Enable/disable Admin pages of web server of SPA	Bool	Yes
Admin Password	The password for administrator	Str63	
User Password	The password for User	Str63	

Network Configuration

Parameter Name	Description	Type	Default
DHCP	Enable/Disable DHCP	Bool	Yes
Host Name	Host Name of SPA	Str31	
Domain	The network domain of SPA	Str127	
Static IP	Static IP address of SPA, which will take effect if DHCP is disabled	IP	0.0.0.0
NetMask	The NetMask used by SPA when DHCP is disabled	IP	255.255.255.0
Gateway	The default gateway used by SPA when DHCP is disabled	IP	0.0.0.0
Primary DNS	DNS server used by SPA in addition to DHCP supplied DNS servers if DHCP is enabled; when DHCP is disabled, this will be the primary DNS server.	IP	0.0.0.0
Secondary DNS	DNS server used by SPA in addition to DHCP supplied DNS servers if DHCP is enabled; when DHCP is disabled, this will be the secondary DNS server.	IP	0.0.0.0
DNS Query Mode	Do parallel or sequential DNS Query	Choice	Parallel
Syslog Server	Specify the Syslog server name and port. This feature specifies the server for logging SPA system information and critical events.	FQDN	
Debug Server	The debug server name and port. This feature specifies the server for logging SPA debug information. The level of detailed output depends on the debug level parameter setting.	FQDN	
Debug Level	The higher the debug level, the more debug information will be generated. Zero (0) means no debug information will be generated.	Choice	0
Primary NTP Server	IP address or name of primary NTP server.	Str127 or IP	
Secondary NTP Server	IP address or name of secondary NTP server	Str127 or IP	
Web Server Port	TCP port through which the SPA web server will communicate	Uns8	80

Notes:

- Parallel DNS query mode: SPA will send the same request to all the DNS servers at the same time when doing a DNS lookup, the first incoming reply will be accepted by SPA.
- To log SIP messages, Debug Level must be set to at least 2.
- If both Debug Server and Syslog Server are specified, _Syslog messages are also logged to the Debug Server.

3.5.4. Provisioning Parameters

Provisioning operations are gated by the Provision_Enable parameter.

Parameter Name	Description	Type	Default
Provision Enable	Master enable for configuration profile resync operations	Bool	yes
Resync On Reset	Resyncs configuration profile from configuration server whenever the SPA resets.	Bool	yes
Resync Random Delay	Spread interval for resync requests	Time0	2
Resync Periodic	Resyncs configuration profile periodically after reset.	Time0	3600
Resync Error Retry Delay	Retry interval following resync failure	Time0	3600
Resync From SIP	Enables resync of configuration profile from a SIP command.	Bool	Yes
Resync After Upgrade Attempt		Bool	Yes
Resync Trigger 1			
Resync Trigger 2			
Profile Rule	Configuration profile URL script.	ProfileScript	/spa.cfg
Profile Rule B		ProfileScript	
Profile Rule C		ProfileScript	
Profile Rule D		ProfileScript	
Log Resync Request Msg	Syslog message generated when attempting a resync	ProfileMsg	See provisioning discussion section
Log Resync Success Msg	Syslog message generated after a successful resync	ProfileMsg	See provisioning discussion section
Log Resync Failure Msg	Syslog message generated after a failed resync	ProfileMsg	See provisioning discussion section
GPP A thru GPP P	General purpose parameter	String	empty
GPP SA thru GPP SD	General purpose parameter	String	empty

Note: In a customized SPA, the profile rule would point to a service provider’s server.

3.5.5. Upgrade Parameters

Parameter Name	Description	Type	Default
Upgrade Enable	Master enable for firmware upgrade	Bool	Yes

	operations		
Upgrade Error Retry Delay	Retry interval following upgrade failure	Time0	3600
Upgrade Rule	Upgrade script.	UpgradeScript	empty
Log Upgrade Request Msg	Syslog message generated when attempting an upgrade	UpgradeMsg	See provisioning discussion section
Log Upgrade Success Msg	Syslog message generated after a successful upgrade	UpgradeMsg	See provisioning discussion section
Log Upgrade Failure Msg	Syslog message generated after a failed upgrade	UpgradeMsg	See provisioning discussion section

Note: In a customized SPA, the upgrade rule would point to a service provider's server.

3.5.6. Protocol Parameters

Parameter Name	Description	Type	Default
SIP Parameters			
Max Forward	SIP Max-Forward value. Range: 1 – 255	Uns8	70
Max Redirection	Number of times to allow an INVITE to be redirected by a 3xx response to avoid an infinite loop. **Note: This parameter currently has no effect: there is no limit on number of redirection.**	Uns8	5
Max Auth	Maximum number of times a request may be challenged (0-255)	Uns8	2
SIP User Agent Name	User-Agent Header to be used by the unit in outbound requests. If empty, the header is not included.	Str63	Sipura/ $version
SIP Server Name	Server Header to used by the unit in responses to inbound responses. If empty, the header is not included.	Str63	Sipura/ $version
SIP Accept Language	Accept-Language Header to be used by the unit. If empty, the header is not included.	Str31	
Remove Last Reg	Remove last registration before registering a new one if value is different one.	Bool	no
DTMF Relay MIME Type	This is the MIME Type to be used in a SIP INFO message used to signal DTMF event.	Str31	application/dtmf-relay
Hook Flash MIME Type	This is the MIME Type to be used in a SIP INFO message used to signal hook flash event.	Str31	application/hook-flash
Use Compact Header	If set to yes, the SPA will use compact SIP headers in outbound SIP messages. If set to no the SPA will use normal SIP headers.	Bool	no

SIP Timer Values (sec)			
SIP T1	RFC 3261 T1 value (RTT Estimate). Range: 0 – 64 sec	Time3	.5
SIP T2	RFC 3261 T2 value (Maximum retransmit interval for non-INVITE requests and INVITE responses). Range: 0 – 64 sec	Time3	4
SIP T4	RFC 3261 T4 value (Maximum duration a message will remain in the network). Range: 0 – 64 sec	Time3	5
SIP Timer B	INVITE time out value. Range: 0 – 64 sec	Time3	32
SIP Timer F	Non-INVITE time out value. Range: 0 – 64 sec	Time3	32
SIP Timer H	INVITE final response time out value. Range: 0 – 64 sec	Time3	32
SIP Timer D	ACK hang around time. Range: 0 – 64 sec	Time3	32
SIP Timer J	Non-INVITE response hang around time. Range: 0 – 64 sec	Time3	32
INVITE Expires	INVITE request Expires header value in sec. 0 = do not include Expires header in INVITE. Range: 0 – (2^{31} – 1)	Time0	180
ReINVITE Expires	ReINVITE request Expires header value in sec. 0 = do not include Expires header in the request. Range: 0 – (2^{31} – 1)	Time0	30
Reg Min Expires	Minimum registration expiration time allowed from the proxy in the Expires header or as a Contact header parameter. If proxy returns something less this value, then the minimum value is used.	Time0	1
Reg Max Expires	Maximum registration expiration time allowed from the proxy in the Min-Expires header. If value is larger than this, then the maximum value is used	Time0	7200
Reg Retry Intvl	Interval to wait before the SPA retries registration again after encountering a failure condition during last registration	Time0	30
Reg Retry Long Interval	When Registration fails with a SIP response code that does no match <Retry Reg RSC>, the SPA will wait for the delay specified in this parameter before retrying. If this parameter is 0, the SPA will stop retrying. This value should be much larger than <Reg Retry Intvl> which should not be 0.	Time0	1200
Response Status Code Handling			
SIT1 RSC[1]	SIP response status code to INVITE on which to play the SIT1 Tone	RscTmplt	
SIT2 RSC[1]	SIP response status code to INVITE on which to play the SIT2 Tone	RscTmplt	
SIT3 RSC[1]	SIP response status code to INVITE on which to play the SIT3 Tone	RscTmplt	
SIT4 RSC[1]	SIP response status code to INVITE on which to play the SIT4 Tone	RscTmplt	

Try Backup RSC	SIP response status code on which to retry a backup server for the current request	RscTmplt	
Retry Reg RSC	Interval to wait before the SPA retries registration again after encountering a failure condition during last registration	Time0	30
RTP Parameters			
RTP Port Min[2]	Minimum port number for RTP transmission and reception	Port	16384
RTP Port Max[2]	Maximum port number for RTP transmission and reception	Port	16482
RTP Packet Size	Packet size in sec. Valid values must be multiple of 0.01s. Range: 0.01 – 0.16	Time3	0.02
RTCP Tx Interval[4]	Controls the interval (sec) to send out RTCP sender report on an active connection. Range: 0 – 255 (s)	Time0	0

Notes:

1. Reorder or Busy Tone will be played by default for all unsuccessful response status code
2. <RTP Port Min> and <RTP Port Max> should define a range that contains at least 4 even number ports, such as 100 – 106
3. If inbound SIP requests contain compact headers, SPA will reuse the same compact headers when generating the response regardless the settings of the <Use Compact Header> parameter. If inbound SIP requests contain normal headers, SPA will substitute those headers with compact headers (if defined by RFC 261) if <Use Compact Header> parameter is set to “yes.”
4. During an active connection, the SPA can be programmed to send out compound RTCP packet on the connection. Each compound RTP packet except the last one contains a SR (Sender Report) and a SDES.(Source Description). The last RTCP packet contains an additional BYE packet. Each SR except the last one contains exactly 1 RR (Receiver Report); the last SR carries no RR. The SDES contains CNAME, NAME, and TOOL identifiers. The CNAME is set to <User ID>@<Proxy>, NAME is set to <Display Name> (or “Anonymous” if user blocks caller ID), and TOOL is set to the Verdor/Hardware-platform-software-version (such as Sipura/SPA2000-1.0.31(b)). The NTP timestamp used in the SR is a snapshot of the SPA’s local time, not the time reported by an NTP server. If the SPA receives a RR from the peer, it will attempt to compute the round trip delay and show it as the <Call Round Trip Delay> value (ms) in the Info section of SPA web page.

3.5.6.1. Dynamic Payload Types

Parameter Name	Description	Type	Default
NSE Dynamic Payload[1,2]	NSE dynamic payload type	Uns8	100
AVT Dynamic Payload[1,2]	AVT dynamic payload type	Uns8	101
G726r16 Dynamic Payload[1,2]	G726-16 dynamic payload type	Uns8	98
G726r24 Dynamic Payload[1,2]	G726-24 dynamic payload type	Uns8	97
G726r40 Dynamic Payload[1,2]	G726-40 dynamic payload type	Uns8	96
G729b Dynamic Payload[1,2]	G729b dynamic payload type	Uns8	99

Notes:

1. Valid range is 96 – 127

2. The configured dynamic payloads are used for outbound calls only where the SPA presents the SDP offer. For inbound calls with a SDP offer, SPA will follow the caller’s dynamic payload type assignments

3.5.6.2. SDP Audio Codec Names

Parameter Name	Description	Type	Default
NSE Codec Name	NSE Codec name used in SDP	Str31	NSE
AVT Codec Name	AVT Codec name used in SDP	Str31	telephone-event
G711a Codec Name	G711a Codec name used in SDP	Str31	PCMA
G711u Codec Name	G711u Codec name used in SDP	Str31	PCMU
G726r16 Codec Name	G726-16 Codec name used in SDP	Str31	G726-16
G726r24 Codec Name	G726-24 Codec name used in SDP	Str31	G726-24
G726r32 Codec Name	G726-32 Codec name used in SDP	Str31	G726-32
G726r40 Codec Name	G726-40 Codec name used in SDP	Str31	G726-40
G729a Codec Name	G729a Codec name used in SDP	Str31	G729a
G729b Codec Name	G729b Codec name used in SDP	Str31	G729ab
G723 Codec Name	G723 Codec name used in SDP	Str31	G723

Notes:

1. SPA uses the configured codec names in its outbound SDP
2. SPA ignores the codec names in incoming SDP for standard payload types (0 – 95).
3. For dynamic payload types, SPA identifies the codec by the configured codec names. Comparison is case-insensitive.

3.5.6.3. NAT Support

Parameter Name	Description	Type	Default
Handle_VIA_received	If set to “yes”, the SPA will process the “received” parameter in the VIA header inserted by the server in a response to any one of its request. Else the parameter is ignored.	Bool	No
Handle_VIA_rport	If set to “yes”, the SPA will process the “rport” parameter in the VIA header inserted by the server in a response to any one of its request. Else the parameter is ignored.	Bool	No
Insert VIA received	Insert received parameter in VIA header in SIP responses if received from IP and VIA sent-by IP differ	Bool	No
Insert VIA rport	Insert rport parameter in VIA header in SIP responses if received-from port and VIA sent-by port differ	Bool	No
Substitute VIA addr	Use nat-mapped IP:port values in VIA header	Bool	No
Send Resp To Src Port	Send response to the request source port instead of the VIA sent-by port	Bool	No
STUN Server	STUN server to contact for NAT mapping discovery	FQDN	
STUN Enable	Enable the use of STUN to discover NAT mapping	Bool	No
STUN Test Enable	If enabled with <STUN Enable> = “yes” and a valid <STUN Server>, the SPA will perform a NAT type discovery operation when first power on by contacting the configured STUN server. The result of the discovery will be reported in a Warning header in all subsequent REGISTER requests – “Warning: 399 spa <stun type>”, where <stun type> is one of the following:	Bool	No

	"Unknown NAT Type", "STUN Server Not Reachable", "STUN Server Not Responding", "Open Internet Detected", "Symmetric Firewall Detected", "Full Cone NAT Detected", "Restricted Cone NAT Detected", "Symmetric NAT Detected"; If the SPA detects Symmetric Nat or Symmetric Firewall, Nat Mapping will be disabled (that is, no substitution of IP address and port with external IP address an nat-mapped port)		
Ext IP	External IP address to substitute for the actual IP address of the unit in all outgoing SIP messages. If “0.0.0.0” is specified, no IP address substitution is performed.	IP	0.0.0.0
Ext RTP Port Min	External port mapping of <RTP Port Min>. If this value is non-zero, the RTP port number in all outgoing SIP messages is substituted by the corresponding port value in the external RTP port range.	Port	0
NAT Keep Alive Intvl	Interval between sending NAT-mapping keep alive message in sec	Uns16	15

Notes:

3.5.7. Line 1 and Line 2 Parameters

Per line parameter tags must be appended with [1] or [2] (corresponding to lines 1 or 2) in the configuration profile. It is omitted below for readability.

3.5.7.1. User Account Information

Parameter Name	Description	Type	Default
Line Enable	Enable this line for service	Bool	Yes
MOH Server[2]	The User ID or URL of the auto-answering SAS to contact for MOH services. Examples: 5000, 1001@music.sipura.com, 66.12.123.15:5061. Note: When only a user-id is given, the current proxy or outbound proxy will be contacted as in the making of a regular outbound call. MOH is disabled if this parameter is not specified (empty).	Str127	Empty
SIP Port	SIP message listening port and transmission port	Port	5060
Ext SIP Port	External port to substitute for the actual SIP port of the unit in all outgoing SIP messages. If “0” is specified, no SIP port substitution is performed.	Port	0
SIP TOS/DiffServ Value	TOS/DiffServ field value in UDP IP Packets carrying a SIP Message	Byte	0x68
RTP TOS/DiffServ Value	TOS/DiffServ field value in UDP IP Packets carrying a RTP data	Byte	0xb8
SAS Enable[3]	Enables the FXS Line to act as a Streaming Audio Source (SAS). If enabled, the line cannot be used for making outgoing calls. Instead, it auto-answers	Bool	No

	incoming calls and streams audio RTP packets to the calling party.		
SAS DLG Refresh Intvl[3]	If non-zero, this is the interval at which SAS sends out session refresh (SIP re-INVITE) messages to detect if connection to the caller is still up. If the caller does not respond to refresh message, SPA will terminate this call with a SIP BYE message. The default = 0 (Session refresh disabled) Range = 0-255 (s)		0
SAS Inbound RTP Sink[3]	The purpose of this parameter is to work around devices that do not play inbound RTP if the SAS line declares itself as a “sendonly” device and tells the client not to stream out audio. This parameter is a FQDN or IP address of a RTP sink to be used by the SPA SAS line in the SDP of its 200 response to inbound INVITE from a client. It will appear in the c = line and the port number and, if specified, in the m = line of the SDP. If this value is not specified or equal to 0, then c = 0.0.0.0 and a=sendonly will be used in the SDP to tell the SAS client to not to send any RTP to this SAS line. If a non-zero value is specified, then a=sendrecv and the SAS client will stream audio to the given address. Special case: If the value is $IP, then the SAS line’s own IP address is used in the c = line and a=sendrecv. In that case the SAS client will stream RTP packets to the SAS line. The default value is [empty].	Str63	
NAT Mapping Enable	Enable the use of externally mapped of IP address and SIP/RTP ports in SIP messages. The mapping may be discovered by any of the supported methods.	Bool	No
NAT Keep Alive Enable	If set to “yes”, the configured <NAT Keep Alive Msg> is sent periodically every <NAT Keep Alive Intvl> seconds.	Bool	No
NAT Keep Alive Msg	Contents of the keep-alive message to be sent to a given destination periodically to maintain the current NAT-mapping. It could be an empty string. If value is $NOTIFY, a NOTIFY message is sent as keep alive. If value is $REGISTER, a REGISTER message w/o Contact is sent.	Str31	$NOTIFY
NAT Keep Alive Dest	Destination to send NAT keep alive messages to. If value is $PROXY, it will be sent to the current proxy or outbound proxy	FQDN	$PROXY
SIP Debug Option	None, 1-line, full, exclude OPTIONS, exclude REGISTER, exclude NOTIFY, …	Choice	none
Network Jitter Level	4 settings are available: very high, high, medium, low. This parameter affects how jitter buffer size is adjusted in the SPA. Jitter buffer size is adjusted dynamically. The minimum jitter buffer size is 30 ms or (10 ms + current RTP frame size), which ever is larger, for all jitter level settings. But the starting jitter buffer size value is larger for higher jitter levels. This parameter controls the rate at which to adjust the jitter buffer size to reach the	Choice	High

	minimum. If the jitter level is set to high, then the rate of buffer size decrement is slower (more conservative), else faster (more aggressive).		
SIP 100REL Enable	Enable the support or the 100rel SIP extension for reliable transmission of provisional responses (18x) and the use of PRACK requests.	Bool	No
Blind Attn-Xfer Enable	If enabled, the SPA performs an attended transfer operation by terminating the current call leg, and blind transferring the other call leg. If disabled, the SPA performs an attended transfer by referring the other call leg to the current call leg while maintaining both call legs.	Bool	No
Proxy and Registration			
Proxy	SIP Proxy Server for all outbound requests	FQDN	
Use Outbound Proxy	Enable the use of <Outbound Proxy>. If set to “no”, <Outbound Proxy> and <Use OB Proxy in Dialog) is ignored.	Bool	No
Outbound Proxy	SIP Outbound Proxy Server where all outbound requests are sent as the first hop.	FQDN	No
Use OB Proxy In Dialog	Whether to forcer SIP requests to be sent to the outbound proxy within a dialog. Ignored if <Use Outbound Proxy> is “no” or <Outbound Proxy> is empty	Bool	Yes
Register	Enable periodic registration with the <Proxy>. This parameter is ignored if <Proxy> is not specified.	Bool	Yes
Make Call Without Reg	Allow making outbound calls without successful (dynamic) registration by the unit. If “No”, dial tone will not play unless registration is successful	Bool	No
Ans Call Without Reg	Allow answering inbound calls without successful (dynamic) registration by the unit	Bool	No
Register Expires[1]	Expires value in sec in a REGISTER request. SPA will periodically renew registration shortly before the current registration expired. This parameter is ignored if <Register> is “no”. Range: 0 – (2^{31} – 1) sec	Time0	3600
Use DNS SRV	Whether to use DNS SRV lookup for Proxy and Outbound Proxy	Bool	No
DNS SRV Auto Prefix	If enabled, the SPA will automatically prepend the Proxy or Outbound Proxy name with _sip._udp when performing a DNS SRV lookup on that name	Bool	No
Proxy Fallback Intvl	This parameter sets the delay (sec) after which the SPA will retry from the highest priority proxy (or outbound proxy) servers after it has failed over to a lower priority server. This parameter is useful only if the primary and backup proxy server list is provided to the SPA via DNS SRV record lookup on the server name. (Using multiple DNS A record per server name does not allow the notion of priority and so all hosts will be considered at the same priority and the SPA will not attempt to fall back after a fail over)	Time0	3600

Subscriber Information			
Display Name	Subscriber's display name to appear in caller-id	Str23	
User ID	Subscriber's user-id. Usually a E.164 number	Str47	
Password	Subscriber's a/c password	Str23	
Auth ID	Subscriber's authentication ID	Str39	
Use Auth ID	If set to "yes", the pair <Auth ID> and <Password> are used for SIP authentication. Else the pair <User ID> and <Password> are used.	Bool	No
Mini Certificate	Base64 encoded of Mini-Certificate concatenated with the 1024-bit public key of the CA signing the MC of all subscribers in the group.	Str508	Empty
SRTP Private Key	Base64 encoded of the 512-bit private key per subscriber for establishment of a secure call.	Str88	Empty

Notes:

1. If proxy responded to REGISTER with a smaller Expires value, the SPA will renew registration based on this smaller value instead of the configured value. If registration failed with an "Expires too brief" error response, the SPA will retry with the value given in the Min-Expires header in the error response.

2. MOH Notes:

• The remote party must indicate that it can receive audio while holding MOH to work. That is the SIP 2xx response from the remote party in reply to the re-INVITE from the SPA to put the call on hold must have the SDP indicate a sendrecv or recvonly attribute and the remote destination address and port must not be 0

3. SAS Notes:

• Either or both of lines 1 and 2 can be configured as an SAS server.

• Each server can maintain up to 5 simultaneous calls. If the second line on the SPA is disabled, then the SAS line can maintain up to 10 simultaneous calls. Further incoming calls will receive a busy signal (SIP 486 Response).

• The streaming audio source must be off-hook for the streaming to occur. Otherwise incoming calls will get a error response (SIP 503 Response). The SAS line will not ring for incoming calls even if the attached equipment is on-hook

• If no calls are in session, battery is removed from tip-and-ring of the FXS port. Some audio source devices have an LED to indicate the battery status. This can be used as a visual indication whether any audio streaming is in progress.

• IVR can still be used on an SAS line, but the user needs to follow some simple steps: a) Connect a phone to the port and make sure the phone is on-hook, b) power on the SPA and c) pick up handset and press * * * * to invoke IVR in the usual way. The idea behind this is that if the SPA boots up and finds that the SAS line is on-hook, it will not remove battery from the line so that IVR may be used. But if the SPA boots up and finds that the SAS line is off-hook, it will remove battery from the line since no audio session is in progress.

• Set up the Proxy and Subscriber Information for the SAS Line as you normally would with a regular user account.

• Call Forwarding, Call Screening, Call Blocking, DND, and Caller-ID Delivery features are not available on an SAS line.

3.5.7.2. Supplementary Services Enablement

The SPA provides native support of a large set of enhanced or supplementary services. All of these services are optional. The parameters listed in the following table are used to enable or disable a specific supplementary service. A supplementary service should be disabled if a) the user has not subscribed for it, or b) the Service Provider intends to support similar service using other means than relying on the SPA.

Parameter Name	Description	Type	Default
Call Waiting Serv	Enable Call Waiting Service	Bool	Yes
Block CID Serv	Enable Block Caller ID Service	Bool	Yes
Block ANC Serv	Enable Block Anonymous Calls Service	Bool	Yes
Dist Ring Serv	Enable Distinctive Ringing Service	Bool	Yes
Cfwd All Serv	Enable Call Forward All Service	Bool	Yes
Cfwd Busy Serv	Enable Call Forward Busy Service	Bool	Yes
Cfwd No Ans Serv	Enable Call Forward No Answer Service	Bool	Yes
Cfwd Sel Serv	Enable Call Forward Selective Service	Bool	Yes
Cfwd Last Serv	Enable Forward Last Call Service	Bool	Yes
Block Last Serv	Enable Block Last Call Service	Bool	Yes
Accept Last Serv	Enable Accept Last Call Service	Bool	Yes
DND Serv	Enable Do Not Disturb Service	Bool	Yes
CID_Serv	Enable Caller ID Service	Bool	Yes
CWCID Serv	Enable Call Waiting Caller ID Service	Bool	Yes
Call Return Serv	Enable Call Return Service	Bool	Yes
Call Back Serv	Enable Call Back Service	Bool	Yes
Three Way Call Serv[1]	Enable Three Way Calling Service	Bool	Yes
Three Way Conf Serv[1,2]	Enable Three Way Conference Service	Bool	Yes
Attn Transfer Serv[1,2]	Enable Attended Call Transfer Service	Bool	Yes
Unattn Transfer Serv	Enable Unattended (Blind) Call Transfer Service	Bool	Yes
MWI Serv[3]	Enable MWI Service	Bool	Yes
VMWI Serv	Enable VMWI Service (FSK)	Bool	Yes
Speed Dial Serv	Enable Speed Dial Service	Bool	Yes
Secure Call Serv	Enable Secure Call Service	Bool	Yes
Referral Serv	Enable Referral Service. See <Referral Services Codes> for more details	Bool	Yes
Feature Dial Serv	Enable Feature Dial Service. See <Feature Dial Services Codes> for more details	Bool	Yes

Notes:

1. Three Way Calling is required for Three Way Conference and Attended Transfer.
2. Three Way Conference is required for Attended Transfer.
3. MWI is available only if a Voice Mail Service is set-up in the deployment.

3.5.7.3. Audio Settings

Parameter Name	Description	Type	Default
Preferred Codec	Select a preferred codec for all calls. However, the actual codec used in a call still depends on the outcome of the codec negotiation protocol.G711u, G711a, G726-16, G726-24, G726-32, G726-40, G729a, G723	Choice	G711u
Use Pref Codec Only	Only use the preferred codec for all calls. The call will fail if the far end does not support this codec.	Bool	No
LBR Codec Enable	*** This parameter has been removed. ***		
Silence Supp Enable	Enable silence suppression so that silent audio frames are not transmitted	Bool	No
Echo Canc Enable	Enable the use of echo canceller	Bool	Yes
Echo Canc Adapt Enable	Enable echo canceller to adapt	Bool	Yes
Echo Supp Enable	Enable the use of echo suppressor. If <Echo Canc Enable> is “no”, this parameter is ignored	Bool	Yes
G729a Enable[1]	Enable the use of G729a codec at 8 kbps.	Bool	Yes
G723 Enable[1]	Enable the use of G723 codec at 6.3 kbps	Bool	Yes
G726-16 Enable[1]	Enable the use of G726 codec at 16 kbps	Bool	Yes
G726-24 Enable[1]	Enable the use of G726 codec at 24 kbps	Bool	Yes
G726-32 Enable[1]	Enable the use of G726 codec at 32 kbps	Bool	Yes
G726-40 Enable[1]	Enable the use of G726 codec at 40 kbps	Bool	Yes
FAX Passthru Enable	*** This parameter has been removed. ***	Bool	Yes
FAX CED Detect Enable	Enable detection of FAX tone.	Bool	Yes
FAX CNG Detect Enable		Bool	Yes
FAX Passthru Codec	Codec to use for fax passthru	{G711u, G711a}	G711u
FAX Codec Symmetric	Force unit to use symmetric codec during FAX passthru	Bool	Yes
FAX Passthru Method	Choices: None / NSE / ReINVITE	Choice	NSE
FAX Process NSE		Bool	Yes
DTMF Tx Method	Method to transmit DTMF signals to the far end: Inband = Send DTMF using the audio path; INFO = Use the SIP INFO method, AVT = Send DTMF as AVT events; Auto = Use Inband or AVT based on outcome of codec negotiation	{InBand, AVT, INFO Auto}	Auto
Hook Flash Tx Method	Select the method to signal Hook Flash events: • None: do not signal hook flash events • AVT: use RFC2833 AVT (event=16) • INFO: use SIP INFO method with the single line “signal = hf” in the message body. The MIME type for this message body is taken from the <Hook Flash MIME Type> paramter	Choice	None
Release Unused Codec		Bool	Yes

Notes:

1. A codec resource is considered as allocated if it has been included in the SDP codec list of an active call, even though it eventually may not be the one chosen for the connection. So, if the G.729a

codec is enabled and included in the codec list, that resource is tied up until the end of the call whether or not the call actually uses G.729a. If the G729a resource is already allocated and since only one G.729a resource is allowed per SPA, no other low-bit-rate codec may be allocated for subsequent calls; the only choices are G711a and G711u. On the other hand, two G.723.1/G.726 resources are available per SPA. Therefore it is important to disable the use of G.729a in order to guarantee the support of 2 simultaneous G.723/G.726 codec.

3.5.7.4. Dial Plan

Parameter Name	Description	Type	Default
Dial Plan	Per-line dial plan script	DialPlanScript	See below
Enable IP Dialing	Enable IP Dialing	Bool	no

See the previous section for explanation of Dial Plan Script syntax.

Default Dial Plan script for each line:

“(*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxx|xxxxxxxxxxxx.)”

Explanation of Default Dial Plan:

Dial Plan Entry	Functionality
*xx	Allow arbitrary 2 digit star code
[3469]11	Allow x11 sequences
0	Operator
00	Int’l Operator
[2-9]xxxxxx	US "local" number
1xxx[2-9]xxxxxx	US 1 + 10-digit long distance number
xxxxxxxxxxxx.	Everything else (Int’l long distance, FWD, ...)

Note: If IP dialing is enabled, one can dial [user-id@]a.b.c.d[:port], where ‘@’, ‘.’, and ‘:’ are dialed by entering “*”, user-id must be numeric (like a phone number) and a, b, c, d must be between 0 and 255, and port must be larger than 255. If port is not given, 5060 is used. Port and User-Id are optional. If the user-id portion matches a pattern in the dial plan, then it is interpreted as a regular phone number according to the dial plan. The INVITE message, however, is still sent to the outbound proxy if it is enabled.

3.5.7.5. Polarity Settings

Parameter Name	Description	Type	Default
Idle Polarity	Polarity before call connected	{Forward,Reverse}	Forward
Caller Conn Polarity	Polarity after outbound call connected	{Forward,Reverse}	Reverse
Callee Conn Polarity	Polarity after inbound call connected	{Forward,Reverse}	Reverse

Notes:

3.5.8. User 1 and User 2 Parameters

User 1/2 refers to the subscriber of Line 1/2. When a call is made from Line 1/2, SPA shall use the user and line settings for that Line; there is no user login support in SPA v1.0. Per user parameter tags must be appended with [1] or [2] (corresponding to line 1 or 2) in the configuration profile. It is omitted below for readability.

3.5.8.1. Call Forward And Selective Call Forward/Blocking Settings

Parameter Name	Description	Type	Default
Cfwd All Dest	Forward number for Call Forward All Service	Phone	
Cfwd Busy Dest	Forward number for Call Forward Busy Service	Phone	
Cfwd No Ans Dest	Forward number for Call Forward No Answer Service	Phone	
Cfwd No Ans Delay	Delay in sec before Call Forward No Answer triggers	Uns8	20
Cfwd Sel1 Caller	Caller number pattern to trigger Call Forward Selective 1	PhTmplt	
Cfwd Sel2 Caller	Caller number pattern to trigger Call Forward Selective 2	PhTmplt	
Cfwd Sel3 Caller	Caller number pattern to trigger Call Forward Selective 3	PhTmplt	
Cfwd Sel4 Caller	Caller number pattern to trigger Call Forward Selective 4	PhTmplt	
Cfwd Sel5 Caller	Caller number pattern to trigger Call Forward Selective 5	PhTmplt	
Cfwd Sel6 Caller	Caller number pattern to trigger Call Forward Selective 6	PhTmplt	
Cfwd Sel7 Caller	Caller number pattern to trigger Call Forward Selective 7	PhTmplt	
Cfwd Sel8 Caller	Caller number pattern to trigger Call Forward Selective 8	PhTmplt	
Cfwd Sel1 Dest	Forward number for Call Forward Selective 1	Phone	
Cfwd Sel2 Dest	Forward number for Call Forward Selective 2	Phone	
Cfwd Sel3 Dest	Forward number for Call Forward Selective 3	Phone	
Cfwd Sel4 Dest	Forward number for Call Forward Selective 4	Phone	
Cfwd Sel5 Dest	Forward number for Call Forward Selective 5	Phone	
Cfwd Sel6 Dest	Forward number for Call Forward Selective 6	Phone	
Cfwd Sel7 Dest	Forward number for Call Forward Selective 7	Phone	
Cfwd Sel8 Dest	Forward number for Call Forward Selective 8	Phone	
Block Last Caller	ID of caller blocked via the “Block Last Caller” service	Phone	
Accept Last Caller	ID of caller accepted via the “Accept Last Caller” service	Phone	
Cfwd Last Caller	The Caller number that is actively forwarded to <Cfwd Last Dest> by using the Call Forward Last activation code	Phone	
Cfwd Last Dest	Forward number for the <Cfwd Last Caller>	Phone	

3.5.8.2. Speed Dial Settings

Parameter Name	Description	Type	Default
Speed Dial 2	Target phone number (or URL) assigned to speed dial “2”	Phone	
Speed Dial 3	Target phone number (or URL) assigned to speed dial “3”	Phone	
Speed Dial 4	Target phone number (or URL) assigned to speed dial “4”	Phone	
Speed Dial 5	Target phone number (or URL) assigned to speed dial “5”	Phone	
Speed Dial 6	Target phone number (or URL) assigned to speed dial “6”	Phone	
Speed Dial 7	Target phone number (or URL) assigned to speed dial “7”	Phone	
Speed Dial 8	Target phone number (or URL) assigned to speed dial “8”	Phone	
Speed Dial 9	Target phone number (or URL) assigned to speed dial “9”	Phone	

3.5.8.3. Supplementary Service Settings

Parameter Name	Description	Type	Default
CW Setting	Call Waiting on/off for all calls	Bool	Yes
Block CID Setting	Block Caller ID on/off for all calls	Bool	No
Block ANC Setting	Block Anonymous Calls on or off	Bool	No
DND Setting	DND on or off	Bool	No
CID Setting	Caller ID Generation on or off	Bool	Yes
CWCID Setting	Call Waiting Caller ID Generation on or off	Bool	Yes
Dist Ring Setting	Distinctive Ring on or off	Bool	Yes
Secure Call Setting	If yes, all outbound calls are secure calls by default	Bool	No

3.5.8.4. Distinctive Ring and Ring Settings

Parameter Name	Description	Type	Default
Ring 1 Caller	Caller number pattern to play Distinctive Ring/CWT 1	PhTmplt	
Ring 2 Caller	Caller number pattern to play Distinctive Ring/CWT 2	PhTmplt	
Ring 3 Caller	Caller number pattern to play Distinctive Ring/CWT 3	PhTmplt	
Ring 4 Caller	Caller number pattern to play Distinctive Ring/CWT 4	PhTmplt	
Ring 5 Caller	Caller number pattern to play Distinctive Ring/CWT 5	PhTmplt	
Ring 6 Caller	Caller number pattern to play Distinctive Ring/CWT 6	PhTmplt	
Ring 7 Caller	Caller number pattern to play Distinctive Ring/CWT 7	PhTmplt	
Ring 8 Caller	Caller number pattern to play Distinctive Ring/CWT 8	PhTmplt	
Default Ring	Default ringing pattern, 1 – 8, for all callers	{1,2,3,4,5,6,7,8}	1
Default CWT	Default CWT pattern, 1 – 8, for all callers	{1,2,3,4,5,6,7,8}	1
Hold Reminder Ring	Ring pattern for reminder of a holding call when the phone is on-hook	{1,2,3,4,5,6,7,8, None}	None
Call Back Ring	Ring pattern for call back notification	{1,2,3,4,5,6,7,8}	None
Cfwd Ring Splash Len[2]	Duration of ring splash when a call is forwarded (0 – 10.0s)	Time3	0
Cblk Ring Splash Len[2]	Duration of ring splash when a call is blocked (0 – 10.0s)	Time3	0
VMWI Ring Splash Len	Duration of ring splash when new messages arrive before the VMWI signal is applied (0 – 10.0s)	Time3	.5
VMWI Ring Policy	The parameter controls when a ring splash is played when a the VM server sends a SIP NOTIFY message to the SPA indicating the status of the subscriber's mail box. 3 settings are available: New VM Available – ring as long as there is 1 or more unread voice mail New VM Becomes Available – ring when the number of unread voice mail changes from 0 to non-zero New VM Arrives – ring when the number of unread voice mail increases	Choice	New VM Available
Ring On No New VM	If enabled, the SPA will play a ring splash when the VM server sends SIP NOTIFY message to the SPA indicating that there are no more unread voice mails. Some equipment requires a short ring to precede the	Bool	No

	FSK signal to turn off VMWI lamp		

Notes:
1. Caller number patterns are matched from Ring 1 to Ring 8. The first match (not the closest match) will be used for alerting the subscriber.
2. Feature not yet available.

3.5.9. Regional Parameters

3.5.9.1. Call Progress Tones

Parameter Name	Description	Type	Default
Dial Tone[1]	Played when prompting the user to enter a phone number	ToneScript	350@-19,440@-19;10(*/0/1+2)
Second Dial Tone	An alternative to <Dial Tone> when user tries to dial a 3-way call	ToneScript	420@-19,520@-19;10(*/0/1+2)
Outside Dial Tone[1]	An alternative to <Dial Tone> usually used to prompt the user to enter an external phone number (versus an internal extension). This is triggered by a "," character encountered in the dial plan.	ToneScript	420@-16;10(*/0/1)
Prompt Tone[1]	Played when prompting the user to enter a call forward phone number	ToneScript	520@-19,620@-19;10(*/0/1+2)
Busy Tone	Played when a 486 RSC is received for an outbound call	ToneScript	480@-19,620@-19;10(.5/.5/1+2)
Reorder Tone[1,2]	Played when an outbound call has failed or after the far end hangs up during an established call	ToneScript	480@-19,620@-19;10(.25/.25/1+2)
Off Hook Warning Tone[2]	Played when the subscriber does not place the handset on the cradle properly	ToneScript	480@-10,620@0;10(.125/.125/1+2)
Ring Back Tone	Played for an outbound call when the far end is ringing	ToneScript	440@-19,480@-19;*(2/4/1+2)
Confirm Tone	This should be a brief tone to notify the user that the last input value has been accepted.	ToneScript	600@-16;1(.25/.25/1)"
SIT1 Tone	An alternative to <Reorder Tone> played when an error occurs while making an outbound call. The RSC to trigger this tone is configurable (see Section ???)	ToneScript	985@-16,1428@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)
SIT2 Tone	See <SIT1 Tone>	ToneScript	914@-16,1371@-16,1777@-16;20(.274/0/1,.274/0/2,.380/0/3,0/4/0)
SIT3 Tone	See <SIT1 Tone>	ToneScript	914@-16,1371@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)
SIT4 Tone	See <SIT 1 Tone>	ToneScript	985@-16,1371@-16,1777@-16;20(.380/0/1,.274/0/2,.380/0/3,0/4/0)

MWI Dial Tone[1]	This tone is played instead of <Dial Tone> when there are unheard messages in the subscriber's mail box	ToneScript	350@-19,440@-19;2(.1/.1/1+2);10(*/0/1+2)
Cfwd Dial Tone	Special dial tone played when call forward all is activated	ToneScript	350@-19,440@-19;2(.2/.2/1+2);10(*/0/1+2)
Holding Tone	Indicate to the local user that the far end has placed the call on hold	ToneScript	600@-16;*(.1/.1/1,.1/.1/1,.1/9.5/1)
Conference Tone	Plays to all parties when a 3-way conference is in progress	ToneScript	350@-16;30(.1/.1/1,.1/9.7/1)
Secure Call Indication Tone	This tone is played when a call is successfully switched to secure mode. It should be played only for a short while (< 30s) and at a reduced level (< -19 dBm) so that it will not interfere with the conversation.	ToneScript	397@-19,507@-19;15(0/2/0,.2/.1/1,.1/2.1/2)

Notes:
1. Reorder Tone is played automatically when <Dial Tone> or any of its alternatives times out
2. Off Hook Warning Tone is played when Reorder Tone times out

3.5.9.2. Ring and CWT Cadence

Parameter Name	Description	Type	Default
Ring1 Cadence	Cadence script for distinctive ring 1	CadScript	60(2/4)"
Ring2 Cadence	Cadence script for distinctive ring 2	CadScript	60(.3/.2, 1/.2,.3/4)"
Ring3 Cadence	Cadence script for distinctive ring 3	CadScript	60(.8/.4,.8/4)
Ring4 Cadence	Cadence script for distinctive ring 4	CadScript	60(.4/.2,.3/.2,.8/4)
Ring5 Cadence	Cadence script for distinctive ring 5	CadScript	60(.4/.2,.3/.2,.8/4)
Ring6 Cadence	Cadence script for distinctive ring 6	CadScript	60(.4/.2,.3/.2,.8/4)
Ring7 Cadence	Cadence script for distinctive ring 7	CadScript	60(.4/.2,.3/.2,.8/4)
Ring8 Cadence	Cadence script for distinctive ring 8	CadScript	60(.4/.2,.3/.2,.8/4)
CWT 1 Cadence	Cadence script for distinctive CWT 1	CadScript	30(.3/9.7)
CWT2 Cadence	Cadence script for distinctive CWT 2	CadScript	30(.1/.1, .1/9.7)"
CWT3 Cadence	Cadence script for distinctive CWT 3	CadScript	30(.1/.1, .1/.1, .1/9.5)
CWT4 Cadence	Cadence script for distinctive CWT 4	CadScript	30(.1/.1, .3/.1, .1/9.3)
CWT5 Cadence	Cadence script for distinctive CWT 5	CadScript	30(.3/.1,.1/.1,.3/9.1)
CWT6 Cadence	Cadence script for distinctive CWT 6	CadScript	30(.1/.1, .3/.1, .1/9.3)
CWT7 Cadence	Cadence script for distinctive CWT 7	CadScript	30(.1/.1, .3/.1, .1/9.3)
CWT8 Cadence	Cadence script for distinctive CWT 8	CadScript	2.3(..3/2)
Ring1 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 1 for the inbound call	Str31	Bellcore-r1
Ring2 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 2 for the inbound call	Str31	Bellcore-r2
Ring3 Name	Name in an INVITE's Alert-Info Header to pick	Str31	Bellcore-r3

	distinctive ring/CWT 3 for the inbound call		
Ring4 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 4 for the inbound call	Str31	Bellcore-r4
Ring5 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 5 for the inbound call	Str31	Bellcore-r5
Ring6 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 6 for the inbound call	Str31	Bellcore-r6
Ring7 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 7 for the inbound call	Str31	Bellcore-r7
Ring8 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 8 for the inbound call	Str31	Bellcore-r8
Ring Waveform	Waveform for the ringing signal	{Sinusoid, Trapezoid}	Sinusoid
Ring Frequency	Frequency of the ringing signal. Valid values are 10 – 100 (Hz)	Uns8	25
Ring Voltage	Ringing voltage. 60-90 (V)	Uns8	70
CWT Frequency	Frequency script of the call waiting tone. All distinctive CWT is based on this tone.	FreqScript	440@-10

Notes:

3.5.9.3. Control Timer Values (sec)

Parameter Name	Description	Type	Default
Hook Flash Timer Min	Minimum on-hook time before off-hook to qualify as hook-flash. Less than this the on-hook event is ignored. Range: 0.1 – 0.4 sec	Time3	0.1
Hook Flash Timer Max	Maximum on-hook time before off-hook to qualify as hook-flash. More than this the on-hook event is treated as on-hook (no hook-flash event). Range: 0.4 – 1.6 sec	Time3	0.9
Callee On Hook Delay	The phone must be on-hook for at this time in sec before the SPA will tear down the current inbound call. It does not apply to outbound calls. Range: 0 – 255 sec	Time0	0
Reorder Delay	Delay after far end hangs up before reorder tone is played. 0 = plays immediately, inf = never plays. Range: 0 – 255 sec	Time0	5
Call Back Expires	Expiration time in sec of a call back activation. Ragne: 0 – 65535 sec	Time0	1800
Call Back Retry Intvl	Call back retry interval in sec. Range: 0 – 255 sec	Time0	30
Call Back Delay	Delay after receiving the first SIP 18x response before declaring the remote end is ringing. If a busy response is received during this time, the SPA still considers the call as failed and keeps on retrying.	Time3	0.5
VMWI Refresh Intvl	Interval between VMWI refresh to the CPE	Time3	0.5
Interdigit Long Timer[2]	Long timeout between entering digits when dialing. Range: 0 – 64 sec	Time0	10
Interdigit Short Timer[2]	Short timeout between entering digits when dialing. Range: 0 – 64 sec	Time0	3
CPC Delay[3,4]	Delay in seconds after caller hangs up when		2

	the SPA will start removing the tip-and-ring voltage to the attached equipment of the called party. Range= 0 to 255(s) Resolution = 1 (s)		
CPC Duration[3,4]	Duration in seconds for which the tip-to-ring voltage is removed after the caller hangs up. After that tip-to-ring voltage is restored and dial tone will apply if the attached equipment is still off hook. CPC is disabled if this value is set to 0. Range= 0 to 1.000 (s) Resolution = 0.001 (s)		0 (CPC disabled)

Notes:
1. The Call Progress Tones and DTMF playback level are not affected by the <FXS Port Output Gain>.
2. The interdigit timer values are used as defaults when dialing. The Interdigit_Long_Timer is used after any one digit, if all valid matching sequences in the dial plan are incomplete as dialed. The Interdigit_Short_Timer is used after any one digit, if at least one matching sequence is complete as dialed, but more dialed digits would match other as yet incomplete sequences.
3. SPA has had polarity reversal feature since release 1.0 which can be applied to both the caller and the callee end. This feature is generally used for answer supervision on the caller side to signal to the attached equipment when the call has been connected (remote end has answered) or disconnected (remote end has hung up). This feature should be disabled for the called party (ie by using the same polarity for connected and idle state) and the CPC feature should be used instead.
4. Without CPC enabled, reorder tone will is played after a configurable delay. If CPC is enabled, dial tone will be played when tip-to-ring voltage is restored.

3.5.9.4. Vertical Service Code Assignment

Parameter Name	Description	Type	Default
Call Return Code	Call the last caller.	ActCode	*69
Blind Transfer Code	Blind transfer current call to the target specified after the activation code	ActCode	*98
Cfwd All Act Code	Forward all calls to the target specified after the activation code	ActCode	*72
Cfwd All Deact Code	Cancel call forward all	ActCode	*73
Cfwd Busy Act Code	Forward busy calls to the target specified after the activation code	ActCode	*90
Cfwd Busy Deact Code	Cancel call forward busy	ActCode	*91
Cfwd No Ans Act Code	Forward no-answer calls to the target specified after the activation code	ActCode	*92
Cfwd No Ans Deact Code	Cancel call forward no-answer	ActCode	*93
Cfwd Last Act Code	Forward the last inbound or outbound calls to the target specified after the activation code	ActCode	*63
Cfwd Last Deact Code	Cancel call forward last	ActCode	*83
Block Last Act Code	Block the last inbound call	ActCode	*60
Block Last Deact Code	Cancel blocking of the last inbound call	ActCode	*80
Accept Last Act Code	Accept the last outbound call. Let it ring through when DND or Call Forward All is in effect	ActCode	*64

Accept Last Deact Code	Cancel Accept Last	ActCode	*84
Call Back Act Code	Callback when the last outbound call is not busy	ActCode	*66
Call Back Deact Code	Cancel callback	ActCode	*86
CW_Act_Code	Enable Call Waiting on all calls	ActCode	*56
CW_Deact_Code	Disable Call Waiting on all calls	ActCode	*57
CW_Per_Call_Act_Code	Enable Call Waiting for the next call	ActCode	*71
CW_Per_Call_Deact_Code	Disable Call Waiting for the next call	ActCode	*70
Block_CID_Act_Code	Block CID on all outbound calls	ActCode	*67
Block_CID_Deact_Code	Unblock CID on all outbound calls	ActCode	*66
Block_CID_Per_Call_Act_Code	Block CID on the next outbound call	ActCode	*81
Blcok_CID_Per_Call_Deact_Code	Unblock CID on the next inbound call	ActCode	*82
Block_ANC_Act_Code	Block all anonymous calls	ActCode	*77
Block_ANC_Deact_Code	Unblock all anonymous calls	ActCode	*87
DND_Act_Code	Enable Do Not Disturb	ActCode	*78
DND_Deact_Code	Disable Do Not Disturb	ActCode	*79
CID_Act_Code	Enable Caller-ID Generation	ActCode	*65
CID_Deact_Code	Disable Call-ID Generation	ActCode	*85
CWCID_Act_Code	Enable Call Waiting Caller-ID generation	ActCode	*25
CWCID_Deact_Code	Disable Call Waiting Caller-ID generation	ActCode	*45
Dist_Ring_Act_Code	Enable Distinctive Ringing	ActCode	*61
Dist_Ring_Deact_Code	Disable Distinctive Ringing	ActCode	*81
Speed Dial Act Code	Assign a speed dial number	ActCode	*74
Secure All Call Act Code	Make all outbound calls secure	ActCode	*16
Secure No Call Act Code	Make all outbound calls not secure	ActCode	*17
Secure One Call Act Code	Make the next outbound call secure. This operation is redundant if all outbound calls are secure by default.	ActCode	*18
Secure One Call Deact Code	Make the next outbound call not secure. This operation is redundant if all outbound calls are not secure by default.	ActCode	*19
Referral Services Codes	One or more *code can be configured into this parameter, such as *98, or *97|*98|*123, etc. Max total length is 79 chars. This parameter applies when the user places the current call on hold (by Hook Flash) and is listening to 2nd dial tone. Each *code (and the following valid target number according to current dial plan) entered on the 2nd dial-tone triggers the SPA to perform a blind transfer to a target number that is prepended by the service *code. For example, after the user dials *98, the SPA plays a special dial tone called the "Prompt Tone" while waiting for the user the enter a target number (which is checked according to dial plan as in normal dialing). When a complete number is entered, the SPA sends a blind REFER to the holding party with the Refer-To target equals to *98<target_number>. This feature allows the SPA to "hand off"	Str79	

	a call to an application server to perform further processing, such as call park. Notes: - The *codes should not conflict with any of the other vertical service codes internally processed by the SPA. You can empty the corresponding *code that you do not want to SPA to process.		
Feature Dial Services Codes	One or more *code can be configured into this parameter, such as *72, or *72|*74|*67|*82, etc. Max total length is 79 chars. This parameter applies when the user has a dial tone (1st or 2nd dial tone). Enter *code (and the following target number according to current dial plan) entered at the dial tone triggers the SPA to call the target number prepended by the *code. For example, after user dials *72, the SPA plays a prompt tone awaiting the user to enter a valid target number. When a complete number is entered, the SPA sends a INVITE to *72<target_number> as in a normal call. This feature allows the proxy to process features like call forward (*72) or BLock Caller ID (*67). Notes: - The *codes should not conflict with any of the other vertical service codes internally processed by the SPA. You can empty the corresponding *code that you do not want to SPA to process. - You can add a paramter to each *code in "Features Dial Services Codes" to indicate what tone to play after the *code is entered, such as *72\`c\`|*67\`p\`. Below are a list of allowed tone parameters (note the use of back quotes surrounding the parmeter w/o spaces) \`c\` = <Cfwd Dial Tone> \`d\` = <Dial Tone> \`m\` = <MWI Dial Tone> \`o\` = <Outside Dial Tone> \`p\` = <Prompt Dial Tone> \`s\` = <Second Dial Tone> \`x\` = No tones are place, x is any digit not used above If no tone parameter is specified, the SPA plays Prompt tone by default. - If the *code is not to be followed by a phone number, such as *73 to cancel call forwarding, do not include it in this parameter. In that case, simple add that	Str79	

	*code in the dial plan and the SPA will send INVITE *73@..... as usual when user dials *73.		

Notes:
1. These codes automatically appended to the dial-plan. So no need to include them in dial-plan (although no harm to do so either).

3.5.9.5. Outbound Call Codec Selection Codes:

Parameter Name	Description	Type	Default
Prefer G711u Code	Dialing code will make this codec the preferred codec for the associated call.	ActCode	*017110
Force G711u Code	Dialing code will make this codec the only codec that can be used for the associated call.	ActCode	*027110
Prefer G711a Code	Dialing code will make this codec the preferred codec for the associated call.	ActCode	*017111
Force G711a Code	Dialing code will make this codec the only codec that can be used for the associated call.	ActCode	*027111
Prefer G723 Code	Dialing code will make this codec the preferred codec for the associated call.	ActCode	*01723
Force G723 Code	Dialing code will make this codec the only codec that can be used for the associated call.	ActCode	*02723
Prefer G726r16 Code	Dialing code will make this codec the preferred codec for the associated call.	ActCode	*0172616
Force G726r16 Code	Dialing code will make this codec the only codec that can be used for the associated call.	ActCode	*0272616
Prefer G726r24 Code	Dialing code will make this codec the preferred codec for the associated call.	ActCode	*0172624
Force G726r24 Code	Dialing code will make this codec the only codec that can be used for the associated call.	ActCode	*0272624
Prefer G726r32 Code	Dialing code will make this codec the preferred codec for the associated call.	ActCode	*0172632
Force G726r32 Code	Dialing code will make this codec the only codec that can be used for the associated call.	ActCode	*0272632
Prefer G726r40 Code	Dialing code will make this codec the preferred codec for the associated call.	ActCode	*0172640
Force G726r40 Code	Dialing code will make this codec the only codec that can be used for the associated call.	ActCode	*0272640
Prefer G729a Code	Dialing code will make this codec the preferred codec for the associated call.	ActCode	*01729
Force G729a Code	Dialing code will make this codec the only codec that can be used for the associated call.	ActCode	*02729

Notes:
1. These codes automatically appended to the dial-plan. So no need to include them in dial-plan (although no harm to do so either).

3.5.9.6. Secure Call Implementation:

A secure call is established in two stages. The first stage is no different form a normal call setup. Right after the call is established in the normal way with both sides ready to stream RTP packets, the second stage starts where the two parties exchange information to determine if the current call can switch over to the secure mode. The information is transported by base64 encoding and embedding in the message body of SIP INFO requests and responses with a proprietary format. If the second stage is successful, the SPA will play a special “Secure Call Indication Tone” for short while to indicate to both parties that the call is secured and that RTP traffic in both directions are encrypted. If the user has a CIDCW capable phone and CIDCW service is enabled, then the CID will be updated with the information extracted from the Mini-Certificate received from the other end. The Name field of this CID will be prepended with a ‘$’ symbol.

The second stage in setting up a secure all can be further divided into two steps. Step 1 the caller sends a “Caller Hello” message (base64 encoded and embedded in the message body of a SIP INFO request) to the called party with the following information:

- Message ID (4B)
- Version and flags (4B)
- SSRC of the encrypted stream (4B)
- Mini-Certificate (252B)

Upon receiving the Caller Hello, the callee responds with a Callee Hello message (base64 encoded and embedded in the message body of a SIP response to the caller’s INFO request) with similar information, if the Caller Hello message is valid. The caller then examines the Callee Hello and proceeds to step 2 if the message is valid. In step 2 the caller sends the “Caller Final” message to the callee with the following information:

- Message ID (4B)
- Encrypted Master Key (16B or 128b)
- Encrypted Master Salt (16B or 128b)

With the master key and master salt encrypted with the public key from the callee’s mini-certificate. The master key and master salt are used by both ends for the derivation of session keys for encrypting subsequent RTP packets. The callee then responds with a Callee Final message (which is an empty message).

A Mini-Certificate contains the following information:
- User Name (32B)
- User ID or Phone Number (16B)
- Expiration Date (12B)
- Public Key (512b or 64B)
- Signature (1024b or 512B)

The signing agent is implicit and must be the same for all SPA’s that intended to communicate securely with each other. The public key of the signing agent is pre-configured into the SPA’s by the administrator and will be used by the SPA to verify the Mini-Certificate of its peer. The Mini-Certificate is valid if a) it has not expired, and b) its signature checks out.

User Interface
The SPA can be set up such that all outbound calls are secure calls by default, or not secure by default. If outbound calls are secure by default, user has the option to disable security when making the next call by dialing *19 before dialing the target number. If outbound calls are not secure by default, user has the option to make the next outbound call secure by dialing *18 before dialing the target number. On the other hand, user cannot force inbound calls to be secure or not secure; it is at the mercy of the caller whether he/she enables security or not for that call.

If the call successfully switches to the secure mode, both parties will hear the “Secure Call Indication Tone” for a short while and the CID will be updated with the Name and Number extracted from the Mini-Certificate sent by the other party, provided CIDCW service and equipment are available: the

CID Name in this case will have a '$' sign inserted at the beginning. The callee should check the name and number again to ensure the identity of the caller. The caller should also double check the name and number of the callee to make sure this is what he/she expects. Note that the SPA will not switch to secure mode if the callee's CID Number from its Mini-Certificate does not agree with the user-id used in making the outbound call: the caller's SPA will perform this check after receiving the callee's Mini-Certificate.

Service Provider Requirements

The SPA Mini-Certificate (MC) has a 512-bit public key used for establishing secure calls. The administrator must provision each subscriber of the secure call service with an MC and the corresponding 512-bit private key. The MC is signed with a 1024-bit private key of the service provider who acts as the CA of the MC. The 1024-bit public key of the CA signing the MC must also be provisioned to each subscriber. The CA public key is used by the SPA to verify the MC received from the other end. If the MC is invalid, the SPA will not switch to secure mode. The MC and the 1024-bit CA public key are concatenated and base64 encoded into the single parameter <Mini Certificate>. The 512-bit private key is base64 encoded into the <SRTP Private Key> parameter, which should be hidden from the SPA's web interface like a password.

Since the secure call establishment relies on exchange of information embedded in message bodies of SIP INFO requests/responses, the service provider must maker sure that their infrastructure will allow the SIP INFO messages to pass through with the message body unmodified.

Sipura provides a configuration tool called gen_mc for the generation of MC and private keys with the following syntax:

gen_mc <ca-key> <user-name> <user-id> <expire-date>

Where:
- ca-key is a text file with the base64 encoded 1024-bit CA private/public key pairs for signing/verifying the MC, such as
9CC9aYU1X5lJuU+EBZmi3AmcqE9U1LxEOGwopaGyGOh3VyhKgi6JaVtQZt87PiJINKW8XQj3B9Qq
e3VgYxWCQNa335YCnDsenASeBxuMIEaBCYd1l1fVEodJZOGwXwfAde0MhcbD0kj7LVlzcsTyk2TZ
YTccnZ75TuTjj13qvYs=
5nEtOrkCa84/mEwl3D9tSvVLyliwQ+u/Hd+C8u5SNk7hsAUZaA9TqH8Iw0J/IqSrsf6scsmundY5j7Z5m
K5J9uBxSB8t8vamFGD0pF4zhNtbrVvIXKI9kmp4vph1C5jzO9gDfs3MF+zjyYrVUFdM+pXtDBxmM+f
GUfrpAuXb7/k=
- user-name is the name of the subscriber, such as "Joe Smith". Maximum length is 32 characters
- user-id is the user-id of the subscriber and must be exactly the same as the user-id used in the INVITE when making the call, such as "14083331234". Maximum length is 16 characters.
- expire-date is the expiration date of the MC, such as "00:00:00 1/1/34" (34=2034). Internally the date is encoded as a fixed 12B string: 000000010134

The tool generates the <Mini Certificate> and <SRTP Private Key> parameters that can be provisioned to the SPA.

For Example:
gen_mc ca_key "Joe Smith" 14085551234 "00:00:00 1/1/34"

Produces:
<Mini Certificate>
Sm9lIFNtaXRoAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAxNDA4NTU1MTIzNAAAAAAAMDAwM
DAwMDEwMTM00OvJakde2vVMF3Rw4pPXL7lAgIagMpbLSAG2+++YlSqt198Cp9rP/xMGFfoPmDK
Gx6JFtkQ5sxLcuwgxpxpxkeXvpZKlYlpsb28L4Rhg5qZA+Gqj1hDFCmG6dffZ9SJhxES767G0JIS+N8l
QBLr0AuemotknSjjjOy8c+1lTCd2t44Mh0vmwNg4fDck2YdmTMBR516xJt4/uQ/LJQlni2kwqlm7scDvll5

k232EvvvVtCK0AYa4eWd6fQOpiESCO9CC9aYU1X5lJuU+EBZmi3AmcqE9U1LxEOGwopaGyGOh3
VyhKgi6JaVtQZt87PiJINKW8XQj3B9Qqe3VgYxWCQNa335YCnDsenASeBxuMIEaBCYd1l1fVEodJZ
OGwXwfAde0MhcbD0kj7LVlzcsTyk2TZYTccnZ75TuTjj13qvYs=

<SRTP Private Key>
b/DWc96X4YQraCnYzl5en1CIUhVQQqrvcr6Qd/8R52IEvJjOw/e+Klm4XiiFEPaKmU8UbooxKG36SEd
Kusp0AQ==

3.5.9.7. Miscellaneous Parameters

Parameter Name	Description	Type	Default
Set Local Date (mm/dd/yyyy)	Setting the local date; year is optional and can be 2-digit or 4-digit	Str10	
Local Time (HH/mm/ss)	Setting the local time; second is optional.	Str8	
Time Zone	Number of hours to add to GMT to form local time for caller-id generation. Choices: GMT-12:00, GMT-11:00,…, GMT, GMT+01:00, GMT+02:00, …, GMT+13:00	Choice	GMT-07:00
FXS Port Impedance	Electrical impedance of the FXS port.	{600, 900, 600+2.16uF, 900+2.16uF, 270+750||150nF, 220+820||120nF, 220+820||115nF, 370+620||310nF}	600
FXS Port Input Gain	Input Gain in dB. Valid values are 6.0 to –infinity. Up to 3 decimal places	dB	-3
FXS Port Output Gain	Similar to <FXS Port Input Gain> but apply to the output signal	dB	-3
DTMF Playback Level	Local DTMF playback level in dBm (up to 1 decimal place)	PwrLevel	-10.0
DTMF Playback Length	Local DTMF playback duration in ms	Time3	.1
Detect ABCD	Enable local detection of DTMF ABCD	Bool	Yes
Playback ABCD	Enable local playback of OOB DTMF ABCD	Bool	Yes
Caller ID Method	The following choices are available: • Bellcore (N.Amer,China): CID, CIDCW, and VMWI. FSK sent after 1st ring (same as ETSI FSK sent after 1st ring) (no polarity reversal or DTAS) • DTMF (Finland,Sweden): CID only. DTMF sent after polarity reversal (and no DTAS) and before 1st ring • DTMF (Denmark): CID only. DTMF sent after polarity reversal (and no DTAS) and before 1st ring • ETSI DTMF: CID only. DTMF sent after DTAS (and no polarity reversal) and before 1st ring • ETSI DTMF With PR: CID only. DTMF sent after polarity reversal and DTAS and before 1st ring • ETSI DTMF After Ring: CID only. DTMF	Choice	Bellcore

	sent after 1st ring (no polarity reversal or DTAS) • ETSI FSK: CID, CIDCW, and VMWI. FSK sent after DTAS (but no polarity reversal) and before 1st ring. Will wait for ACK from CPE after DTAS for CIDCW. • ETSI FSK With PR (UK): CID, CIDCW, and VMWI. FSK is sent after polarity reversal and DTAS and before 1st ring. Will wait for ACK from CPE after DTAS for CIDCW. Polarity reversal is applied only if equipment is on hook.		
FXS Port Power Limit	Options: 1, 2, 3, 4, 5, 6, 7, 8	Choice	3
Protect IVR Factory Reset			No

Notes:

1. It should be noted that the choice of CID method will affect the following features:

• On Hook Caller ID Associated with Ringing – This type of Caller ID is used for incoming calls when the attached phone is on hook. See figure below (a) – (c). All CID methods can be applied for this type of caller-id

• On Hook Caller ID Not Associated with Ringing – This feature is used for send VMWI signal to the phone to turn the message waiting light on and off (see Figure 1 (d) and (e)). This is available only for FSK-based caller-id methods: “Bellcore”, “ETSI FSK”, and “ETSI FSK With PR”

• Off Hook Caller ID – This is used to delivery caller-id on incoming calls when the attached phone is off hook. See figure below (f). This can be call waiting caller ID (CIDCW) or to notify the user that the far end party identity has changed or updated (such as due to a call transfer). This is only available if the caller-id method is one of “Bellcore”, “ETSI FSK”, or “ETSI FSK With PR”.

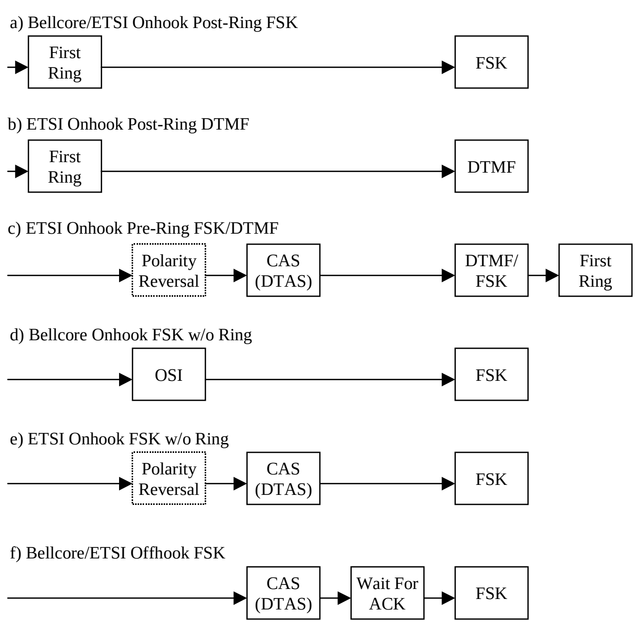


Figure: SPA Caller ID Delivery Architecture

3.6. Call Statistics Reporting

The following lists the statistics collected by the SPA during normal operation. These statistics are presented in the SPA web-page (under the "Info" tab). Line status is reported for each line (1 and 2). Each line maintains up to 2 calls: Call 1 and 2.

System Status	
Current Time	Current time and date. E.g., 10/3/2003 16:43:00
Elapsed Time	Total time elapsed since last reboot. E.g., 25 days and 18:12:36
Broadcast Pkts Sent	Total number of broadcast packets sent
Broadcast Pkts Recv	Total number of broadcast packets received
Broadcast Bytes Sent	Total number of broadcast bytes sent
Broadcast Bytes Recv	Total number of broadcast bytes received and processed
Broadcast Packets Dropped	Total number of broadcast packets received but not processed
Broadcast Bytes Dropped	Total number of broadcast bytes received but not processed
RTP Packets Sent	Total number of RTP packets sent (including redundant packets)
RTP Packets Received	Total number of RTP packets received (including redundant packets)
RTP Bytes Sent	Total number of RTP bytes sent
RTP Bytes Received	Total number of RTP bytes received
SIP Messages Sent	Total number of SIP messages sent (including retransmissions)
SIP Messages Received	Total number of SIP messages received (including retransmissions)
SIP Bytes Sent	Total number of bytes of SIP messages sent (including retransmissions)
SIP Bytes Received	Total number of bytes of SIP messages received (including retransmissions)
External IP	External IP address used for NAT mapping
Line 1/2 Status	
Hook State	State of the hook switch: On or Off
Registration State	Registration state of the line: Not Registered, Registered or Failed
Last Registration At	Local time of the last successful registration
Next Registration In	Number of seconds before the next registration renewal
Message Waiting	Indicate whether new voice mails available: Yes or No
Call Back Active	Indicate whether a call back request is in progress: Yes or No
Last Called Number	The last number called
Last Caller Number	The number of the last caller
Mapped SIP Port	NAT Mapped SIP Port
Call 1/2 Status	
State	State of the call: Idle, Dialing, Calling, Proceeding, Ringing, Answering, Connected, Hold, Holding, Resuming, or Reorder
Tone	Tone playing for this call: Dial, 2nd Dial, Outside Dial, Ring Back, Ring, Busy, Reorder, SIT1– 4, Call Waiting, Call Forward, Conference, Prompt, Confirmation, or Message-Waiting
Encoder	Encoder in use: G711u, G711a, G726-16/24/32/40, G729a, or G729ab
Decoder	Decoder in use: G711u, G711a, G726-16/24/32/40, G729a, or G729ab
FAX	Indicate whether FAX pass-through mode has been initiated: Yes or No
Type	Indicate the call type: Inbound or Outbound
Remote Hold	Indicate whether the remote end has placed the call on hold: Yes or No
Call Back	Indicate whether the call is triggered by a call back request: Yes or No
Peer Name	Name of the peer
Peer Phone	Phone number of the peer
Duration	Duration of the call in hr/min/sec format
Packets Sent	Number of RTP packets sent
Packets Recv	Number of RTP packets received
Bytes Sent	Number of RTP bytes sent
Bytes Recv	Number of RTP bytes received
Decode Latency	Decoder latency in milliseconds

Jitter	Receiver jitter in milliseconds
Round Trip Delay	Network round trip delay (ms); available if the peer supports RTCP
Packets Lost	Total number of packets lost
Packet Error	Number of RTP packets received that are invalid
Mapped RTP Port	NAT mapped RTP port

4. SPA-3000 Configuration

4.1. Overview

The SPA-3000 has 1 FXS and 1 FXO port. Each port is a RJ11 connector – the FXS Port is labeled "PHONE" and the FXO Port "LINE."

A standard analog telephone can be connected to the FXS/PHONE port to provide VoIP services just as with the SPA-1000 and SPA-2000. The FXO/LINE port can be connected to a standard PSTN line or other phone service – including another VoIP service. With the FXO/LINE port, the SPA-3000 can bridge a PSTN and a VoIP service. This functionality is referred to as a *Gateway*. We refer to the VoIP-To-PSTN calling function as a *PSTN Gateway*, and PSTN-To-VoIP calling function as a *VoIP Gateway*. We also define:

- VoIP Caller – one who calls the SPA-3000 via VoIP to obtain PSTN service.
- VoIP User – a VoIP Caller which has a user account (user-id and password) on the SPA-3000
- PSTN Caller – one who calls the SPA-3000 from the PSTN to obtain VoIP service.

Two VoIP services can be configured in the SPA-3000: one accessed from the FXS port and the other from the FXO port. In this document, the VoIP service that is accessed from the FXS/PHONE port is referred to as the **Line 1**, and the VoIP service that is accessed from the FXO/LINE port is referred to as the **PSTN Line**.

Notes:
- The term **PSTN line** (case sensitive), on the other hand, stands for the PSTN service connected to the FXO/LINE port.
- The notations [Line 1], [PSTN Line], etc., refer to the web page tabs appear on the SPA configuration web page. Each tab represents a logical group of configuration parameters.

The configuration of Line 1 is similar to Line 1 in the SPA-2000, with several additional options for PSTN-VoIP gateway configurations. Line 1 can be configured with a regular VoIP account and used in the same way as the Line 1 of the SPA-2000. A second VoIP account can be configured to support PSTN gateway calls exclusively. The options for controlling Line 1 and PSTN Line are configured under the [Line 1] and [PSTN Line] tabs on the SPA-3000 configuration web page respectively. Line 1 works almost independently of the PSTN Line. In fact, Line 1 can be disabled without affecting most of the operations on the PSTN Line. A different <SIP Port> parameter, however, should be assigned to Line 1 and PSTN Line. The same VoIP account may be used for both Line 1 and the PSTN Line as long as each line uses a different <SIP Port>.

The FXS/PHONE port can be electrically connected to either the FXO/LINE port or the SLIC inside the SPA-3000, by opening or closing an internal PHONE Port Relay (under the control of SPA firmware). Before power is applied to the unit, the relay is open and the phone is connected to the PSTN line.

After power is applied to the unit, the ACT LED will turn on and blink to indicate network activity (transmit or receive). The STATUS LED will also blink slowly to indicate DHCP discovery (if DHCP option is enabled in the SPA) and hardware initialization. If all goes well, the STATUS LED will turn off after 10-20s to indicate successful completion of hardware initialization. The ACT LED should remain mostly steady on with occasional blinks. Whenever you pick up the handset of the PHONE port phone, the STATUS LED should become steady on until the phone is off-hook again.

Following successful hardware initialization, the relay must be closed at some point for normal operation. However, there is a chance that the PHONE port phone may be using the PSTN line (via the FXO/LINE port) and closing the relay will therefore interrupt the call that is in progress (which would be highly undesirable especially if the call is an emergency). The SPA-3000 uses the following algorithm to determine when to close the relay:

```
Off-Hook FXO port
Close PHONE Port Relay
If (Loop Current is 0) {
  Done (since PSTN line is not connected or inactive)
}
Else {
  If (PHONE Port is on-hook) {
    Done (since it is not using the PSTN line)
  }
  Else {
    Open Phone Port Relay (since PHONE port phone is using the PSTN line)
    While (1) {
      Monitor FXO port Tip-To-Ring Voltage
      If (PSTN line is not in use) {
          Close Phone Port Relay
          Done
      }
    }
  }
}
```

Notes:
- A mechanical click sound can be heard whenever the relay is open or closed.
- The STATUS light will be steady on if the FXO port or the FXS port is off-hook.
- Once the PHONE Port Relay is closed, it will not be open again until the power is removed.
- To be able to invoke the IVR (voice configuration menu), the PHONE Port Relay must be closed.

4.2. SPA-3000 Voice Configuration Organization

The SPA-3000 can be thought of as having 4 logical voice interfaces, namely,

- FXS Interface
- FXO Interface
- VoIP1 Interface
- VoIP2 Interface

Figure 4.1 shows a block diagram of the voice interfaces and the group of configuration parameters for controlling their operations. It also shows the possible voice paths with each segment labeled by a number in parenthesis. We shall refer to the voice path of a call as a connected list of the path labels, such as (1) → (2) → (3) and (5) → (6) → (7). Please take a moment to familiarize yourself with this diagram. We will make reference to it very often as we describe the different configuration scenarios.

Like the SPA-2000, the SPA-3000 configuration parameters are organized into 8 groups, with each group accessed by clicking the corresponding tab on the top of the SPA web page. These 8 groups are:

1. System – Network parameters, DNS, NTP, Syslog, and Debug Servers
2. Provisioning – Profile rule, resync intervals and policy, and GPP
3. SIP – SIP stack control parameters for both VoIP interfaces

4. Regional – Call progress tones and cadences, ring cadences, * codes, international control for the FXS Interface. The tone and cadence parameters also apply to the VoIP2 and FXO interfaces
5. Line 1 – Audio, NAT, SIP, Network, Gateway, Supplementary Services, Polarity parameters for VoIP1 and FXS
6. PSTN Line – Audio, NAT, SIP, and Network parameters for VoIP2
7. User 1 – User options for VoIP1 and FXS
8. PSTN User – User options for VoIP2 and FXO

In addition, there is a 9th group under the Info tab which shows a number of read-only parameters and status information.

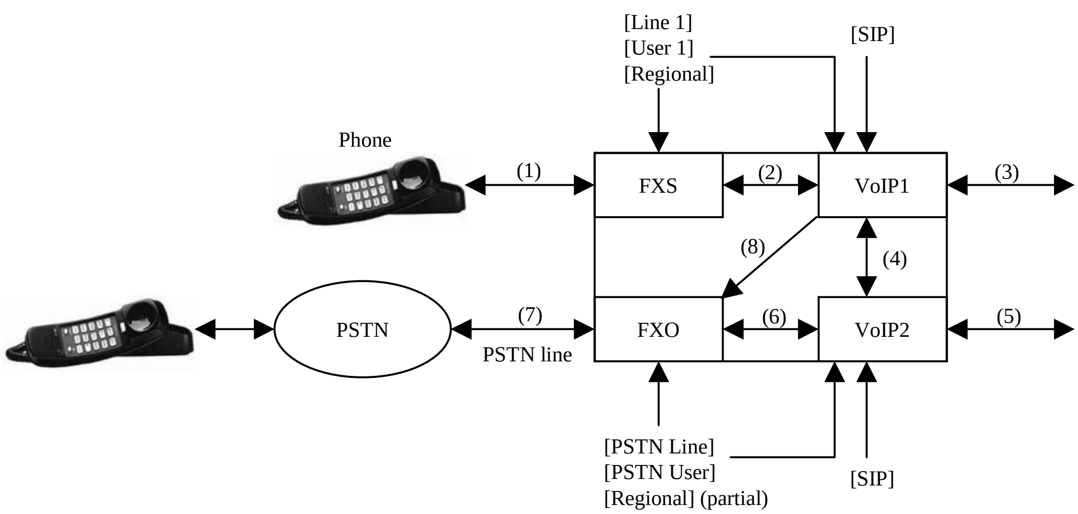


Figure 4.1: SPA-3000 Voice Paths and Configuration Architecture

4.2.1. FXS Interface

This is the interface to the FXS/PHONE Port through which the user accesses the Line 1 VoIP service. It controls how the SPA exchanges signals with the phone attached to FXS/PHONE port. It supplies power to the phone and senses its on/off hook state. This interface can be configured under the [Regiona], [Line 1], and [User 1] tabs.

Options that can be configured on the FXS Interface include:

- Polarity reversal signal generation
- CPC signal generation
- Ringer characteristics
- Call progress tones generation
- Transmit and receive gains
- DTMF playback level and timing
- Caller-ID delivery signal format
- Voice-Mail messages waiting indication
- Impedance
- On/off hook and hook flash detection timing
- FAX tones detection

The FXS interface is the same as that in the SPA-2000. Please refer to the SPA-2000 section for more details on configuration of this interface.

4.2.2. FXO Interface

This is the interface to the FXO/LINE port. It controls the exchange of signals between the SPA and the PSTN line attached to the FXO/LINE port. It draws power from the PSTN line and controls the on/off hook state. This interface can be configured under the [Regional] and [PSTN Line] tabs, with the following functions:

- Polarity reversal detection: SPA can detect polarity reversal of the tip-to-ring voltage as a PSTN disconnect signal (section 4.7)
- CPC signal detection: SPA can detect CPC or momentary removal of tip-to-ring voltage as a PSTN disconnect signal (section 4.7)
- Disconnect Tone detection: SPA can detect the occurrence of disconnect tone on the PSTN line; the characteristics of this tone is configurable (section 4.7)
- PSTN voice activity detection: SPA can monitor voice activity on the PSTN line and consider the call has ended if no activity for a long period (section 4.7)
- Ring detection: The characteristics of the ringing signal to detect can be finely adjusted with the following parameters:
    - o <Ring Frequency Min> - lower limit of the frequency of valid ring signal
    - o <Ring Frequency Max> - upper limit of the frequency of valid ring signal
    - o <Ring Validation Time> - minimum duration of valid ring signal
    - o <Ring Threshold> - minimum Vrms of a valid ring signal
    - o <Ring Timeout> - delay in de-asserting a ring signal after it is removed from the PSTN line
    - o <Ring Indication Delay> - delay in assert ing a ring signal after it is detected on the PSTN line

- DTMF detection: Detects DTMF digits on the PSTN line
- FAX Tones detection: Detects FAX CED and CNG tones on the PSTN line
- Parallel handset detection: SPA detects if the PSTN line is being used by another extension sharing the line, if the tip-to-ring voltage drops below the <Line-In-Use Voltage>
- <SPA to PSTN Gain>: Increase or decrease the signal level sent to the PSTN line. The valid range is from -15 dB to 12 dB in 1 dB increment. Note: Increasing this gain may increase the level of echo heard on the VoIP call leg, while decreasing it may reduce the level of the same echo
- <PSTN to SPA Gain>: Increase or decrease the signal level received from the PSTN line. The valid range is from -15 dB to 12 dB in 1 dB increment. Note: Increasing this gain may increase the level of the echo heard on the VoIP call leg, while decreasing it may reduce the level of the same echo
- Caller-ID detection and decoding: SPA can detect and decode Bellcore Type I Caller-ID (FSK) signal on the PSTN line after the first ring
- <FXO Port Impedance>: SPA supports 16 impedance settings
- Sends these signals to the PSTN Line: Tones, DTMF, On/Off Hook
- Miscellaneous parameters for international compliance control:
    - o <Tip/Ring Voltage Adjust> - Adjust the Tip/Ring voltage on the PSTN line
    - o <Operational Loop Current Min> - Adjust the minimum loop current at which the SPA can operate
    - o <Current Limiting Enable> - If enabled, it limits the loop current to a maximum of 60mA per the TBR21 standard
    - o <Ringer Impedance> - Set the impedance of the ringer
    - o <On-Hook Speed> - Adjust the time for the loop current to drop to 0 after the SPA takes the FXO port on-hook

To get the most out of your SPA-3000, it is highly recommended to connect a PSTN service to the LINE port with Type I Caller-ID subscription. This will allow the following additional functionalities:

- Limit the use of the gateway by PSTN Caller-ID number
- Selectively forward PSTN callers to different VoIP destinations

Notes:
- The <Caller-ID Method> parameter under [Regional] tab only controls the Caller-ID signal format sent by the SPA to the Caller-ID device attached to the FXS/PHONE port; it does not apply to the caller-id signal format sent to the SPA by the PSTN switch via the FXO/LINE port. At present the SPA can only decode Bellcore FSK style Caller-ID signal sent by the PSTN switch.

4.2.3. VoIP Interfaces

There are 2 VoIP interfaces in the SPA-3000. Each VoIP interface can be configured to register with a VoIP Service Provider (VSP), and to receive and make calls over the IP network. Depending on the functionality you have in mind, you can configure either or both interfaces. The VoIP1 and VoIP2 interfaces correspond to the Line 1 VoIP service, and the PSTN Line VoIP service, respectively.

The [SIP] parameter group and a portion of the [Regional] parameter group apply to both VoIP interfaces. The [Line 1] and the [User 1] parameter groups are dedicated to VoIP1, while the [PSTN Line] and the [PSTN User] parameter groups to VoIP2.

VoIP1 and VoIP2 interfaces can be configured independently with the same or different VSP. The same VSP account can be configured for both interfaces but the <SIP Port> parameter must be different for each interface in this case.

Most VSP require the following parameters configured on a VoIP interface:
- <Proxy>
- <User ID>
- <Password>
- <Register>
- <Register Expires>

Some VSP may also require the following parameters to be configured:
- <Outbound Proxy> (from VSP) and <Use Outbound Proxy> = yes
- <Auth ID> and <Use Auth ID> = yes

4.2.4. Call Types

The type of calls supported by the SPA-3000 can be described in terms of the originating, intermediate, and terminating interfaces involved:

1. FXS to VoIP1
2. VoIP1 to FXS
3. FXO to VoIP2
4. VoIP2 to FXO
5. VoIP1 to FXO
6. FXO to VoIP2 to VoIP1 to FXS
7. FXS to VoIP1 to VoIP2 to FXO

#1 and #2 are conventional VoIP calls. #3 lets a PSTN caller hop-on to use VoIP service. #4 lets a VoIP caller hop-off to use PSTN service. #5 is similar to #4 except the caller establishes the VoIP call leg by calling the VoIP1 interface instead of the VoIP2 interface. #6 allows the PSTN call to ring the FXS port phone; we called this “ringing thru”. #7 allows you to call the PSTN from the phone.

Notes:

- VoIP1-to-VoIP2 and VoIP2-to-VoIP1 are internal SIP calls. Signaling and media packets are sent to the loop back address 127.0.0.1. Hence call types #6 and #7 are possible even if the Ethernet port is not connected

The following sections describe each type of calls in more details.

4.2.5. Determining the Availability of the PSTN line

SPA determines that the PSTN line is not available if the one of the following conditions is true:

- PSTN line is not connected (loop current is 0 or Tip/Ring RMS voltage is below 1V)
- PSTN line is being used by another extension. Tip/Ring RMS voltage lower than the threshold set in <Line-In-Use Voltage>
- PSTN line is ringing
- PSTN line is being used by the SPA to serve another VoIP caller

If the PSTN line is not available, the PSTN gateway function will be rejected; any VoIP caller requesting PSTN gateway functions will be turned down with a “Service Not Available” response.

4.3. Gateway Call Restriction by Dial Plan

Gateway calls can be restricted on a per caller basis using dial plans. Up to 8 dial plans can be configured to restrict gateway calls in either direction. The corresponding SPA parameters are:

- [PSTN Line]<Dial Plan *n*>, *n* = 1 to 8

Notes:

- The dial plan length limit for <Dial Plan 1> through <Dial Plan 8> is 511 characters. This is less than that for the <Dial Plan> under [Line 1], which is 2047 characters.
- “gw0” – “gw4” syntaxes are not applicable to <Dial Plan 1> – <Dial Plan 8> (section 4.8)
- While PSTN-To-VoIP calls must have dial plan (1 – 8) specified, the dial plan can be set to “none” for VoIP-To-PSTN calls. If the dial plan is “none”, the SPA will take the FXO port off-hook and the VoIP caller will hear the PSTN dial tone directly from the PSTN switch immediately after successful authentication; the target number dialed by the VoIP caller will be directly interpreted by the PSTN company. On the other hand, if the dial plan is not “none”, the VoIP caller will hear the “Outside Dial Tone” generated by the SPA, where the caller will dial the target PSTN number. This number will be received and processed by the SPA with the chosen dial plan (the FXO port is still on-hook while the caller is entering the target number). If the target number is valid, the SPA then takes the FXO port off-hook and automatically dials the final target number (according to the chosen dial plan) out to the PSTN line. The characteristics of the Outside Dial Tone can be changed by modifying the <Outside Dial Tone> parameter (under [Regional] tab).
- You can “forward” all VoIP callers of the PSTN Line to a certain PSTN number (with or without authentication) by setting a hotline rule in the dial plan, such as (S0<:14089991234>) which sends all callers to 14089991234 automatically once the SPA auto-answers and authenticates the VoIP call. To make this works more transparently to the caller, you may disable authentication or add the caller to the access list.

4.4. Authentication Methods

VoIP Callers can be authenticated by one of the following methods by setting the <VoIP Caller Auth Method> parameter:

1. No Authentication: All callers will be accepted for service. The dial plan to be used for all VoIP callers for this case is the one selected in <VoIP Caller Default DP>.
2. PIN: Caller is prompted to enter a VoIP PIN right after the call is answered (i.e., after the SPA replies with a 200 response). Up to 8 PIN's, <VoIP Caller 1 PIN> to <VoIP Caller 8 PIN>, can be setup to access the PSTN gateway and each PIN can be assigned a different dial plan using the <VoIP Caller 1 DP> to <VoIP Caller 8 DP> parameters, respectively. The caller will hear a beep-beep-beep tone, known as the VoIP PIN Tone, as the prompt to enter the VoIP PIN. The tone will be played repeatedly until a PIN digit is received. After the first digit, the user by default will have 10s to enter each subsequent PIN digit. If no new PIN digit is entered for more than 10s, the SPA will terminate the call. This inter-PIN-digit timeout is set in the <VoIP PIN Digit Timeout> parameter. The PIN number entry must be ended by a pound (#) key. By default the caller will be given 3 chances to enter a valid PIN. If no valid PIN is received after 3 trials, the SPA will terminate the call. The number of VoIP PIN entry trials to allow can be changed by setting the <VoIP PIN Max Retry> parameter. If the last PIN entry is invalid, the VoIP PIN Tone will resume and the caller can re-enter the PIN again. The characteristics of the VoIP PIN Tone can be changed by setting <VoIP PIN Tone> ([Regional] tab). The VoIP PIN Tone should be set with a finite timeout (the default VoIP PIN tone has a timeout value of 10s). If no valid PIN is received after the maximum number of trials, the SPA will terminate the call.
3. HTTP Digest: SIP INVITE must contain a valid Authorization header that is computed based on an Auth ID and a password using MD5 digest algorithm. The Auth ID must be specified in the username parameter in the Authorization header. Up to 8 Auth ID/Passwords, <VoIP User 1 Auth ID>/<VoIP User 1 Password> to <VoIP User 8 Auth ID>/<VoIP User 8 Password>, can be setup to access the PSTN gateway. Each Auth ID/Password can be assigned a different dial plan using the <VoIP User 1 DP> to <VoIP User 8 DP> parameters, respectively. If the inbound INVITE to the PSTN Line does not have an Authorization header or the credentials it contains are invalid, the SPA will reply with a 401 response. If the username parameter in the Authorization header does not match any of the <VoIP User *n* Auth ID>, *n* = 1 to 8, the SPA will reply with a 403 response. If the INVITE request a target PSTN number (as in one-stage dialing) that is not allowed by the corresponding dial plan for that caller, the SPA will also reply with a 403 response.

VoIP callers can also be gated by a list of Caller-ID patterns before authentication rules are applied. These patterns are specified in <VoIP Caller ID Pattern> which is a comma separate list of Caller-ID patterns. The VoIP Caller-ID is extracted from the inbound INVITE request FROM header User-ID field. If the FROM header has "Anonymous" (case-insensitive) in the display-name field, however, the SPA treats the VoIP Caller-ID as "Anonymous". Each Caller-ID pattern is a case insensitive alpha-numeric string with special wildcard characters '?' and '*' , which stands for "any single digit" and "0 or more of any digits" respectively. For example: 1408*,15101234567,18??*,anonymous,jsmith. It is recommended not to insert white spaces before and after the comma, although they are allowed. If <VoIP Caller ID Pattern> is blank, all VoIP callers will be processed by the SPA for authentication and subsequent gateway services upon successful authentication. If <VoIP Caller ID Pattern> is not blank, then the VoIP Caller ID must match one of the given patterns or else will be rejected by the SPA with a 403 response without further processing.

In addition, a VoIP Caller can be automatically accepted for PSTN gateway access without going through the authentication process if the source IP address of the inbound INVITE request matches one of the pattern specified in the <VoIP Access List>. This is a comma separated list of IP address patterns, also with special wildcard characters '?' and '*'. For example:192.168.2.*,66.12?.12?.4. If the list is not blank and the source IP address of the inbound INVITE matches any of the patterns in the list, the VoIP caller will be granted access to the PSTN gateway as if the Authentication Method is set to “none” (and so the <VoIP Caller Default DP> applies and 1-stage dialing is possible in this case).

Notes:

- One-stage dialing is possible only if <VoIP Caller Auth Method> is “none” or “HTTP Digest”. Unless the caller number is in the <VoIP Access List >, the “PIN” method requires 2-stage dialing: Caller will need to dial the target PSTN number after entering a valid PIN. One-stage dialing can be globally disabled by setting <One Stage Dialing> to “no”, for which case all VoIP callers (including Line 1) will be required to dial the PSTN target number upon successful authentication.
- If the VoIP Caller is calling from Line 1 of the same unit. Authentication is skipped regardless the setting of <VoIP Caller Auth Method>. The dial plan to use in this case in <Line 1 VoIP Caller DP> for normal operation, or <Line 1 Fallback DP> for Line-1-Fallback-To-PSTN operation (when network link is down or Line 1 registration fails).

PSTN Callers can be authenticated by one of the following methods by setting the <PSTN Caller Auth Method> parameter:

1. No Authentication: All Callers will be accepted for service. In this case the dial plan to be used for all PSTN callers is taken from <PSTN Caller Default DP>.
2. PIN: Caller is prompted to enter a PSTN PIN right after the call is auto-answered by the SPA (by taking the FXO port off-hook). Up to 8 PIN’s, <PSTN Caller 1 PIN> to <PSTN Caller 8 PIN>, can be setup to access the VoIP gateway and each PIN can be assigned a different dial plan using the <PSTN Caller 1 DP> to <PSTN Caller 8 DP> parameters, respectively. The caller will hear a beep-beep-beep tone, known as the PSTN PIN Tone, as the prompt to enter the PSTN PIN. The tone will be played repeatedly until a PIN digit is received. After the first digit, the user by default will have 10s to enter each subsequent PIN digit. If no new PIN digit is entered for more than 10s, the SPA will play the Reorder Tone and then terminate the call by taking the FXO port on-hook. This inter-PIN-digit timeout is set in the <PSTN PIN Digit Timeout> parameter. The PIN number entry must be ended by a pound (#) key. By default the caller will be given 3 chances to enter a valid PIN. If no valid PIN is received after 3 trials, the SPA will terminate the call. The number of PSTN PIN entry trials to allow can be changed by setting the <PSTN PIN Max Retry> parameter. If the last PIN entry is invalid, the PSTN PIN Tone will resume and the caller can re-enter the PIN again. The characteristics of the PSTN PIN Tone can be changed by modifying <PSTN PIN Tone> ([Regional] tab). The PSTN PIN Tone should be set with a finite timeout (the default PSTN PIN tone has a timeout value of 10s). If no valid PIN is received after the maximum number of trials, the SPA will play reorder tone and then terminate the call.

PSTN callers can also be gated by a list of Caller-ID patterns before authentication rules are applied. These patterns are specified in <PSTN Caller ID Pattern>, a comma separate list of Caller-ID patterns. The PSTN Caller-ID is decoded from the signal delivered by the PSTN switch. It is highly recommended that the PSTN line connected to the SPA comes with the caller-id delivery service. If Caller-ID signal is not present or the Caller-ID number is blocked, the SPA treats the PSTN Caller-ID as “Anonymous”. Each Caller-ID number is a case-insensitive alphanumeric string with special wildcard characters ‘?’ and ‘*’ , which stands for “any single digit” and “0 or more of any digits” respectively. For example: 1408*,15101234567,18??*,anonymous. It is recommended not to insert white spaces before and after the comma, although they are allowed. If <PSTN Caller ID Pattern> is blank, all PSTN callers will be processed by the SPA for authentication and subsequent gateway services upon successful authentication. If <PSTN Caller ID Pattern> is not blank, then the PSTN Caller ID must match one of the given patterns or else the SPA will not answer the call.

In addition, a PSTN caller can be automatically accepted for VoIP gateway access without going through the authentication process if the Caller-ID number matches one of the patterns specified in the <PSTN Access List>. This is a comma separated list of Caller-ID patterns, also with special wildcard characters ‘?’ and ‘*’ (same syntax as <PSTN Caller ID Pattern>). If the list is not blank and the Caller-ID number matches any of the patterns in the list, that PSTN caller will be granted access to the VoIP gateway as if the Authentication Method is set to “none” (and so the <PSTN Caller Default DP> applies in this case).

Notes:
- Only 2-stage dialing is possible with PSTN-To-VoIP gateway calls

The configuration parameters mentioned in this section are:

- [PSTN Line]<One Stage Dialing>
- [PSTN Line]<VoIP Auth Method>
- [PSTN Line]<VoIP Caller ID Pattern>
- [PSTN Line]<VoIP Access List>
- [PSTN Line]<VoIP PIN Max Retry>
- [PSTN Line]<VoIP PIN Digit Timeout>
- [Regional]<VoIP PIN Tone>
- [PSTN Line]<VoIP Caller Default DP>
- [PSTN Line]<VoIP Caller *n* PIN>, *n* = 1 to 8
- [PSTN Line]<VoIP Caller *n* DP>, *n* = 1 to 8
- [PSTN Line]<VoIP User *n* Auth ID>, *n* = 1 to 8
- [PSTN Line]<VoIP User *n* Password>, *n* = 1 to 8
- [PSTN Line]<VoIP User *n* DP>, *n* = 1 to 8
- [PSTN Line]<Line 1 VoIP Caller DP>
- [PSTN Line]<Line 1 Fallback DP>
- [PSTN Line]<PSTN Auth Method>
- [PSTN Line]<PSTN Caller ID Pattern>
- [PSTN Line]<PSTN Access List>
- [PSTN Line]<PSTN PIN Max Retry>
- [PSTN Line]<PSTN PIN Digit Timeout>
- [Regional]<PSTN PIN Tone>
- [PSTN Line]<PSTN Caller Default DP>
- [PSTN Line]<PSTN Caller *n* DP>, *n* = 1 to 8

4.5. VoIP-To-PSTN Calls (Call Type #4)

In order to obtain PSTN services via VoIP, the VoIP caller must establish connection with the SPA-3000 by way of a standard SIP INVITE request addressed to the SIP account configured under the [PSTN Line] tab. The PSTN gateway can be configured to support 1-stage and 2-stage dialing as described below. This is call type #4 and the voice path for this type of calls is (5)→(6)→(7)

4.5.1. One-Stage Dialing

One-stage dialing is possible if <One Stage Dialing> is “yes”, <VoIP Authentication Method> is “none” or “HTTP Digest” or if the source IP of the inbound INVITE matches one of the patterns specified in <VoIP Access List>. To perform one-stage dialing, the Request-URI of the INVITE to the PSTN Line should have the form *dialed-number@SPA-Address*, where *dialed-number* is the target PSTN number as “dialed” by the VoIP caller, and *SPA-Address* is a valid address of the SPA, such as 10.0.0.100:5061.

If the PSTN line is currently not available, the SPA replies to the INVITE with a 503 response. Otherwise, it compares the *dialed-number* with the <User ID> configured for the PSTN Line. If the *dialed-number* is not specified or is the same as <User ID>, the SPA interprets this as a request for 2-stage dialing (see next section). Otherwise, the SPA processes the *dialed-number* by a corresponding dial plan. If the dial plan processing fails, the SPA replies with a 403 response. Otherwise, it replies with a 200 response and at the same time takes the FXO port off hook and dials the final number returned from the dial plan to the PSTN switch.

4.5.2. Two-Stage Dialing

In 2-stage dialing, the VoIP caller will need to dial the target PSTN number upon successful authentication. If the dial plan configured for this VoIP caller is “none”, SPA will take the FXO port off-hook but will not dial any digits automatically after accepting the caller for gateway service. Hence the caller will hear the dial tone directly provided by the PSTN switch, which will interpret the target number dialed by the VoIP caller. If the dial plan is not “none”, the SPA will play the Outside Dial Tone to direct the VoIP caller to dial the PSTN number; the FXO port will stay on-hook while the SPA collects a complete PSTN target number from the caller according to the selected dial plan. If the dialed number is valid, the SPA takes the FXO port off-hook and dials the final target number returned from the dial plan to the PSTN switch accordingly. If the dialed number is invalid, the SPA terminates the call immediately.

To invoke 2-stage dialing, the VoIP caller can form a SIP INVITE request to send the PSTN Line without a user-id field in the Request-URI or with a user-id that matches exactly the <User ID> of the PSTN Line. Other user-id in the Request-URI will be treated as a request for 1-stage dialing (Section 4.5.1) if 1-stage dialing is enabled, or dropped by the SPA (as if no user-id is given) if 1-stage dialing is disabled.

The VoIP PIN digits and target number digits must be sent to the SPA out-of-band using the RFC2833 protocol (a.k.a. AVT Tone). The SPA does not accept any DTMF digits sent to it in-band over VoIP.

Notes:
- VoIP-To-PSTN Gateway function can be globally disabled by setting <VoIP-To-PSTN Gateway Enable> to “no”; SPA will reply with a 503 response to inbound INVITE sent to the PSTN Line
- If the PSTN line is not connected, or is in use by another extension or another VoIP caller, the SPA will reply with a 503 response to inbound INVITE sent to the PSTN Line
- The <User ID> of the PSTN Line can be blank. In that case Registration should be disabled for the PSTN Line
- When the SPA decides to accept an incoming INVITE, it immediately sends a 180 response to the VoIP caller, and eventually a 200 response to “answer” the call. You can set the desired delay before the SPA sends out the 200 response after the 180 response in the <VoIP Answer Delay> parameter. This delay can be 0.
- You can insert a small amount of delay before the SPA starts auto-dialing the final target number to the PSTN line after the SPA takes the FXO port off-hook. This delay is specified in the <PSTN Dialing Delay> parameter. This delay is used to make sure the PSTN switch is ready to receive DTMF before the SPA starts dialing.

Below is the pseudo code for accepting a VoIP caller for PSTN gateway service.

```
If (VoIP Caller-ID Pattern Blank or VoIP Caller-ID Matches a VoIP Caller-ID Pattern) {
  If (VoIP Caller in VoIP Access-List or Authentication Disabled) {
    Reply 200
    Start PSTN Gateway Service
  }
  Else {
```

```
      If (Authentication Method is HTTP) {
         If (Authorization Invalid) {
            Reply 401
         }
         Else {
            If (Target Number Valid) {
               Reply 200
               Start PSTN Gateway Service
            }
            Else {
               Reply 403
            }
         }
      }

      Else {
         Reply 200
         Get VoIP PIN from Caller
         If (Valid PIN) {
            Start PSTN Gateway Service
         }
         Else {
            Send BYE
         }
      }
   }
}

Else {
   Reply 403
}
```

The configuration parameters mentioned in this section are:

- [PSTN Line]<One Stage Dialing>
- [PSTN Line]<VoIP Caller Auth Method>
- [PSTN Line]<VoIP Access List>
- [PSTN Line]<VoIP Caller ID Pattern>
- [PSTN Line]<VoIP Caller *n* PIN>, *n* = 1 to 8
- [PSTN Line]<VoIP User *n* Auth ID>, *n* = 1 to 8
- [PSTN Line]<VoIP User *n* Password>, *n* = 1 to 8
- [PSTN Line]<VoIP Caller *n* DP>, *n* = 1 to 8
- [PSTN Line]<VoIP User *n* DP>, *n* = 1 to 8
- [PSTN Line]<User ID>
- [PSTN Line]<VoIP-To-PSTN Gateway Enable>
- [PSTN Line]<VoIP Answer Delay>
- [PSTN Line]<PSTN Dialing Delay>

4.6. PSTN-To-VoIP Calls (Call Type #3)

This is call type #3 with the voice path (7)→(6)→(5). PSTN-To-VoIP Calls can be made with 2-stage dialing only. The only authentication method available is the PIN method. The SPA auto-answers (i.e., takes the FXO port off-hook) after the PSTN line rings for a certain number of seconds. This auto-answer delay is configured in the <PSTN Answer Delay> parameter, and should be set to a larger enough value to allow enough time for the SPA to decode the Caller-ID signal sent by the switch (for US telcos, this value should be at least 3-4s). If <PSTN Caller Auth Method> is "PIN", the SPA then prompts the caller to enter the PIN number. If the given PIN matches one of <PSTN Caller *n* PIN> values, the SPA plays the Dial Tone to the FXO port and is ready to accept digits of the VoIP target number from the PSTN caller. The collected digits will be processed by the dial plan associated with the PIN number. If <PSTN Caller Auth Method> is "none", the SPA plays the Dial Tone immediately right after it auto-answers the PSTN call; the <PSTN Caller Default DP> is used for all PSTN Callers for this case. Note that the chosen dial plan for any PSTN caller cannot be "none".

If the caller enters a valid PSTN number according to the chosen dial plan, the PSTN generates an INVITE to establish the VoIP call leg via the VoIP2 interface configured under the [PSTN Line] tab. The From header of the INVITE will be the VoIP2 account, or substituted with the PSTN Caller-ID name and number if they are sent by the PSTN switch and decoded successfully by the SPA and <PSTN CID For VoIP CID> is set to "yes". Furthermore, the SPA will prepend the PSTN Caller-ID name and number with the prefixes configured in <PSTN CID Name Perfix> and <PSTN CID Number Prefix> respectively.

Notes:
- PSTN-To-VoIP Gateway service can be globally disabled by setting <PSTN-To-VoIP Gateway Enable> to "no". In that case, the SPA will not auto-answer any PSTN calls
- If the <Line Enable> ([PSTN Line] tab) is "no", or VoIP2 is not successfully registered and <Make Call Without Reg> ([PSTN Line] tab) is "no", or the Ethernet link is down, the SPA will not auto-answer the PSTN call as if <PSTN-To-VoIP Gateway Enable> is set to "no".
- Speed dial is allowed by the PSTN caller; 8 speed dials can be configured under [PSTN User] tab
- You can "forward" PSTN callers unconditionally with a hotline or warmline rule in the chosen dial plan, after the caller is successfully authenticated
- SPA supports selective call forwarding or blocking of PSTN callers such that a PSTN caller is either "forwarded" to a fixed VoIP number or "barred" from accessing the VoIP gateway (the call will not be answered by the SPA in the latter case). Selective call forwarding and blocking functions are configured under [PSTN User]. SPA does not authenticate the PSTN caller if he is configured to be forwarded.

Below is the pseudo code for accepting a PSTN caller for VoIP gateway service.

```
If (PSTN Caller ID Pattern Blank or PSTN Caller-ID Matches a PSTN Caller-ID Pattern) {
  Auto Answer (Off-Hook FXO Port)
  If (PSTN Caller-ID in PSTN Access-List or Authentication Disabled) {
start_service:
    Play Dial Tone
    Collect VoIP Target Number Digits from Caller
    If (Target Number Valid) {
      Start VoIP Call
    }
    Else {
      Play Reorder Tone
      Hang Up
    }
  }

  Else {
    Get PSTN PIN from Caller
    If (Valid PIN) {
      Goto start_service
    }
    Else {
      Play Reorder Tone
      Hang Up
    }
```

```
  }
}
```

The configuration parameters mentioned in this section are:

- [PSTN Line]<PSTN-To-VoIP Gateway Enable>
- [PSTN Line]<Line Enable>
- [PSTN Line]<Make Call Without Reg>
- [PSTN Line]<PSTN Caller ID Pattern>
- [PSTN Line]<PSTN Access List>
- [PSTN Line]<PSTN Answer Delay>
- [PSTN Line]<PSTN CID For VoIP CID>
- [PSTN Line]<PSTN CID Number Prefix>
- [PSTN Line]<PSTN CID Name Prefix>
- [PSTN Line]<PSTN Caller Auth Method>
- [PSTN Line]<PSTN Caller *n* PIN>, *n* = 1 to 8
- [PSTN User] <Speed Dial *n*>, *n* = 2 to 9
- [PSTN User]<Cfwd Sel*n* Caller>, *n* = 1 to 8
- [PSTN User]<Cfwd Sel*n* Dest>, *n* = 1 to 8

4.7. Terminating Gateway Calls

A gateway call has two call legs: the PSTN call leg and the VoIP call leg. A gateway call is terminated when either call leg is ended. It is very important that the SPA takes the FXO port on-hook when the call terminates or else the PSTN line cannot be used again. The SPA detects that the PSTN call leg is ended when one of the following conditions occur during a call:

1. The PSTN line tip-to-ring voltage drops to a very low value (< 1V) for a finite duration of time. This can happen if the PSTN line is disconnected from the FXO port, or when the PSTN switch sends a CPC signal to indicate that the call has been disconnected. The duration of this very low voltage must last for at least the length as specified in <Min CPC Duration>. The detection of CPC signal can be turned off by setting <Detect CPC> to “no”.
2. A polarity reversal is detected at the FXO port. The polarity reversal must last for at least 100ms or it will be ignored by the SPA. The detection of polarity reversal signal can be turned off by setting <Detect Polarity Reversal> to “no”
3. Disconnect Tone detected on the FXO port. The characteristics of the Disconnect Tone can be changed by modifying the <Disconnect Tone> parameter. The detection of Disconnect Tone can be turned off by setting <Detect Disconnect Tone> to “no”
4. When there is no voice activity detected from the PSTN line for a continuous period of time. The condition must last in continuation for at least the length of time specified in the <PSTN Long Silence Duration> parameter. The sensitivity of the detection can be adjusted by setting <PSTN Silence Threshold>; the higher the threshold, the easier the SPA will detect voice activity. The detection of this condition can be turned off by setting <Detect PSTN Long Silence> to “no”.

The SPA determines that the VoIP connection is ended (or broken) if:

1. SPA receives a SIP BYE request from the VoIP peer
2. No RTP packets received from the VoIP Peer for a continuous period of time larger than <VoIP Long Silence Duration> seconds. The detection of this condition can be turned off by setting <Detect VoIP Long Silence> to “no”
3. VoIP peer fails to respond to a periodic dialog refresh request from the SPA. The interval between periodic refresh messages is set in the <VoIP DLG Refresh Intvl>. The refresh message

is a SIP Re-INVITE request which the VoIP peer must reply with a 200 class response. The sending of dialog refresh messages can be disabled by setting <VoIP DLG Refresh Intvl> to 0.

When any of the above occurs, the SPA takes the FXO port on hook and sends the proper SIP signaling messages to end the VoIP call leg. In addition, you can limit the total duration of a VoIP gateway call and of a PSTN gateway call by setting the <PSTN-To-VoIP Call Max Dur> and <VoIP-To-PSTN Call Max Dur> parameters, respectively (setting either of these parameters to 0 imply the total duration of the gateway call is unlimited).

Finally, the VoIP or PSTN caller can enter * * # before hanging up to force the SPA to hang up the FXO port and tear down the VoIP call leg. This command can only be sent by the calling party; the SPA will not act on the command if sent by the called party. This command is especially useful if none of the PSTN disconnect signals can be detected reliably by the SPA.

The configuration parameters mentioned in this section are:

- [PSTN Line]<Detect CPC>
- [PSTN Line]<Min CPC Duration>
- [PSTN Line]<Detect PSTN Long Silence>
- [PSTN Line]<PSTN Long Silence Duration>
- [PSTN Line]<Detect VoIP Long Silence>
- [PSTN Line]<VoIP Long Silence Duration>
- [PSTN Line]<Disconnect Tone>
- [PSTN Line]<Detect Polarity Reversal>
- [PSTN Line]<Detect Disconnect Tone>
- [PSTN Line]<PSTN Silence Threshold>
- [PSTN Line]<VoIP DLG Refresh Intvl>
- [PSTN Line]<PSTN-To-VoIP Call Max Dur>
- [PSTN Line]<VoIP-To-PSTN Call Max Dur>

4.8. Line 1 VoIP Outbound Call Routing (Call Type #7)

The voice path for this call type is (1)→(2)→(4)→(6)→(7). Calls made from Line 1 are routed through the configured Line 1 service provider by default. This behavior can be overridden by IP dialing where the calls can be routed to any IP address entered by the user. SPA-3000 allows more flexible call routing with the addition of 4 sets of gateway parameters and new dial plan parameters:

- [Line 1]<Gateway *n*>, *n* = 1 to 4
- [Line 1]<GW*n* NAT Mapping Enable>, *n* = 1 to 4
- [Line 1]<GW*n* Auth ID>, *n* = 1 to 4
- [Line 1]<GW*n* Password>, *n* = 1 to 4

Gateways 1 to 4 are specified in a dial plan with the special identifiers “gw1”, … “gw4”. In addition, the identifier “gw0” represents the internal PSTN gateway via the FXO port. One can specify in the dial plan to use gw*n* (*n* = 0,1,2,3,4) when making certain calls. If more gateways are needed, one can specify any gateway address in the dial plan. There are 3 dial plan parameters that can be used with call routing: “usr”, “pwd”, and “nat” which are, respectively, the user-id (or authentication-id) and password to be used for authentication with the given gateway, and whether to enable NAT mapping when routing calls through that gateway. Below are some examples

Examples	Description
<9,:>xx.<:@gw1>	User dials 9 to start Outside Dial Tone, followed by 1 or more digits, and the SPA routes the call to Gateway 1.
[93]11<:@gw0>	Route 911 and 311 calls to the local PSTN gateway

<8,:1408>xxxxxxx<:@pstn.sipura.com:5061;usr=joe;pwd=joe_pwd;nat>	User dials 8 to start Outside Dial Tone. When user dials a 7-digit number, SPA prepends it with 1408 and routes the call to pstn.sipura.com:5061, with user-id = joe, and pwd = bell_pwd, and NAT mapping enabled
<8,:1408>xxxxxxx<:@gw2:5061;usr="Alex Bell";pwd="anything";nat=no>	User dials 8 to start Outside Dial Tone. When user dials a 7-digit number, SPA prepends it with 1408 and routes the call to Gateway 2, but to port 5061 with user-id = "Alex Bell", pwd = bell_pwd, and NAT mapping disabled

With the call-routing capability, one can use the same phone to make outbound calls to Line 1 VoIP or the PSTN line attached to the FXO port. In fact, the SPA can bridge a 3-way conference with one VoIP leg and one PSTN call leg. One can also setup multiple PSTN gateways at different locations and configured Line 1 to use different gateway when dialing certain numbers.

Notes:

- gw0 – gw4 identifiers can only be used in [Line 1]<Dial Plan>. They are not allowed in [PSTN Line]<Dial Plan *n*>, *n* = 1 to 8
- The "usr", "pwd", and "nat" parameters, on the other hand, are allowed in all dial plans
- The PSTN gateway will apply the <Line 1 VoIP Caller DP> to further limit the calls that can be made by the Line 1 caller to the PSTN; this dial plan may be set to "none". In general calls routed from Line 1 to the PSTN are processed by 2 dial plans.
- The SPA does not support call transferring the VoIP peer to the PSTN peer, or vice versa, in a 3-way call or 3-way conference that involves one VoIP call leg and one PSTN call leg.

4.9. Line 1 VoIP Fallback to PSTN

When power is removed from the SPA-3000, the FXS port will be connected to the FXO port. In this case, the telephone attached to the FXS port is electrically connected to the PSTN service via the FXO port. When power is applied to the SPA, the FXS port will be disconnected from the FXO port. However, if the PSTN line is in use when the power is applied to the SPA, the relay will not be flipped until the PSTN line is released. This is done so that the SPA will not interrupt any call in progress on the PSTN line.

When Line 1 VoIP service is down (due to registration failure or loss of Ethernet link), SPA can be configured to automatically route all outbound calls to the internal gateway if <Auto PSTN Fallback> ([Line 1] tab) is set to "yes". The PSTN gateway applies the <Line 1 Fallback DP> to further limit the calls that can be made by the Line 1 caller during the fallback operation; this dial plan may be set to "none". This case also belongs to call type #7 and the voice path is (1)→(2)→(4)→(6)→(7).

4.10. VoIP-To-PSTN Calls Via VoIP1 Interface (Call Type #5)

All PSTN gateway calls can be routed from the VoIP2 interface if the user can dedicated one VoIP account for the PSTN Line. Line 1 and the PSTN Line can also be configured with the same VoIP account if they use different <SIP Port>. If the service provider allows multiple REGISTER contacts and simultaneous ringing, both VoIP interfaces can register periodically with the service provider. In this case, both VoIP interfaces will receive inbound calls to this shared account. The PSTN Line should be configured with a sufficiently long <VoIP Answer Delay> before the call is automatically answered to provide PSTN gateway function.

If the service provider does not allow more than one REGISTER contacts, then the PSTN Line should not register. In this case, only Line 1 will ring on the inbound call to this VoIP account (since it is the only line registered with the service provider). Line 1 can have the call "forwarded" to the PSTN line after a configurable delay using the Call-Forward-On-No-Answer feature with "gw0" as the forward destination. This is Call Type #5 and the voice path is (3)→(8)→(7). Similarly, Line 1 can apply Call-Forward-All, Call-Forward-On-Busy, and Call-Forward-Selective features and send the caller to use the PSTN gateway.

Only the PIN Authentication method is allowed when a VoIP caller is forwarded to access the PSTN gateway from Line 1. If <VoIP Caller Auth Method> is “HTTP Digest”, the SPA treats it as if VoIP caller authentication is disabled. Line 1 call forward destinations are configured under the [User 1] tab.

An extension to the Forward-To-GW0 feature is to forward the caller to a specific PSTN number, using the syntax *PSTN-number*@gw0 in the call forward destination in [User 1]. When using this with Call-Forward-Selective for instance, one can come up with some very interesting applications. For example, you can forward all callers with 408 area code to 14081234567, or all callers with 800 area code to 18005558355. When this syntax is used, authentication is not skipped regardless the settings of <VoIP Caller Auth Method> and the target PSTN number will be automatically dialed by the SPA once the caller is forwarded to the gw0.

4.11. PSTN Call Ring Thru Line 1 (Call Type #6)

The voice path is (7)→(6)→(4)→(2)→(1). This feature is enabled by setting <PSTN Ring Thru Line 1> to “yes”. If enabled, all incoming PSTN calls will ring the Line 1 phone regardless the VoIP gateway function is enabled on the SPA or not. Hence the same phone can be used to receive calls from Line 1 VoIP and from the PSTN. If Line 1 is already engaged in another VoIP call at the moment the PSTN line rings, the SPA presents the PSTN call alert signal to the user by playing a call-waiting tone, and the user can then switch between the PSTN call and the VoIP call by hook-flashing (as she normally would with 2 VoIP calls).

SPA implements the ring-thru feature by making an internal VoIP call from the VoIP2 interface to the VoIP1 interface. A by-product of this approach is that the call forwarding and distinctive ringing settings on Line 1 also apply to the PSTN call ringing thru Line 1. Ring thru stops as soon as the VoIP gateway auto-answers the PSTN call, or the VoIP1 interface returns a busy signal to VoIP2.

If Caller-ID is available with the PSTN call, SPA will present it in the FROM header of the internal SIP INVITE message sent from the VoIP2 to the VoIP1 interface, which then passes it on to the phone attached to the FXS port as Type I or Type II Caller-ID. For this to happen, the SPA must be allowed sufficient time to completely decode the PSTN Caller-ID signal before it can relay the decoded caller-id information to the phone. For US Type I Caller-ID, the CID signal starts shortly after the first ring and ends before the second ring. The first ring usually lasts for about 2s, the interval between the first and second ring is about 4s, and a typical Caller-ID signal lasts for about 1s. If the first ring that SPA sends to the phone is of the same length as the first ring sent by the PSTN switch to the SPA, the SPA should be configured to ring thru Line 1 at least 1s after it detects that the PSTN line is ringing, such that by the time the SPA sends the Type I Caller-ID signal to the phone, it would have completely decoded the PSTN Caller-ID signal. This ring thru delay is configured in <PSTN Ring Thru Delay>. In a call-waiting scenario, the SPA needs to send Type II Caller-ID to the phone. In that case the delay should include the length of the PSTN first ring, which is usually much longer than the length of the first burst of call-waiting tone generated by the SPA. In this example, the delay should be about 3s and it can be configured in <PSTN Ring Thru CWT Delay>. In both cases, the <PSTN Answer Delay> should be set large enough for the complete Caller-ID signal to be sent to the phone before the SPA auto-answers the call. If PSTN Caller-ID is not available or the SPA has not completely decoded the PSTN Caller-ID signal by the time it sends Caller-ID signal to the phone, the signal it sends will be encoded with the VoIP account information configured for the [PSTN Line] instead of the PSTN Caller-ID name and number. Note that <PSTN CID For VoIP CID> does not apply when ringing thru Line 1.

When the PSTN call rings thru Line 1, you can assign a different <Default Ring> under the <PSTN User] tab so that it will sound differently from normal VoIP calls to Line 1. The <Default Ring> value affects both ringing signal and call-waiting tone. Unlike the <Default Ring> setting under the [User 1] tab, the <Default Ring> parameter under the [PSTN User] tab has the extra choice of “Follow Line 1”, which means to follow the Line 1 ringer settings (including distinctive ringing rules) instead of forcing it to use a particular ring cadence.

When the SPA inserts the decoded PSTN caller-id name and number into the SIP INVITE message sent to the VoIP1 interface, it also prepends the name and number by the prefixes configured in <PSTN CID Number Perfix> and <PSTN CID Name Prefix> respectively.

If the PSTN caller hangs up before Line 1 or the VoIP gateway answers the call, the Line 1 phone may continue to ring a little longer since it takes a few seconds for the SPA to realize that the PSTN line has indeed stopped ringing. This delay can be modified by setting the <PSTN Ring Timeout> value (default is 5s).

The configuration parameters mentioned in this section are:

- [PSTN Line]<PSTN Ring Thru Line 1>
- [PSTN Line]<PSTN Ring Thru Delay>
- [PSTN Line]<PSTN Ring Thru CWT Delay>
- [PSTN Line]<PSTN Ring Timeout>
- [PSTN Line]<PSTN Answer Delay>
- [PSTN Line]<PSTN CID For VoIP CID>
- [PSTN Line]<PSTN CID Name Prefix>
- [PSTN Line]<PSTN CID Number Prefix>
- [PSTN Line]<PSTN Ring Timeout>
- [PSTN User]<Default Ring>

4.12. Symmetric RTP

In a normal VoIP connection, the SPA sends RTP packets to the destination as specified in the SDP sent by the VoIP peer. When <Symmetric RTP> is set to “yes”, however, SPA will change the destination to send RTP packets to the source IP address and port of the inbound RTP packet last received by the SPA. This setup can facilitate symmetric NAT traversal at the remote end. The parameter is available under both [Line 1] and [PSTN Line] tabs.

- [Line 1] <Symmetric RTP>
- [PSTN Line] <Symmetric RTP>

4.13. Configuration Examples and Call Scenarios

In this section we walk through some typical scenarios where the SPA-3000 can be applied.

4.13.1. Setup VoIP1 and VoIP2 With Separate VoIP Accounts

You have 2 FWD (Free World Dialup) accounts A and B, where A is for regular VoIP service and B dedicated for gateway functions.

a) Without Using STUN or NAT Mapping

[SIP]<STUN Enable> = no
[SIP]<Substitute VIA Addr> = no
[Line 1]<Line Enable> = yes
[Line 1]<SIP Port> = 5060
[Line 1]<NAT Mapping Enable] = no
[Line 1]<NAT Keep Alive Enable] = no

[Line 1]<Proxy> = fwd.pulver.com
[Line 1]<Outbound Proxy> = fwdnat.pulver.com:5082
[Line 1]<Use Outbound Proxy> = yes
[Line 1]<Use OB Proxy in Dlg> = yes
[Line 1]<User ID> = userid_A
[Line 1]<Password> = password_A
[Line 1]<Register> = yes
[Line 1]<Register Expires> = 3600

[PSTN Line]<Line Enable> = yes
[PSTN Line]<SIP Port> = 5061
[PSTN Line]<NAT Mapping Enable] = no
[PSTN Line]<NAT Keep Alive Enable] = no
[PSTN Line]<Proxy> = fwd.pulver.com
[PSTN Line]<Outbound Proxy> = fwdnat.pulver.com:5082
[PSTN Line]<Use Outbound Proxy> = yes
[PSTN Line]<Use OB Proxy in Dlg> = yes
[PSTN Line]<User ID> = userid_B
[PSTN Line]<Password> = password_B
[PSTN Line]<Register> = yes
[PSTN Line]<Register Expires> = 3600

b) With STUN and NAT Mapping

Similar to (a) with the following changes:
[SIP]<STUN Enable> = yes
[SIP]<NAT Keep Alive Intvl] = 15
[SIP]<Substitute VIA Addr> = yes
[SIP]<STUN Server> = stun.fwdnet.net
[Line 1]<NAT Mapping Enable] = yes
[Line 1]<NAT Keep Alive Enable] = yes
[Line 1]<NAT Keep Alive Msg] = (blank)
[Line 1]<NAT Keep Alive Dest> = $PROXY
[Line 1]<Use Outbound Proxy> = no
[PSTN Line]<NAT Mapping Enable> = yes
[PSTN Line]<NAT Keep Alive Enable] = yes
[PSTN Line]<NAT Keep Alive Msg] = (blank)
[PSTN Line]<NAT Keep Alive Dest> = $PROXY
[PSTN Line]<Use Outbound Proxy> = no

4.13.2. Setup VoIP1 and VoIP2 with Same VoIP Account

You have only one VoIP account and your service provider does not allow multiple registration. Setup the subscriber and proxy information as in the last section (with A = B) and make the following changes:

[PSTN Line]<Register> = no
[PSTN Line]<Make Call Without Reg> = yes
[PSTN Line]<Ans Call Without Reg> = yes

4.13.3. PSTN-To-VoIP Call Without Ringing Thru Line 1

Assume the VoIP2 interface is properly configured and successfully registered, and

[PSTN Line]<PSTN Ring Thru Line> = no
[PSTN Line]<PSTN Answer Delay> = 14
[PSTN Line]<PSTN Caller 1 PIN> = 1234
[PSTN Line]<PSTN Caller *n* PIN> = (blank), *n* = 2 to 8
[PSTN Line]<PSTN Caller 1 DP> = 1
[PSTN Line]<Dial Plan 1> = (1408xxxxxxx|408xxxxxxx|xxxxxxx|1800xxxxxxx|800xxxxxxx)
[PSTN Line]<PSTN Caller Auth Method> = PIN
[PSTN Line]<PSTN PIN Max Retry> = 2
[PSTN Line]<PSTN CID For VoIP CID> = no
[Regional]<PSTN PIN Tone> = (default)
[Regional]<Dial Tone> = (default)
[Regional]<Reorder Tone> = (default)

When a user calls the PSTN line connected to the FXO port, the SPA VoIP gateway answers the call after 3 rings (about 14s). The SPA then prompts the caller to enter a PIN number by playing a beep-beep-beep tone. The user enters 1 and the tone goes again. The user continues to enter 234#. The SPA then plays a regular dial tone to prompt the user to enter a VoIP target number. The caller in this case is limited by the chosen dial plan to call 7-digit numbers, and 408 and 800 numbers with 10 or 11 digit dialing. The VoIP target will see the Caller-ID of the VoIP2 interface instead of the name and number of the PSTN caller.

When the call is done, either side can hang up. Alternatively, the caller can enter * * # to force the SPA to take the FXO port on-hook, or enter * * 1 to start a new VoIP gateway call (without the need for authentication again). For the latter case, the caller will hear regular dial tone again.

Should the caller enter a wrong PIN the first time, he will have a second chance to re-enter the valid PIN. If the caller enters a wrong PIN the second time, the SPA will play Reorder Tone for 10s and then hang up.

4.13.4. PSTN Call Answered By Line 1

The setup is similar to the last example except for the following changes

- [PSTN Line]<PSTN Ring Thru Line> = yes
- [PSTN Line]<PSTN Ring Thru Delay> = 1
- [PSTN Line]<PSTN Ring Thru CWT Delay> = 3
- [PSTN Line]<PSTN CID Name Prefix> = +
- [PSTN Line]<PSTN CID Number Prefix> = 990
- [PSTN User]<Default Ring> = 2
- [User 1]<Default Ring> = 1
- [Regional]<Ring*n* Cadence> = (default), *n* = 1 to 8

Assume the PSTN line has Type I Caller-ID service and the caller's name is "Joe Smith" and the caller's number is 14089991234.

a) Line 1 idle at the time the PSTN rings

When the PSTN line rings, Line 1 rings also. If Line 1 is picked up before the VoIP gateway auto-answers, it will be connected to the PSTN call. Once the call is answered by Line 1, the VoIP gateway will not be activated. The Line 1 phone rings differently with <Ring 2 Cadence> as compared to the other inbound calls received from the VoIP1 interface. The Caller-ID shown on the Line 1 phone should be "+Joe Smith, 99014089991234", with the prepending of the configured name and number prefixes.

b) Line 1 busy at the time the PSTN rings

If Line 1 is busy when the PSTN line rings, the SPA will not attempt to ring through, even if Line 1 later becomes idle while the PSTN is still ringing.

c) Line 1 connected on a VoIP call (via the VoIP1 interface) at the time the PSTN rings

When the PSTN line rings, the SPA plays Call Waiting Tone to alert the Line 1 user and shows call-waiting Caller-ID to the Line 1 phone. The user can switch between the VoIP call and the PSTN call by hook-flashing the phone.

4.13.5. VoIP-to-PSTN Call via VoIP2 Interface With PIN Authentication

Assume the PSTN Line has a dedicated VoIP account separated from Line 1, and setup

- [PSTN Line]<VoIP Caller ID Pattern> = (blank)
- [PSTN Line]<VoIP Access List> = (blank)
- [PSTN Line]<VoIP Answer Delay> = 0
- [PSTN Line]<VoIP Caller Auth Metho> = PIN
- [PSTN Line]<VoIP Caller 1 PIN> = 4321
- [PSTN Line]<VoIP Caller *n* PIN> = (blank), *n* = 2 to 8
- [PSTN Line]<VoIP Caller 1 DP> = none
- [Regional]<VoIP PIN Tone> = (default)

a) PSTN line available at the time of the call

When the VoIP2 interface is called, the SPA auto-answers immediately (by replying a 200 response to the inbound INVITE). The SPA then prompts VoIP caller for a PIN with a beep-beep-beep tone. There is only 1 PIN (4321) configured on the SPA, so the caller must enter 4321#. When a valid PIN is received, the SPA immediately takes the FXO port off-hook since no dial plan is set for this PIN. If the PSTN line is in service, the user shall hear PSTN dial tone at this point. The caller can now dial any target PSTN number.

Now you want to limit the VoIP caller to call only 408 area code and 7-digit numbers by making the following changes:

- [PSTN Line]<VoIP Caller 1 DP> = 2
- [PSTN Line]<Dial Plan 2> = (1408xxxxxxx|xxxxxxx)
- [PSTN Line]<PSTN Dialing Delay> = 3

With these new settings in place, the SPA plays the Outside Dial Tone after receiving the valid PIN, while leaving the FXO Port on-hook. The caller then enters the digits of the target PSTN number which will be received and analyzed by the SPA. If the target number is valid according to the dial plan, the SPA takes the FXO port off-hook, wait for 3s, and then dials the final number returned from the dial plan. The caller will also hear the PSTN dial tone for about 3s after the FXO port is taken off-hook.

b) PSTN line not available at the time of the call

In this case the SPA will reply to the inbound INVITE with a 503 response. Note that the SPA does not detect Dial Tone on the PSTN line. So if the PSTN line is out of service, the caller will get dead air after the SPA takes the FXO port off-hook.

4.13.6. VoIP-to-PSTN Call via VoIP2 Interface With HTTP Digest Authentication:

Assume the PSTN line is available and in service and the VoIP2 interfaces are properly configured. Setup:

- [PSTN Line]<VoIP Caller Auth Method> = HTTP Digest
- [PSTN Line]<VoIP User 1 Auth ID> = jdole
- [PSTN Line]<VoIP User *n* Auth ID> = (blank), *n* = 2 to 8
- [PSTN Line]<VoIP User 1 Password> = silicon-valley
- [PSTN Line]<VoIP User 1 DP> = 3
- [PSTN Line]<Dial Plan 3> = (1408xxxxxxx|1510xxxxxxx|<:1408>xxxxxxx)
- [PSTN Line]<One Stage Dialing> = yes
- [PSTN Line]<User ID> = 8899
- [PSTN Line]<PSTN Dialing Delay> = 2
- [PSTN Line]<VoIP Answer Delay> = 6

a) One-Stage Dialing

The caller device sends a SIP INVITE request to the VoIP2 interface with the User-ID 1112233 in the Request-URI. The SPA challenges the INVITE request with a 401 response. The caller device then retries the INVITE request with the proper credentials computed with the authentication ID “jdole” and the password “silicon-valley” using the MD5 digest algorithm, and embedded them in the Authorization header. The Authorization header must have the username parameter set to “jdole”, the authentication ID, or else the SPA will reply with a 403 response. If the credentials are incorrect, the SPA will challenge the INVITE again. Otherwise, the SPA takes the user-id 1112233 from the Request-URI and processed it by the corresponding dial plan. If the target number is invalid according to the dial plan, the SPA also replies 403 to the INVITE. In the current example, the target number is valid and the final number returned from the dial plan is 14081112233. The SPA immediately replies a 180 response and waits for about 6s. Then the SPA sends a 200 response, takes the FXO port off-hook, and waits for another 2s before dialing the final number to the PSTN line.

b) Two-Stage Dialing

Same as (a) but the INVITE does not specify a User-ID in the Request-URI or the User-ID is equal to “8899”, same as the User-ID of the VoIP2 interface. The initial handling of the call is the same as (a) up to the point where the SPA sends out a 200 response. Then, instead of taking the FXO port off-hook, the SPA plays the Outside Dial Tone and collects digits from the caller and processed them by the dial plan. Once a complete valid number is received, the SPA takes FXO off-hook and dials the final number to the PSTN line.

4.13.7. Line 1 Forward-On-No-Answer to PSTN Gateway

Assume the PSTN line is available and the VoIP1 interface is properly configured and successfully registered, and

- [User 1]<Cfwd No Ans Dest> = gw0
- [User 1]<Cfwd No Ans Delay> = 20

When the caller sends an INVITE request to the VoIP1 interface, the Line 1 phone starts ringing. If no one picks up the phone for about 20s, the call will be automatically answered by the SPA by replying to the INVITE from the VoIP1 interface with a 200 response. The Line 1 phone stops ringing and the PSTN gateway is ready to serve the VoIP caller. From then on the call is handled like a VoIP-To-PSTN gateway call.

Notes:
- Note that in this case HTTP authentication is not allowed since the VoIP1interface does not authenticate inbound INVITE requests. If you must authenticate VoIP callers via VoIP1, you have to set <VoIP Caller Auth Method> to PIN. Otherwise caller authentication is disabled.
- If the PSTN line is not available the moment the SPA attempts to forward the call to gw0, the SPA will not answer the VoIP call. The call forward rule is ignored and Line 1 will continue to ring.

- The <VoIP Caller ID Pattern> and <VoIP Access List> parameters still apply to the VoIP callers when they access the PSTN gateway via the VoIP1 interface. If the caller is not allowed by the <VoIP Caller ID Pattern> the SPA will not answer the call. If the caller belongs to the <VoIP Access List>, authentication is not required.

4.13.8. Line 1 Forward-All to PSTN Gateway

Assume PSTN line is available and VoIP1 is properly configured and successfully registered, and

- [User 1]<Cfwd All Dest> = gw0

This is the same as the Forward-On-No-Answer case, except that the PSTN gateway auto-answers the VoIP call to Line 1 immediately after the inbound INVITE is received by the VoIP1 interface. If the PSTN line is not available at the moment, the SPA will not answer the call.

4.13.9. Line 1 Forward-On-No-Answer to a Particular PSTN Number

Assume PSTN line is available, and VoIP1 is properly configured and successfully registered, and

- [User 1]<Cfwd No Ans Dest> = *target-PSTN-number*@gw0
- [User 1]<Cfwd No Ans Delay> = 20

This is very similar to the Forward-On-No-Answer-To-PSTN case, except that the SPA will automatically dial the given *target-PSTN-number* on the PSTN line right after it answers the VoIP call leg. This is a special case of 1-stage dialing where the target number is hard-wired in the configuration. The caller will not be authenticated in this case regardless the setting of <VoIP Caller Auth Method>. However the caller is still limited by <VoIP Caller ID Pattern>.

4.13.10. Line 1 Forward-Selective to PSTN Gateway or Number

This case is similar as the above cases of call forwarding to gw0, but applies only when VoIP caller's number matches a specific Caller-ID pattern. For example:

- [User 1]<Cfwd Sel1 Dest> = gw0
- [User 1]<Cfwd Sel1 Caller> = 1408*
- [User 1]<Cfwd Sel2 Dest> = 14154455666@gw0
- [User 1]<Cfwd Sel2 Caller> = 1510*

With this setup any VoIP caller in the 408 area code will be connected to the PSTN gateway for service, while any VoIP caller in the 510 area code will be forwarded to the PSTN number 1415445566.

4.13.11. From Line 1 Dials 9 to Access PSTN-Gateway for Local Calls

Insert the rule "<9,:1408>xxxxxxx<:@gw0>" to [Line 1]<Dial Plan>, and set

- [PSTN Line]<Line 1 VoIP Caller DP> = none

When the user picks up the Line 1 phone to make a call, he can dial 9 to invoke the Outside Dial, followed by a 7-digit number. The SPA then prepends the 7-digit number with 1408 and dials the final 11-digit number to the PSTN line. The Line 1 caller will not be authenticated.

4.13.12. From Line 1 Route 311 and 911 Calls to PSTN-Gateway

Insert the rule “[39]11<:@gw0>” to [Line 1]<Dial Plan>, and set

- [PSTN Line]<Line 1 VoIP Caller DP> = none

When the user picks up the Line 1 phone and dials 311 or 911, the call is routed to the PSTN gateway.

4.14. Summary of SPA-3000 Configuration Parameters

This section summarizes the parameters that are specific to the SPA-3000 only. Other parameters not included here are similarly defined as in SPA-2000; please consult the SPA administration guide for details for those common parameters.

4.14.1. PSTN Line – Dial Plans

Parameter	Description	Default	Range
Dial Plan 1	The first of 8 dial plans in the dial plan pool to be associated with a VoIP Caller or a PSTN Caller. Each dial plan in the pool is referenced by a index 1 to 8 corresponding to Dial Plan 1 to 8. The dial plan syntax is the same as that used for Line 1 (except that the identifiers gw0 – gw4 are not supported here)	(xx.)	Str512
Dial Plan 2–8	Same as above with ‘1’ replaced by ‘2’ – ‘8’	(xx.)	Str512

4.14.2. PSTN Line – VoIP-To-PSTN Gateway Setup

Parameter	Description	Default	Range
VoIP-To-PSTN Gateway Enable	Enable/Disable VoIP-To-PSTN Gateway functionality	yes	Bool
VoIP Caller Authentication Method	Method to be used to authenticate a VoIP Caller before granting access th PSTN gateway. Choose from {none, PIN, HTTP Digest}	none	Choice
VoIP PIN Max Retry	Number of trials to allow VoIP caller to enter a PIN number (used only if authentication method is set to PIN)	3	1 – 10
One Stage Dialing	Enable one-stage dialing (applicable if authentication method is none, or HTTP Digest, or caller is in the Access List)	yes	Bool
Line 1 VoIP Caller DP	Index of the dial plan in the dial plan pool to be used when the VoIP Caller is calling from Line 1 of the same SPA-3000 unit during normal operation (ie, not due to fallback to PSTN service when Line 1 VoIP service is down). Choose from {none, 1, 2, 3, 4, 5, 6, 7, 8} Note: Authentication is skipped for Line 1 VoIP caller	1	Choice
VoIP Caller Default DP	Index of the dial plan in the dial plan pool to be used when the VoIP Caller is not authenticated. Choose from {none, 1, 2, 3, 4, 5, 6, 7, 8}	1	Choice
Line 1 Fallback DP	Index of the dial plan in the dial plan pool to be used when the VoIP Caller is calling from Line 1 of the same SPA-3000 unit due to fallback to PSTN service when Line 1 VoIP service is down or no Ethernet	1	Choice

	link. Choose from {none, 1, 2, 3, 4, 5, 6, 7, 8}. Note: Authentication is skipped for Line 1 VoIP caller		
VoIP Caller ID Pattern	A comma separated list of caller number templates such that callers with numbers not matching any of these templates will be rejected for PSTN gateway service regardless of the setting of the authentication method. The comparison is applied before access list is applied. If this parameter is blank (not specified), all callers will be considered for PSTN gateway service. For example: 1408*, 1512???1234. Note: ‘?’ matches any single digit; ‘*’ matches any number of digits	[blank]	Str127
VoIP Access List	A comma separated list of IP address templates, such that callers with source IP address matching any of the templates will be accepted for PSTN gateway service without further authentication. For example: 192.168.*.*, 66.43.12.1??	[blank]	Str127
VoIP Caller 1 PIN	The first of 8 PIN numbers that can be specified to control access to the PSTN gateway by a VoIP Caller, when the <VoIP Caller Authentication Method> is set to “PIN”.	[blank]	Str31
VoIP Caller 2–8 PIN	Same as above with “1” replaced by “2”–“8”	[blank]	Str31
VoIP Caller 1 DP	Index of the dial plan in the dial plan pool to be associated with the VoIP caller who enters the PIN that matches <VoIP Caller 1 PIN>.	1	Choice
VoIP Caller 2–8 DP	Same as above with “1” replaced by “2”–“8”	1	Choice

4.14.3. PSTN Line – VoIP Users and Passwords (HTTP Authentication)

Parameter	**Description**	**Default**	**Range**
VoIP User 1 Auth ID	The first of 8 user-id’s that a VoIP Caller can use to authenticate itself to the SPA using the HTTP Digest method (ie, by embedding an Authorization header in the SIP INVITE message sent to the SPA. If the credentials are missing or incorrect, the SPA will challenge the caller with a 401 response). The VoIP caller whose authentication user-id equals to this ID is referred to VoIP User 1 of this SPA.	[blank]	Str31
VoIP User 2–8 Auth ID	Same as above with ‘1’ replaced by ‘2’ – ‘8’. Note: If the caller specifies an authentication user-id that does not match any of the VoIP User Auth ID’s, the INVITE will be rejected with a 403 response.	[blank]	Str31
VoIP User 1 DP	Index of the dial plan in the dial plan pool to be used with VoIP User 1.	1	Choice
VoIP User 2–8 DP	Same as above with ‘1’ replaced by ‘2’ – ‘8’	1	Choice
VoIP User 1 Password	The password to be used with VoIP User 1. The user assumes the identity of VoIP User 1 must therefore	[blank]	Str31

	compute the credentials using this password, or the INVITE will be challenged with a 401 response		
VoIP User 2–8 Password	Same as above with ‘1’ replaced by ‘2’ – ‘8’	[blank]	Str31

4.14.4. PSTN Line – PSTN-To-VoIP Gateway Setup

Parameter	**Description**	**Default**	**Range**
PSTN-To-VoIP Gateway Enable	Enable or disable PSTN-To-VoIP Gateway functionality. If set to “no”, gateway is disabled but PSTN calls still ring through Line 1 (if <PSTN Ring Thru Line 1> is enabled)	yes	Bool
PSTN CID For VoIP CID	If set to “yes”, the outbound VoIP call will assume the caller-id of the PSTN caller, if PSTN caller ID is available. Otherwise, the PSTN Line’s VoIP account information is used. The PSTN Caller ID is after the application of <PSTN CID Name Prefix> and <PSTN CID Number Prefix>	yes	Bool
PSTN CID Number Prefix	A prefix to prepend to the PSTN caller ID number when ringing through Line 1 or used in outbound VoIP call. Note that most caller-id devices can only display 0-9 for the caller number field.	[blank]	Str7
PSTN CID Name Prefix	A prefix to prepend to the PSTN caller ID name when ringing through Line 1 or used in outbound VoIP call.	[blank]	Str7
PSTN Caller Auth Method	Method to be used to authenticate a PSTN Caller for VoIP gateway services. Choose from {none, PIN}	none	Choice
PST PIN Max Retry	Number of trials to allow a PSTN Caller to enter a valid PIN number.	3	1 – 10
PSTN Ring Thru Line 1	If enabled, incoming calls will also ring Line 1 (after a delay as specified in <PSTN Ring Thru Delay>). Hence the Line 1 user can accept call waiting from the PSTN side. The caller-id from the PSTN service, if available, will be passed onto Line 1 also when ringing through (For this to work, the ring through delay must be set long enough such that PSTN caller-id is completely decoded before the SPA sends caller-id signal to the FXS port).	yes	Bool
PSTN Caller Default DP	Index of the dial plan in the dial plan pool to be used for the PSTN caller who does not require authentication (when authentication method is none, or when the caller’s number is in the access list). Choose from {1, 2, 3, 4, 5, 6, 7, 8}	1	Choice
PSTN Caller ID Pattern	A comma separated list of caller number templates such that PSTN callers with numbers not matching any of these templates will be rejected for VoIP gateway service regardless of the setting of the authentication method. The comparison is applied before access list is applied. If this parameter is blank (not specified), all callers will be considered for VoIP gateway service. The PSTN service must include Type I Caller-ID Delivery Service for this feature to work properly. If caller-id is blocked or not available, the caller-id is assumed to be	[blank]	Str127

	"Anonymous". For example: 1408*, 1512???1234, Anonymous		
PSTN Access List	A comma separated list of caller number templates such that PSTN callers with numbers matching any of these templates will be accepted for VoIP gateway service without authentication.	[blank]	Str127
PSTN Caller 1 PIN	The first of 8 PIN numbers for authenticating PSTN callers to obtain VoIP gateway services. The PSTN Caller entering a PIN same as this PIN is referred as PSTN Caller 1	[blank]	Str31
PSTN Caller 2–8 PIN	Same as above with '1' replaced by '2' – '8'.	[blank]	Str31
PSTN Caller 1 DP	Index of the dial plan in the dial plan pool to be used with PSTN Caller 1. Choose from {1, 2, 3, 4, 5, 6, 7, 8}	1	Choice

4.14.5. PSTN Line – FXO Timer Values – In seconds

Parameter	**Description**	**Default**	**Range**
VoIP Answer Delay	Delay in seconds before auto-answering inbound VoIP calls for the FXO account	3	0-255
PSTN Answer Delay	Delay in seconds before auto-answering inbound PSTN calls after the PSTN starts ringing	16	0-255
VoIP PIN Digit Timeout	Timeout to wait for the 1st or subsequent PIN digits from a VoIP caller	10	2-255
PSTN PIN Digit Timeout	Timeout to wait for the 1st or subsequent PIN digits from a PSTN caller	10	2-255
VoIP DLG Refresh Intvl	Interval between (SIP) Dialog refresh messages sent by the SPA to detect if the VoIP call-leg is still up. If value is set to 0, SPA will not send refresh messages and VoIP call-leg status is not checked by the SPA. The refresh message is a SIP ReINVITE and the VoIP peer must response with a 2xx response. If VoIP peer does not reply or response is not greater than 2xx, the SPA will disconnect both PSTN and VoIP call legs automatically.	30	0-255
PSTN Ring Thru Delay	Delay in seconds before starting to ring thru Line 1 after the PSTN starts ringing. In order for Line 1 to have the caller-id information, the delay should be set to larger than the delay required to complete the PSTN caller-id delivery.	1	0-255
PSTN Ring Thru CWT Delay	Similar to <PSTN Ring Thru Delay> but applies when Line 1 is already on a call (where the SPA alerts the Line 1 user of the PSTN call by playing a call-waiting tone instead of ringing).	3	0-255
PSTN-To-VoIP Call Max Dur	Limit on the duration of a PSTN-To-VoIP Gateway Call. Unit is in seconds. 0 means unlimited.	0	0-2147483647
VoIP-To-PSTN Call Max Dur	Limit on the duration of a VoIP-To-PSTN Gateway Call. Unit is in seconds. 0 means unlimited	0	0-2147483647
PSTN Dialing Delay	Delay after hook before the SPA dials a PSTN number	1	0-255

PSTN Ring Timeout	Delay after a ring burst before the SPA decides that PSTN ring has ceased	5	0-255

4.14.6. PSTN Line – PSTN Disconnect Detection

Parameter	**Description**	**Default**	**Range**
Detect CPC	CPC is a brief removal of Tip-and-ring voltage. If enabled, the SPA will disconnect both call legs when this the signal is detected during a gateway call	yes	Bool
Detect Polarity Reversal	If enabled, SPA will disconnect both call legs when this signal is detected during a gateway call. If it is a PSTN gateway call, the 1st polarity reversal is ignored and the 2nd one triggers the disconnection. For VoIP gateway call, the 1st polarity reversal triggers the disconnection.	yes	Bool
Detect PSTN Long Silence	If enabled, SPA will disconnect both call legs when the PSTN side has no voice activity for a duration longer than the length specified in <PSTN Long Silence Duration> during a gateway call	yes	Bool
Detect VoIP Long Silence	If enabled, SPA will disconnect both call legs when no RTP packets are received from the VoIP peer for a duration longer than the length specified in <VoIP Long Silence Duration> during a gateway call	yes	Bool
Detect Disconnect Tone	If enabled, SPA will disconnect both call legs when it detects the disconnect tone from the PSTN side during a gateway call. Disconnect tone is specified in the <Disconnect Tone> parameter, which depends on the PSTN service.	yes	Bool
PSTN Long Silence Duration	This is minimum length of PSTN silence (or inactivity) in seconds to trigger a gateway call disconnection if <Detect Long Silence> is "yes"	30	5-255
VoIP Long Silence Duration	This is minimum length of VoIP silence (i.e., NO-RTP-PACKETS) in seconds to trigger a gateway call disconnection if <Detect VoIP Long Silence> is "yes"	30	5-255
PSTN Silence Threshold	This parameter adjusts the sensitivity of PSTN silence detection. Choose from {very low, low, medium, high, very high}. The lower the setting, the easier to detect silence and hence easier to trigger a disconnection.	medium	Choice
Disconnect Tone	This is the tone script which describes to the SPA the tone to detect as a disconnect tone. The syntax follows a standard Tone Script with some restrictions. Default value is standard US reorder (fast busy) tone, for 4 seconds. Restrictions: 1. 2 frequency components must be given. If single frequency is desired, the same frequency is used for both 2. The tone level value is not used. –30 (dBm) should be used for now. 3. Only 1 segment set is allowed 4. Total duration of the segment set is interpreted as the minimum duration of the tone to trigger	480@-30,620@-30;4(.25/.25/1+2)	Str255

	detection 5. 6 segments of on/off time (seconds) can be specified. A 10% margin is used to validated cadence characteristics of the tone		
Min CPC Duration	Minimum duration (in seconds) of a low tip-and-ring voltage (below 1V) for the SPA to recognize as a CPC signal or PSTN line removal.	0.2	0.05 – 0.99

4.14.7. PSTN Line – International Control

Parameter	**Description**	**Default**	**Range**
FXO Port Impedance	Desired impedance of the FXO Port. Choose from { 600, 900, 270+750||150nF, 220+820||120nF, 370+620||310nF, 320+1050||230nF, 370+820||110nF, 275+780||115nF, 120+820||110nF, 350+1000||210nF, 0+900||30nF, 600+2.16uF, 900+1uF, 900+2.16uF, 600+1uF, Global}	600	Choice
SPA To PSTN Gain	dB of digital gain (or attenuation if negative) to be applied to the signal sent from the SPA to the PSTN side.	0	–15 to 12
PSTN To SPA Gain	dB of digital gain (or attenuation if negative) to be applied to the signal sent from the PSTN side to the SPA.	0	–15 to 12
Tip/Ring Voltage Adjust	Adjust the Tip/Ring voltage on the PSTN line. Choose from {3.1V, 3.2V, 3.35V, 3.5V}	3.5	Choice
Operational Loop Current Min	Adjust the minimum loop current at which the SPA can operate. Chose from {10mA, 12mA, 14mA, 16mA}	10	Choice
On-Hook Speed	Adjust the time for the loop current to drop to 0 after the SPA takes the FXO port on-hook. Choose from {Less than 0.5ms, 3ms (ETSI), 26ms (Australia)}	Less than 0.5ms	Choice
Current Limiting Enable	If enabled, it limits the loop current to a maximum of 60mA per the TBR21 standard	no	Bool
Ring Frequency Min	Lower limit of ring frequency to detect ringing signal	10	5-100
Ring Frequency Max	Upper limit of ring frequency to detect ringing signal	100	5-100
Ring Validation Time	Minimum duration of the ringing signal to be qualified as a ringing signal by the SPA. Choose from {100, 150, 200, 256, 384, 512, 640, 1024} (ms)	256ms	Choice
Ring Indication	Delay in asserting a ringing signal after detecting it	512ms	Choice

Delay	on the PSTN line. Choose from {0, 512, 768, 1024, 1280, 1536, 1792} (ms)		
Ring Timeout	Delay in de-asserting a ringing signal after detecting that it has been removed from the PSTN line. Choose from {0, 128, 256, 384, 512, 640, 768, 896, 1024, 1152, 1280, 1408, 1536, 1664, 1792, 1920} (ms)	640ms	Choice
Ring Threshold	Minimum Vrms threshold to detect ringing. Choose from {13.5–16.5, 19.35–2.65, 40.5–49.5} (Vrms)	13.5–16.5 Vrms	Choice
Ringer Impedance	Set the impedance of the ringer. Choose from {High, Synthesized(Poland, S.Africa, Slovenia)}	High	Choice
Line-In-Use Voltage	This is the absolute tip-to-ring voltage threshold below which the SPA will consider the PSTN line as being used by another extension and will therefore not permit PSTN gateway service.	30	5-48

4.14.8. Line 1 and PSTN Line – Audio Configuration

Parameter	Description	Default	Range
Symmetric RTP	If enabled, the SPA will stream RTP packets to the source IP address and port of the last received RTP packet. If not enabled or before the 1st RTP packet arrives, SPA will stream RTP packets according to the information extracted from the SDP sent by the peer.	yes	Bool

4.14.9. Line 1 – Gateway Accounts

Parameter	Description	Default	Range
Gateway 1	The first of 4 gateways that can be specified to be used in the <Dial Plan> to facilitate call routing specification (that overrides the given proxy information). This gateway is represented by "gw1" in the <Dial Plan>. For example, the rule "1408xxxxxxx<:@gw1> can be added to the dial plan such that when the user dials 1408+7digits, the call will be routed to Gateway 1. Without the <:@gw1> syntax, all calls are routed to the given proxy by default (except IP dialing)	[blank]	FQDN
Gateway 2–4	Same as above with '1' replaced by '1' – '4'	[blank]	FQDN
GW1 NAT Mapping Enable	If enabled, the SPA will use NAT mapping when contacting Gateway 1	no	Bool
GW1–4 NAT Mapping Enable	Same as above with '1' replaced by '1' – '4'	no	Bool
GW1 Auth ID	This is the authentication user-id to be used by the SPA to authenticate itself to Gateway 1	[blank]	Str31
GW1–4 Auth ID	Same as above with '1' replaced by '1' – '4'	[blank]	Str31
GW1 Password	This is the password to be used by the SPA to authenticate itself to Gateway 1	[blank]	Str31
GW1–4 Password	Same as above with '1' replaced by '1' – '4'	[blank]	Str31

4.14.10. Line 1 – VoIP Fallback To PSTN

Parameter	Description	Default	Range
Auto PSTN Fallbak	If enabled, the SPA will automatically route all calls to the PSTN gateway when the Line 1 proxy is down (registration failure or network link down).	yes	Bool

4.14.11. Line 1 – Dial Plan

Parameter	Description	Default	Range
Dial Plan	The dial plan syntax is expanded in the SPA-3000 to allow the specification of 3 parameters to be used with a certain gateway: 1. uid – the authentication user-id, 2. pwd – the authentication password, and 3. nat – if this parameter is present, use NAT mapping Each parameter is separated by a semi-colon (;). Furthermore, it recognizes “gw0”, “gw1”, …, “gw4” as the locally configured gateways, where “gw0” represents the local PSTN gateway in the same SPA-3000 unit. Example 1: “*1xxxxxxxxxx<:@fwdnat.pulver.com:5082;uid=jsmith;pwd=xyz” Example 2: “*1xxxxxxxxxx<:@fwd.pulver.com;nat;uid=jsmith;pwd=xyz” Example 3: “[39]11<:@gw0>	[blank]	Str2047

4.14.12. User1 – Call Forward Settings

Parameter	Description	Default	Range
Cfwd All Dest	In addition to normal call forward destination as used in the SPA-1000 and SPA-2000, one can specify in the SPA-3000: 1. gw0 – forward the caller to use the PSTN gateway 2. *PSTN-number*@gw0 – forward to caller to the PSTN number (dialed automatically by the SPA through the PSTN gateway)	[blank]	Str39
Cfwd Busy Dest	Same as above	[blank]	Str39
Cfwd No Ans Dest	Same as above	[blank]	Str39

4.14.13. User1 – Selective Call Forward Settings

Parameter	Description	Default	Range
Cfwd Sel1 Dest	Same as Cfwd All Dest	[blank]	Str39
Cfwd Sel2–8 Dest	Same as Cfwd All Dest	[blank]	Str39
Cfwd Last Dest	Same as Cfwd All Dest	[blank]	Str39

4.14.14. Regional – Call Progress Tones

Parameter	Description	Default	Range
VoIP PIN Tone	Specification of the tone played to prompt a VoIP caller for a PIN number (if PIN authentication is selected and the caller requires authentication to use the PSTN gateway)	600@-10;*(0/1/1,.1/.1/1,.1/.1/1,.1/.5/1)	Tone-Script
PSTN PIN Tone	Specification of the tone played to prompt a PSTN caller for a PIN number (if PIN authentication is selected and the caller requires authentication to use the VoIP gatway)	600@-10;*(0/.7/1,.2/.1/1,.2/.1/1,.2/.5/1)	Tone-Script

4.14.15. PSTN User – PSTN-To-VoIP Selective Call Forward Settings

Parameter	Description	Default	Range
Cfwd Sel1 Caller	First of 8 PSTN Caller Number Patterns to be blocked for VoIP gateway services or forwarded to a certain VoIP number. If the caller is blocked, the SPA will not auto-answers the call. If the caller is forwarded, SPA will skip authentication		Str39
Cfwd Sel2-8 Caller	Same as above with 1 replaced with ‘2’ – ‘8’		Str39
Cfwd Sel1 Dest	VoIP destination to forward a PSTN caller matching <Cfwd Sel1 Caller>. If this entry is [blank], the PSTN caller is blocked for VoIP service.		Str39
Cfwd Sel2-8 Dest	Same as above with 1 replaced with ‘2’ – ‘8’		Str39

4.14.16. PSTN User – PSTN-To-VoIP Speed Dial Settings

Parameter	Description	Default	Range
Speed Dial 2	The VoIP number to call when the PSTN caller dials a single digit ‘2’		Str39
Speed Dial 3-9	Same as above with ‘2’ replaced with ‘3’ – ‘9’		Str39

4.14.17. PSTN User – PSTN Ring Thru Line 1 Distinctive Ring Settings

Parameter	Description	Default	Range
Ring1 Caller	First of 8 PSTN Caller Number Patterns such that the corresponding ring will be used to ring through Line 1 if the PSTN caller matches this pattern.		Str39
Ring2-8 Caller	Same as above with ‘1’ replaced with ‘2’ – ‘8’		Str39

4.14.18. PSTN User – PSTN Ring Thru Line 1 Ring Settings

Parameter	Description	Default	Range
Default Ring	The default ring to be used to ring through Line 1. Choose from {1,2,3,4,5,6,7,8,Follow Line 1}. If "Follow Line 1" is selected, the ring to be used is determined by Line 1's distinctive ring settings.	1	Choice

4.14.19. Info – PSTN Line Status

State Parameters	Description
Hook State	Hook state of the FXO port. Either On or Off
Line Voltage	Tip-to-Ring Voltage at the FXO port
Loop Current	Loop current to the FXO port
Last Called VoIP Number	The last VoIP number called from the PSTN Line
Last VoIP Caller	The last VoIP caller to the PSTN Line
Last PSTN Disconnect Reason	Reason for SPA hanging up the FXO port. Can be one of the following: • PSTN Disconnect Tone • PSTN Activity Timeout • CPC Signal • Polarity Reversal • VoIP Call Failed • VoIP Call Ended • Invalid VoIP Destination • Invalid PIN • PIN Digit Timeout • VoIP Dialing Timeout • PSTN Gateway Call Timeout • VoIP Gateway Call Timeout • VoIP Activity Timeout
Last Called PSTN Number	The PSTN number dialed by the SPA (logged only if a non-trivial dial plan is used)
Last PSTN Caller	Name and number of the last PSTN caller
PSTN Activity Timer	Shows the time (ms) before the SPA disconnects the current gateway unless the PSTN side has some audio activity.
Call Type	May take one of the following values: PSTN Gateway Call = VoIP-To-PSTN Call (#4) VoIP Gateway Call = PSTN-To-VoIP Call (#3) PSTN To Line 1 = PSTN call ring through and answered by Line 1 (#6) Line 1 Forward to PSTN Gateway = VoIP calls Line 1 then forwarded to PSTN GW (#5) Line 1 Forward to PSTN Number =VoIP calls Line 1 then forwarded to PSTN number (#5) Line 1 To PSTN Gateway (#7) Line 1 Fallback To PSTN Gateway (#7)
PSTN State	May take one of the following values: - Idle - Collecting PSTN PIN - Invalid PSTN PIN

	- PSTN Caller Accepted - Connected to PSTN - PSTN Offhook/VoIP Ended
PSTN Tone	Indicate what tone is being played to the PSTN call leg
PSTN Peer Name	Name of the party at the PSTN call leg
PSTN Peer Number	Phone number of the party at the PSTN call leg
VoIP State	Same as Line 1 Call 1
Mapped SIP Port	Same as Line 1
Registration State	Same as Line 1
Last Registration At	Same as Line 1
Next Registration In	Same as Line 1
VoIP Tone	Same as Line 1 Call 1 (Indicate what tone is being played to the VoIP call leg)
VoIP Peer Name	Same as Line 1 Call 1 (Name of the party at the VoIP call leg)
VoIP Peer Number	Same as Line 1 Cal 1 (Phone number of the party at the VoIP call leg)
VoIP Call Encoder	Same as Line 1 Call 1 (Audio encoder being used for the VoIP call leg)
VoIP Call Decoder	Same as Line 1 Call 1 (Audio decoder being used for the VoIP call leg)
VoIP Call FAX	Same as Line 1 Call 1
VoIP Call Remote Hold	Same as Line 1 Call 1
VoIP Call Duration	Same as Line 1 Call 1
VoIP Call Packets Sent	Same as Line 1 Call 1
VoIP Call Packets Recv	Same as Line 1 Call 1
VoIP Call Bytes Sent	Same as Line 1 Call 1
VoIP Call Bytes Recv	Same as Line 1 Call 1
VoIP Call Decode Latency	Same as Line 1 Call 1
VoIP Call Jitter	Same as Line 1 Call 1
VoIP Call Round Trip Delay	Same as Line 1 Call 1
VoIP Call Packets Lost	Same as Line 1 Call 1
VoIP Call Packet Error	Same as Line 1 Call 1
VoIP Call Mapped RTP Port	Same as Line 1 Call 1

4.14.20. PSTN/VoIP Caller Commands via DTMF

Command	Description
**#	Disconnect the PSTN line (SPA will take the FXO port on-hook)
**1	End the current call and restarts dial tone

4.15. SPA-3000 Configuration Profile

The SPA-3000 can be remotely provisioned just like the SPA-2000, by uploading a configuration profile to the unit using TFTP, HTTP, or HTTPS. The SPA-3000 configuration profile has a similar format and syntax as the other members of the SPA family, other than the fact it contains some additional parameters. You must use a different version of SPC.EXE to convert the SPA-3000 profile into a binary format before uploading to the unit. The parameter name used in the profile is derived from the name of the corresponding parameter as appear on the SPA web page. The convention for converting from a web page parameter name to a profile parameter name is to replace each occurrence of space with an underscore. In addition, for parameters in the [Line 1] and [User 1] groups, append [1] to the end of the name. For parameters in the [PSTN Line] and [PSTN User] groups, append [2] to the end of the name. Below are some examples:

[Line 1]<Proxy>	Proxy[1]
[PSTN Line]<PSTN Answer Delay>	PSTN_Answer_Delay[2]
[SIP]<NAT Keep Alive Intvl>	NAT_Keep_Alive_Intvl

5. User Guidelines

The SPA can be configured to the custom requirements of the service provider, so that from the subscriber's point of view, the service behaves exactly as the service provider wishes – with varying degrees of control left with the end user. This means that a service provider can leverage the programmability of the SPA to offer sometimes subtle yet continually valuable and differentiated services optimized for the network environment or target market(s).

This section of the Administration Guide, describes how some of the supported basic and enhanced, or supplementary services could be implemented. The implementations described below by no means are the only way to achieve the desired service behavior.

To understand the specific implementation options of the below features, including parameters, requirements and contingencies please refer the section Configuration Parameters, section 0.

5.1. Basic Services

5.1.1. Originating a Phone Call

Service Description	Placing telephone a call to another telephone or telephony system (IVR, conference bridge, etc.). This is the most basic service.
User Action Required to Activate or Use	When the user picks up the handset, the SPA provides dial tone and is ready to collect dialing information via DTMF digits from the telephone Touchtone key pad.
Expected Call and Network Behavior	While it is possible to support overlapped dialing within the context of SIP, the SPA collects a complete phone number and sends the full number in a SIP INVITE message to the proxy server for further call processing. In order to minimize dialing delay, the SPA maintains a dial plan and matches it against the cumulative number entered by the user. The SPA also detects invalid phone numbers not compatible with the dial plan and alerts the user via a configurable tone (Reorder) or announcement.
User Action Required to Deactivate or End	Hang-up the telephone.

5.1.2. Receiving a Phone Call

Service Description	The SPA can receive calls from the PSTN or

	other IP Telephony subscribers
User Action Required to Activate or Use	When the telephone rings, pick up the handset and begin talking.
Expected Call and Network Behavior	Each subscriber is assigned an E.164 ID (phone number) so that they may be reached from wired or wireless callers on the PSTN or IP network. The SPA supplies ring voltage to the attached telephone set to alert the user of incoming calls.
User Action Required to Deactivate or End	Hang-up the telephone.

5.2. Enhanced Services

5.2.1. Caller ID

Service Description	If available, the SPA supports the generation and pass through of Caller ID information.
User Action Required to Activate or Use	No user action required. The user's telephone equipment must support Caller ID to display the caller's name and/or number.
Expected Call and Network Behavior	In between ringing bursts, the SPA can generate a Caller-ID signal to the attached phone when the phone is on-hook. As part of the INVITE message, the SPA sends the caller's name and number as it is configured in the profile.
User Action Required to Deactivate or End	No user action required. See CLIP and CLIR.

5.2.2. Calling Line Identification Presentation (CLIP)

Service Description	Some users will elect to block their Caller ID information for all outgoing calls. However, there may be circumstances where sending Caller ID information for a call is desired, i.e. trying to reach a party that does not accept Caller ID blocked calls.
User Action Required to Activate or Use	Lift the receiver Listen for dial tone Press *__ Listen for dial tone Dial the telephone number you are calling

Expected Call and Network Behavior	Caller ID will be sent to the distant party for this call only. Users must repeat this process at the start of each call.
User Action Required to Deactivate or End	No action required. This service is only in effect for the duration of the current call.

5.2.3. Calling Line Identification Restriction (CLIR) – Caller ID Blocking

Service Description	This feature allows the user to block the delivery of their Caller ID to the number they are calling. This feature must be activated prior to dialing each call and is only in effect for the duration of each call.
User Action Required to Activate or Use	Lift the receiver Listen for dial tone Press *__ Listen for dial tone Dial the telephone number you are calling You must repeat this process at the start of each call
Expected Call and Network Behavior	The user activates this service to hide his Caller ID when making an outgoing call.
User Action Required to Deactivate or End	No action required. This service is only in effect for the duration of the current call.

5.2.4. Call Waiting

Service Description	The user can accept a call from a 3rd party while engaging in an active call. The SPA shall alert the subscriber of the 2nd incoming call by playing a call waiting tone.
User Action Required to Activate or Use	If the you choose to answer the second call either: Press and release your phone's switch hook (the button you release when you take your phone off the hook) or Press the flash button (if your phone has one). This puts your first call on hold and automatically connects you to your second call. To put your second caller back on hold and

	return to your first caller, press the switch hook or flash button again. (You can alternate between calls as often as you like.)
Expected Call and Network Behavior	If the user is on a call when another call comes in they will hear a series of beeps / tones alerting them to the second call. The person calling will hear normal ringing.
User Action Required to Deactivate or End	See Cancel Call Waiting.

5.2.5. Disable or Cancel Call Waiting

Service Description	The SPA supports disabling of call waiting permanently or on a per call basis.
User Action Required to Activate or Use	To temporarily disable Call Waiting (for the length of one call): Before placing a call: Lift Receiver Press *__ Listen for dial tone then dial the number you want to call. Call Waiting is now disabled for the duration of this call only. To deactivate Call Waiting while on a call: Press the switch hook or flash button briefly. This puts the first call on hold. Listen for three short tones and then a dial tone. Press *__ Listen for dial tone then return to your call by pressing the switch hook or flash button. Call Waiting is now disabled for the duration of this call. To deactivate Call Waiting while on a permanent basis (until cancelled): Lift the receiver Listen for dial tone Press *__ You will hear a confirmation tone signaling your request to cancel Call Waiting has been accepted.

Expected Call and Network Behavior	Callers who dial your number will receive a busy signal or, if available, the caller will be forwarded to voice mail or another predetermined forwarding number.
User Action Required to Deactivate or End	If you have cancelled Call Waiting temporarily, no user action is required. If you deactivated call waiting and wish to reinstate the service, do the following: Lift the receiver Listen for dial tone Press *__ You will hear a confirmation tone signaling your request to cancel Call Waiting has been accepted.

5.2.6. Call-Waiting with Caller ID

Service Description	When the user is on the phone and has Call Waiting active, the new caller's Caller ID information will be displayed on the users phone display screen at the same time the user is hearing the Call Waiting beeps / tones.
User Action Required to Activate or Use	The telephone equipment connected to the SPA must support Call-Waiting with Caller ID.
Expected Call and Network Behavior	In between call waiting tone bursts, the SPA can generate a Caller-ID signal to the attached phone when it is off hook.
User Action Required to Deactivate or End	Not applicable.

5.2.7. Voice Mail

Service Description	Service Providers may provide voice mail service to their subscribers. Users have the ability to retrieve voice mail via the telephone connected to the SPA.
User Action Required to Activate or Use	The SPA indicates that a message is waiting by, playing stuttered dial tone when the user picks up the handset. To retrieve messages: Lift the receiver

	Listen for dial tone Dial the phone number assigned to the SPA You will be connected to the voice mail server and prompted by a voice response system with instructions to listen to your messages.
Expected Call and Network Behavior	When voice mail is available for a subscriber, a notification message will be sent from the Voice Mail server to the SPA. When the user dials their own phone number, the SPA connects the subscriber their voice mail system which can then connect them to their individual voice mail box.
User Action Required to Deactivate or End	Follow instructions of the voice mail system or simply hang-up the telephone.

5.2.8. Attendant Call Transfer

Service Description	Attendant Call Transfer lets a customer use their Touchtone phone to send a call to any other phone, inside or outside their business, including a wireless phones.
User Action Required to Activate or Use	While in a call with the party to be transferred: Press the switch hook or flash button on the phone to place the party on hold Listen for three short tones followed by dial tone Dial the number to which you will transfer the caller Stay on the line until the called number answers Announce the call Press the switch hook or flash button adding the held party to the call Hang up to connect the two parties and transfer the call Note: You can hook flash while the 3rd party is ringing to start an early conference. Then hang up to complete the transfer without waiting for the 3rd party to answer first.
Expected Call and Network Behavior	When the user presses the switch hook or flash button, the transferee is placed on hold. When the user successfully dials the transfer number and the party answers the transferee can be

	added to the call by pressing the switch hook or flash button creating a three-way conference. When the user hangs up the phone the transferee and the called party remain in a call.
User Action Required to Deactivate or End	Not applicable.

5.2.9. Unattended or "Blind" Call Transfer

Service Description	Unattended or "Blind" Call Transfer lets a customer use their Touchtone phone to send a call to any other phone, inside or outside their business, including a wireless phones.
User Action Required to Activate or Use	While in a call with the party to be transferred: Press the switch hook or flash button on the phone to place the party on hold Enter *__ Dial the number to which you will transfer the caller The call is transferred when a complete number is entered. You will hear a short confirmation tone, followed by regular dial tone
Expected Call and Network Behavior	When the user presses the switch hook or flash button, the transferee is placed on hold. When the user successfully dials the transfer number, the transferee will automatically call the dialed number.
User Action Required to Deactivate or End	No applicable.

5.2.10. Call Hold

Service Description	Call Hold lets you put a caller on hold for an unlimited period of time. It is especially useful on phones without the hold button. Unlike a hold button, this feature provides access to a dial tone while the call is being held.
User Action Required to Activate or Use	Press the switch hook or flash button on the phone to place the first party on hold. You will hear a dial tone. To make another call: Enter the new number

	To return to call on hold: Hang up and the phone set will ring with the first call on the line (or Hook Flash again)
Expected Call and Network Behavior	
User Action Required to Deactivate or End	Hang-up the telephone.

5.2.11. Three-Way Calling

Service Description	The user can originate a call to a 3rd party while engaging in an active call.
User Action Required to Activate or Use	Press the switch hook or flash button on the phone to place the first party on hold Listen for three short tones followed by dial tone Dial the number of the 3rd party. When the 3rd party answers you may have a conversation with them while the other party is on hold. To hold a conference with the party on hold and the 3rd party, simply press the switch hook or flash button
Expected Call and Network Behavior	The SPA supports up to two calls per line. The SPA can conference two calls by bridging the 2nd and 3rd parties.
User Action Required to Deactivate or End	Hang-up the telephone.

5.2.12. Three-Way Ad-Hoc Conference Calling

Service Description	This feature allows the user to conference up to two other numbers on the same line to create a three-way call.
User Action Required to Activate or Use	If you are already on a call and wish to add a third party: Press the switch hook or flash button Listen for dial tone Dial the third party normally When the third party number starts to ring press the switch hook or flash button again You now have the original caller and the third

	party together with you on the same call. If you want to initiate a new Three Way Call: Call the first party in the normal manner Follow the directions for adding a third party (see instructions above)
Expected Call and Network Behavior	The SPA can host a 3-way conference and perform 3-way audio mixing (without the need of an external conference bridge device or service). If you also have Call Transfer you can also hang up at any time to transfer the original caller to the third party
User Action Required to Deactivate or End	

5.2.13. Call Return

Service Description	The SPA supports a service that allows the SPA to automatically dial the last caller's number.
User Action Required to Activate or Use	Pick up the receiver Listen for dial tone Press *__ to dial back the last caller that tried to reach you.
Expected Call and Network Behavior	This service gives the user the convenience of recalling the last incoming call to their number automatically.
User Action Required to Deactivate or End	No user action required

5.2.14. Automatic Call Back

Service Description	This feature allows the user to place a call to the last number they tried to reach whether the call was answered, unanswered or busy by dialing an activation code.
User Action Required to Activate or Use	Pick up the receiver Listen for dial tone Press *__
Expected Call and Network Behavior	If the number called is idle the call will ring through and complete normally. If the called

	number is busy the user will hear a special announcement and the feature will monitor the called number for up to 30 minutes. When both lines are idle, the user hears a special ring. During the monitoring process the user can continue to originate and receive calls without affecting the Call Return on Busy request. Call Return on Busy requests can be canceled by dialing the deactivation code.
User Action Required to Deactivate or End	Lift the receiver Listen for dial tone Press *__

5.2.15. Call FWD – Unconditional

Service Description	All calls are immediately forwarded to the designated forwarding number. The SPA will not ring or provide call waiting when Call FWD – Unconditional is activated.
User Action Required to Activate or Use	Lift the receiver Listen for dial tone Press *__ Listen for dial tone and enter the telephone number you are forwarding your call to. Activation will be confirmed with three short bursts of tone and your forwarding will be activated. Alternatively, the user can activate this feature from a web browser interface.
Expected Call and Network Behavior	This feature allows a user the option to divert (forward) all calls to their telephone number to any number using the touchtone keypad of their telephone or web browser interface. This service is activated or deactivated from the phone being forwarded or the web browser interface.
User Action Required to Deactivate or End	Lift the receiver Listen for dial tone Press *__ You will hear a confirmation tone signaling your change has been accepted. Alternatively, the user can deactivate this

	feature from a web browser interface.

5.2.16. Call FWD – Busy

Service Description	Calls are forwarded to the designated forwarding number if the subscriber's line is busy because of the following; Primary line already in a call, primary and secondary line in a call or conference.
User Action Required to Activate or Use	Lift the receiver Listen for dial tone Press *__ Listen for dial tone and enter the telephone number you are forwarding your call to. Activation will be confirmed with three short bursts of tone and your forwarding will be activated. Alternatively, the user can activate this feature from a web browser interface.
Expected Call and Network Behavior	This feature allows a user the option to divert (forward) calls to their telephone number to any number when their phone is busy or in conference by using the touchtone keypad of their telephone or web browser interface. This service is activated or deactivated from the phone being forwarded or the web browser interface.
User Action Required to Deactivate or End	Lift the receiver Listen for dial tone Press *__ You will hear a confirmation tone signaling your change has been accepted. Alternatively, the user can deactivate this feature from a web browser interface.

5.2.17. Call FWD - No Answer

Service Description	Calls are forwarded to the designated forwarding number after a configurable time period elapses while the SPA is ringing and does not answer.

User Action Required to Activate or Use	Lift the receiver Listen for dial tone Press *__ Listen for dial tone and enter the telephone number you are forwarding your call to. Activation will be confirmed with three short bursts of tone and your forwarding will be activated. Alternatively, the user can activate this feature from a web browser interface. Note: The forward delay is entered from the web interface. Default is 20s
Expected Call and Network Behavior	This feature allows a user the option to divert (forward) calls to their telephone number to any other dialable number when their phone is not answered by using the touchtone keypad of their telephone or web browser interface. This service is activated or deactivated from the phone being forwarded or the web browser interface.
User Action Required to Deactivate or End	Lift the receiver Listen for dial tone Press *__ You will hear a confirmation tone signaling your change has been accepted. Alternatively, the user can deactivate this feature from a web browser interface.

5.2.18. Anonymous Call Blocking

Service Description	By setting the corresponding configuration parameter on the SPA, the subscriber has the option to block incoming calls that do not reveal the caller's Caller ID.
User Action Required to Activate or Use	Pick up the receiver Listen for dial tone To Activate Press *__
Expected Call and Network Behavior	When activated by the user, callers will hear (busy) tone.
User Action Required to Deactivate or End	To De-activate Press *__

5.2.19. Distinctive / Priority Ringing and Call Waiting Tone

Service Description	The SPA supports a number of ringing and call waiting tone patterns to be played when incoming calls arrive. The choice of alerting pattern to use is carried in the incoming SIP INVITE message inserted by the SIP Proxy Server (or other intermediate application server in the Service Provider's domain).
User Action Required to Activate or Use	Pick up the receiver Listen for dial tone Press *__
Expected Call and Network Behavior	With this service, incoming calls from up to __ telephone numbers can be automatically identified by distinctive ringing. A distinctive ringing pattern (i.e. short-long-short) accompanies incoming calls from the designated telephone numbers. If the user is engaged in conversation and a call from one of the designated numbers arrives, a distinctive call waiting tone (i.e. short-long-short) accompanies the incoming call. Calls from other telephone numbers ring normally.
User Action Required to Deactivate or End	

5.2.20. Speed Calling – Up to Eight (8) Numbers or IP Addresses

Service Description	The SPA supports user programming of up to 8 long distance, local, international or emergency numbers and/or IP addresses for fast and easy access.
User Action Required to Activate or Use	Pick up the receiver Listen for dial tone Press *__ Dial the single digit code under which the number is to be stored (2-9) Dial the complete number to be stored just as if you were going to dial it yourself Listen for Confirmation tone (two short beeps) Hang up or repeat the sequence Note: To enter IP addresses, a graphical user

	interface like a web browser must be used.
Expected Call and Network Behavior	Pick up the receiver Listen for dial tone Press single digit code assigned to the stored number (2-9) Press # to signal dialing complete The number is automatically dialed normally.
User Action Required to Deactivate or End	None

6. Troubleshooting

6.1. Symptoms and Corrections

6.2. Error and Log Reporting

The SPA Error Status Code (ESC) is used to indicate the current operation status of the SPA unit. An error state can be a relatively long transient state or a steady state. The state is also represented by a special blinking pattern of the Status LED (next to the RJ-11 ports). The Error Status Code is a 4 digit number. The first digit indicates the error class: 1xxx represents normal operation states while 2xxx – 9xxx represent error states that must be fixed for the unit to function properly. The status code values can be read from the IVR option XXX or from the SPA web-page.

6.2.1. LED Blink Rate Definitions

ON – LED remains solid on

OFF – LED remains solid off

LONG (Long On) – 3.0s on, 1s off continuously

FAST – 0.125s on, 0.125s off continuously

SLOW – 0.5s on, 0.5s off continuously

VSLO (Very Slow) – 1.0s on, 1.0s off continuously

HB (Heart Beat) – 0.1s on, 0.1s off, 0.1s on, 1s off continuously

Note: The Link LED will blink on transmit and receive (TX/RX) of packets. The LED will display solid off if no link is available. The LED will display solid on if link is up but no TX/RX activity is present.

Error Status Code	Status/Error Description	LED Blink Rate	Handset Behavior
0000	System Initializing	SLOW	Silent
1000	Normal Operation – Both Lines on hook	OFF	Normal
1001	Normal Operation – Either line off hook	ON	Normal
1100[3]	Downloading new firmware	SLOW	Silent
1101	Writing firmware to flash	FAST	Silent
2000	Looking for DHCP Server	SLOW	Silent
2001	SPA is using last known good IP (backup IP)	ERR1	Normal
2002	IP Address Confilict	ERR2	Silent

2999	No DHCP Server	SLOW	Silent
2999	Unknown DHCP Error	SLOW	Silent
5000	No Ethernet Link Detected Error	ERR0	Silent

7. Feature Descriptions

The SPA is a full featured, fully programmable phone adapter that can be custom provisioned within a wide range of configuration parameters. The below feature descriptions are written as a high-level overview to provide a basic understanding of the feature breadth and capabilities of the SPA. To understand the specific implementation of the below features, including parameters, requirements and contingencies please refer the section SPA Feature Configuration Parameters, section 0.

7.1. Data Networking Features

7.1.1. MAC Address (IEEE 802.3)

7.1.2. IPv4 – Internet Protocol Version 4 (RFC 791) upgradeable to v6 (RFC 1883)

7.1.3. ARP – Address Resolution Protocol

7.1.4. DNS – A Record (RFC 1706), SRV Record (RFC 2782)

7.1.5. DiffServ (RFC 2475) and ToS – Type of Service (RFC 791/1349)

7.1.6. DHCP Client – Dynamic Host Configuration Protocol (RFC 2131)

7.1.7. ICMP – Internet Control Message Protocol (RFC792)

7.1.8. TCP – Transmission Control Protocol (RFC793)

7.1.9. UDP – User Datagram Protocol (RFC768)

7.1.10. RTP – Real Time Protocol (RFC 1889) (RFC 1890)

7.1.11. RTCP – Real Time Control Protocol (RFC 1889)

7.2. Voice Features

7.2.1. SIPv2 – Session Initiation Protocol Version 2 (RFC 3261-3265)

7.2.1.1. SIP Proxy Redundancy – Static or Dynamic via DNS SRV

In typical commercial IP Telephony deployments, all calls are established through a SIP proxy server. An average SIP proxy server may handle tens of thousands subscribers. It is important that a backup server is available so that an active server can be temporarily switched out for maintenance. The SPA supports the use of backup SIP proxy servers so that service disruption should be next to non-existent.

Static Redundancy:

A relatively simple way to support proxy redundancy is to configure a static list of SIP proxy servers to the SPA in its configuration profile where the list is arranged in some order of priority. The SPA will attempt to contact the highest priority proxy server whenever possible. When the currently selected proxy server is not responding, the SPA automatically retries the next proxy server in the list.

Dynamic Redundancy:

The dynamic nature of SIP message routing makes the use of a static list of proxy servers inadequate in some scenarios. In deployments where user agents are served by different domains, for instance, it would not be feasible to configure one static list of proxy servers per covered domain into an SPA. One solution to this situation is through the use DNS SRV records. The SPA can be instructed to contact a SIP proxy server in a domain named in SIP messages. The SPA shall consult the DNS server to get a list of hosts in the given domain that provides SIP services. If an entry exists, the DNS server will return a SRV record which contains a list of SIP proxy servers for the domain, with their host names, priority, listening ports, etc. The SPA shall try to contact the list of hosts in the order of their stated priority.

7.2.1.2. Re-registration with Primary SIP Proxy Server

If the SPA is currently using a lower priority proxy server, it should periodically probe the higher priority proxy to see if it is back on line and attempt to switch back to the higher priority proxy whenever possible. It is very important that switching proxy server should not affect calls that are already in progress.

7.2.1.3. SIP Support in Network Address Translation Networks – NAT

7.2.2. Codec Name Assignment

Negotiation of the optimal voice codec is sometimes dependent on the SPA device's ability to "match" a codec name with the far-end device/gateway codec name. The SPA allows the network administrator to individually name the various codecs that are supported such that the correct codec successfully negotiates with the far end the equipment.

7.2.3. Secure Calls

A user (if enabled by service provider or administrator) has the option to make an outbound call secure in the sense that the audio packets in both directions are encrypted.

7.2.4. Voice Algorithms:

7.2.4.1. G.711 (A-law and mµ-law)

This very low complexity codec supports uncompressed 64 kbps digitized voice transmission at one through ten 5 ms voice frames per packet. This codec provides the highest voice quality and uses the most bandwidth of any of the available codecs.

7.2.4.2. G.726

This low complexity codec supports compressed 16, 24, 32 and 40 kbps digitized voice transmission at one through ten 10 ms voice frames per packet. This codec provides the high voice quality.

7.2.4.3. G.729A

The ITU G.729 voice coding algorithm is used to compress digitized speech. Sipura supports G.729. G.729A is a reduced complexity version of G.729. It requires about half the processing power to code G.729. The G.729 and G.729A bit streams are compatible and interoperable, but not identical.

7.2.4.4. G.723.1

The SPA supports the use of ITU G.723.1 audio codec at 6.4 kbps. Up to 2 channels of G.723.1 can be used simultaneously. For example, Line 1 and Line 2 can be using G.723.1 simultaneously, or Line 1 or Line 2 can initiate a 3-way conference with both call legs using G.723.1.

7.2.5. Codec Selection

The administrator can select which low-bit-rate codec to be used for each line. G711a and G711u are always enabled.

7.2.6. Dynamic Payload

When no static payload value is assigned per RFC 1890, the SPA can support dynamic payloads for G.726.

7.2.7. Adjustable Audio Frames Per Packet

This feature allows the user to set the number of audio frames contained in one RTP packet. Packets can be adjusted to contain from 1 – 10 audio frames. Increasing the number of packets decreases the bandwidth utilized – but it also increases delay and may affect voice quality.

7.2.8. Fax Tone Detection Pass-Through

Users can connect a fax terminal to the SPA telephone port(s). Fax terminals transmit a single tone when they answer a call. The SPA detects the type of equipment in use on the basis of its answer tone. When it detects the equipment answering the call, the SPA performs a switchover from the current audio codec to G.711 codec.

7.2.9. DTMF: In-band & Out-of-Band (RFC 2833) (SIP INFO *)

The SPA may relay DTMF digits as out-of-band events to preserve the fidelity of the digits. This can enhance the reliability of DTMF transmission required by many IVR applications such as dial-up banking and airline information.

7.2.10. Call Progress Tone Generation

The SPA has configurable call progress tones. Parameters for each type of tone may include number of frequency components, frequency and amplitude of each component, and cadence information.

7.2.11. Call Progress Tone Pass Through

This feature allows the user to hear the call progress tones (such as ringing) that are generated from the far-end network.

7.2.12. Jitter Buffer – Dynamic (Adaptive)

The SPA can buffer incoming voice packets to minimize out-of-order packet arrival. This process is known as jitter buffering. The Jitter Buffer size will proactively adjust or adapt in size depending on changing network conditions.

The SPA has a Network Jitter Level control setting for each line of service. The jitter level decides how aggressively the SPA will try to shrink the jitter buffer over time to achieve a lower overall delay. If the jitter level is higher, it shrinks more gradually. If jitter level is lower, it shrinks more quickly.

7.2.13. Full Duplex Audio

Full-duplex is the ability to communicate in two directions simultaneously so that more than one person can speak at a time. Half-duplex means that only one person can talk at a time – like a CB radio or walkie-talkie, which is unnatural in normal free-flowing two-way communications. The SPA supports full-duplex audio.

7.2.14. Echo Cancellation – Up to 8 ms Echo Tail

The SPA supports hybrid line echo cancellation. This feature uses the G.165 echo canceller to eliminate up to 8 ms of line echo. This feature does not provide acoustic echo cancellation on endpoint devices – that is, an end user's speakerphone.

7.2.15. Voice Activity Detection with Silence Suppression & Comfort Noise Generation

Voice Activity Detection (VAD) and Silence Suppression is a means of increasing the number of calls supported by the network by reducing the required bi-directional bandwidth for a single call. VAD uses a very sophisticated algorithm to distinguish between speech and non-speech signals. Based upon the current and past statistics, the VAD algorithm decides whether or not speech is present. If the VAD algorithm decides speech is not present, the silence suppression and comfort noise generation is activated. This is accomplished by removing and not transmitting the natural silence that occurs in normal 2-way connection – the IP bandwidth is used only when someone is speaking. During the silent periods of a telephone call additional bandwidth is available for other voice calls or data traffic since the silence packets are not being transmitted across the network. Comfort Noise Generation provides artificially generated background white noise (sounds), designed to reassure callers that their calls are still connected during silent periods. If Comfort Noise Generation is not used, the caller may think the call has been disconnected because of the "dead silence" periods created by the VAD and Silence Suppression feature.

7.2.16. Attenuation / Gain Adjustment

7.2.17. Signaling Hook Flash Event

The SPA can signal hook flash events to the remote party on a connected call. This feature can be used to provide advanced mid-call services with third-party-call-control. Depending on the features that the service provider will offer using third-party-call-control, the following three SPA features may be disabled to correctly signal a hook-flash event to the softswitch:

1. Call Waiting Service
2. Three Way Call Service
3. Three Way Conf Service

7.2.18. Configurable Flash / Switch Hook Timer

7.2.19. Configurable Dial Plan with Interdigit Timers

The SPA has three configurable interdigit timers:

- Initial timeout (T) = handset off hook, no digit pressed yet.
- Long timeout (L) = one or more digits pressed, more digits needed to reach a valid number (as per the dial plan).
- Short timeout (S) = current dialed number is valid, but more digits would also lead to a valid number.

7.2.20. Message Waiting Indicator Tones – MWI

7.2.21. Polarity Control

The SPA allows the polarity to be set when a call is connected and when a call is disconnected. This feature is required to support some pay phone system and answering machines.

7.2.22. Calling Party Control – CPC

CPC signals to the called party equipment that the calling party has hung up during a connected call by removing the voltage between the tip and ring momentarily. This feature is useful for auto-answer equipment which then knows when to disengage.

7.2.23. International Caller ID Delivery

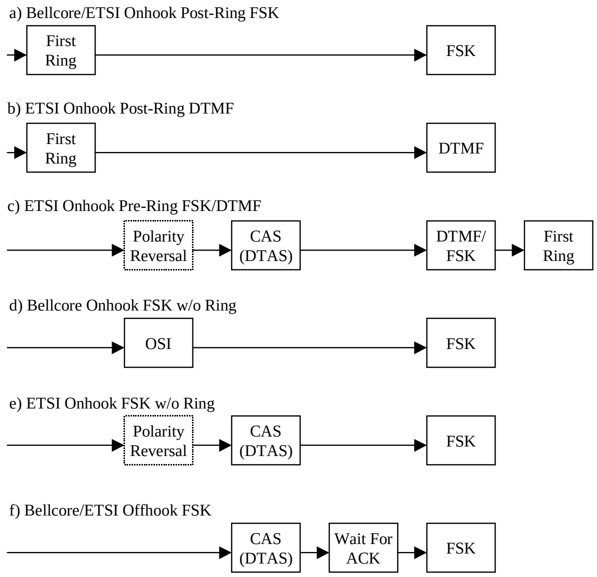
In addition to support of the Bellcore (FSK) and Swedish/Danish (DTMF) methods of Caller ID (CID) delivery, release 2.0 adds a large subset of ETSI compliant methods to support international CID equipment. The figure below shows the CID/CIDCW architecture used in the SPA. Different flavors of CID delivery method can be obtained by mixing-and-matching some of the steps as shown.

It should be noted that the choice of CID method will affect the following features:

• On Hook Caller ID Associated with Ringing – This type of Caller ID is used for incoming calls when the attached phone is on hook (see Figure 1 (a) – (c). All SPA CID methods can be applied for this type of caller-id

• On Hook Caller ID Not Associated with Ringing – In the SPA-2000 this feature is used for send VMWI signal to the phone to turn the message waiting light on and off (see Figure 1 (d) and (e)). This is available only for FSK-based caller-id methods: "Bellcore", "ETSI FSK", and "ETSI FSK With PR"

• Off Hook Caller ID – This is used to delivery caller-id on incoming calls when the attached phone is off hook (see Figure 1 (f)). This can be call waiting caller ID (CIDCW) or to notify the user that the far end party identity has changed or updated (such as due to a call transfer). This is only available if the caller-id method is one of “Bellcore”, “ETSI FSK”, or “ETSI FSK With PR”.



SPA Caller ID Delivery Architecture

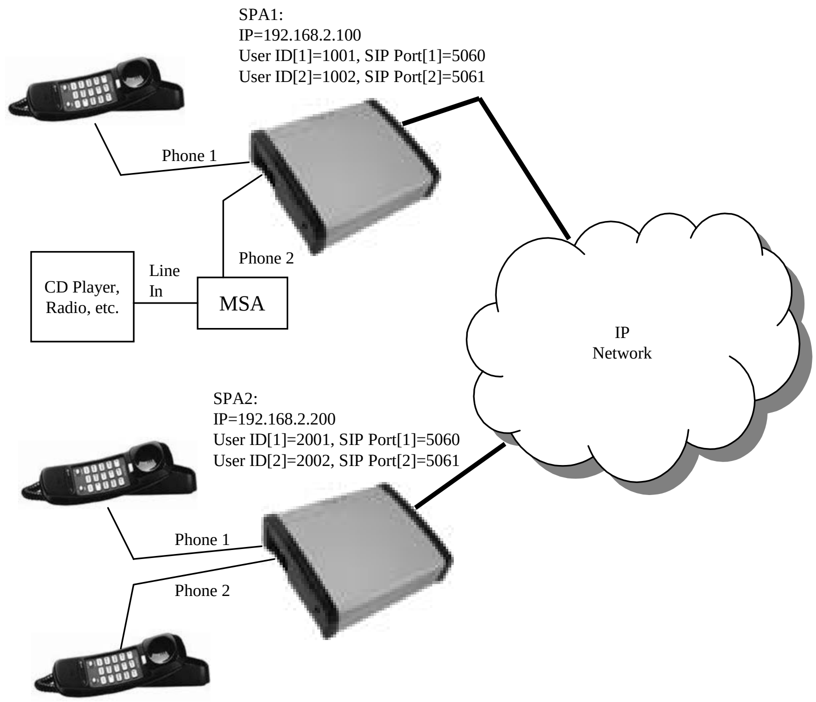
7.2.24. Streaming Audio Server – SAS

This feature allows one to attach an audio source to one of the SPA FXS ports and use it as a streaming audio source device. The corresponding Line (1 or 2) can be configured as a streaming audio server (SAS) such that when the Line is called, the SPA answers the call automatically and starts streaming audio to the calling party provided the FXS port is off-hook. If the FXS port is on-hook when the incoming call arrives, the SPA replies with a SIP 503 response code to indicate “Service Not Available.” If an incoming call is auto-answered, but later the FXS port becomes on-hook, the SPA does not terminate the call but continues to stream silence packets to the caller. If an incoming call arrives when the SAS line has reached full capacity, the SPA replies with a SIP 486 response code to indicate “Busy Here”.

The SAS line can be setup to refresh each streaming audio session periodically (via SIP re-INVITE) to detect if the connection to the caller is down. If the caller does not respond to the refresh message, the SAS line will terminate the call so that the streaming resource can be used for other callers.

7.2.25. Music On Hold – MOH

On a connected call, the SPA may place the remote party on call (the only way to do this on te SPA-2000 is to perform a hook-flash to initiate a 3-way call or to swap 2 calls during call-waiting). If the remote party indicates that they can still receive audio while the call is holding, the SPA-2000 can be setup to contact an auto-answering SAS as described in Section 4 and have it stream audio to the holding party. When used this way, the SAS is referred to as a MOH Server.



Example configuration for MOH application with a SPA line configured as a SAS

SAS Configuration Examples:

The following configuration examples are based on the setup as depicted in Figure.

Example 1: SAS Line not registered with the Proxy Server for the other subscribers

On SPA 1:

SAS Enable[1] = no

MOH Server [1] = 1002@192.168.2.100:5061 or 1002@127.0.0.1:5061

SAS Enable[2] = yes

On SPA 2:

SAS Enable[1] = no

MOH Server [1] = 1002@192.168.2.100:5061

SAS Enable[2] = no

MOH Server [2] = 1002@192.168.2.100:5061

Example 2: SAS Line registered with the Proxy Server as the other subscribers

On SPA 1:

SAS Enable[1] = no

MOH Server [1] = 1002

SAS Enable[2] = yes

On SPA 2:

SAS Enable[1] = no

MOH Server [1] = 1002

SAS Enable[2] = no

MOH Server [2] = 1002

7.3. Security Features

7.3.1. Multiple Administration Layers (Levels and Permissions)

7.3.2. HTTP Digest – Encrypted Authentication via MD5 (RFC 1321)

7.3.3. HTTPS with Client Certificate

7.4. Administration and Maintenance Features

7.4.1. Web Browser Administration and Configuration via Integral Web Server

7.4.2. Telephone Key Pad Configuration with Interactive Voice Prompts

7.4.3. Automated Provisioning & Upgrade via TFTP, HTTP and HTTPS

7.4.4. Periodic Notification of Upgrade Availability via NOTIFY or HTTP

7.4.5. Non-Intrusive, In-Service Upgrades

7.4.6. Report Generation and Event Logging

The SPA reports a variety of status and error reports to assist service providers to diagnose problems and evaluate the performance of their services. The information can be queried by an authorized agent (using HTTP with digested authentication, for instance). The information may be organized as an XML page or HTML page.

7.4.7. Syslog and Debug Server Records

The SPA supports detailed logging of all activities for further debugging. The debug information may be sent to a configured Syslog server. Via the configuration parameters, the SPA allows some settings to select which type of activity/events should be logged – for instance, a debug level setting.

8. Acronyms

A/D	Analog To Digital Converter
ANC	Anonymous Call
B2BUA	Back to Back User Agent
Bool	Boolean Values. Specified as “yes” and “no”, or “1” and “0” in the profile
CA	Certificate Authority
CAS	CPE Alert Signal
CDR	Call Detail Record
CID	Caller ID
CIDCW	Call Waiting Caller ID
CNG	Comfort Noise Generation
CPC	Calling Party Control
CPE	Customer Premises Equipment
CWCID	Call Waiting Caller ID

CWT	Call Waiting Tone
D/A	Digital to Analog Converter
dB	decibel
dBm	dB with respect to 1 milliwatt
DHCP	Dynamic Host Configuration Protocol
DNS	Domain Name Server
DRAM	Dynamic Random Access Memory
DSL	Digital Subscriber Loop
DSP	Digital Signal Processor
DTAS	Data Terminal Alert Signal (same as CAS)
DTMF	Dual Tone Multiple Frequency
ETSI	European ???
FQDN	Fully Qualified Domain Name
FSK	Frequency Shift Keying
FXS	Foreign eXchange Station
GW	Gateway
ITU	International Telecommunication Union
HTML	Hypertext Markup Language
HTTP	Hypertext Transfer Protocol
HTTPS	HTTP over SSL
ICMP	Internet Control Message Protocol
IGMP	Internet Group Management Protocol
ILEC	Incumbent Local Exchange Carrier
IP	Internet Protocol
ISP	Internet Service Provider
ITSP	IP Telephony Service Provider
IVR	Interactive Voice Response
LAN	Local Area Network
LBR	Low Bit Rate
LBRC	Low Bit Rate Codec
MC	Mini-Certificate
MGCP	Media Gateway Control Protocol
MOH	Music On Hold
MOS	Mean Opinion Score (1-5, the higher the better)
ms	Millisecond
MSA	Music Source Adaptor
MWI	Message Waiting Indication
OSI	Open Switching Interval
PCB	Printed Circuit Board
PR	Polarity Reversal
PS	Provisioning Server
PSQM	Perceptual Speech Quality Measurement (1-5, the lower the better)
PSTN	Public Switched Telephone Network
NAT	Network Address Translation
OOB	Out-of-band
REQT	(SIP) Request Message
RESP	(SIP) Response Message
RSC	(SIP) Response Status Code, such as 404, 302, 600
RTP	Real Time Protocol
RTT	Round Trip Time
SAS	Streaming Audio Server
SDP	Session Description Protocol
SDRAM	Synchronous DRAM
sec	seconds

SIP	Session Initiation Protocol
SLIC	Subscriber Line Interface Circuit
SP	Service Provider
SPA	Sipura Phone Adaptor
SSL	Secure Socket Layer
TFTP	Trivial File Transfer Protocol
TCP	Transmission Control Protocol
UA	User Agent
uC	Micro-controller
UDP	User Datagram Protocol
URL	Uniform Resource Locator
VM	Voice Mail
VMWI	Visual Message Waiting Indication/Indicator
VQ	Voice Quality
WAN	Wide Area Network
XML	Extensible Markup Language

9. Glossary

ACD (Automatic Call Distribution): A switching system designed to allocate incoming calls to certain positions or agents in the order received and to hold calls not ready to be handled (often with a recorded announcement).

Area Code: A 3-digit code used in North America to identify a specific geographic telephone location. The first digit can be any number between 2 and 9. The second and third digits can be any number.

Billing Increment: The division by which the call is rounded. In the field it is common to see full-minute billing on the local invoice while 6-second rounding is the choice of most long-distance providers that bill their customers directly.

Blocked Calls: Caused by an insufficient network facility that does not have enough lines to allow calls to reach a given destination. May also pertain to a call from an originating number that is blocked by the receiving telephone number.

Bundled Service: Offering various services as a complete package.

Call Completion: The point at which a dialed number is answered.

Call Termination: The point at which a call is disconnected.

CDR (Call Detail Records): A software program attached to a VoIP/telephone system that records information about the telephone number's activity.

Carrier's Carrier: Companies that build fiber optic and microwave networks primarily selling to resellers and carriers. Their main focus is on the wholesale and not the retail market.

Casual Access: Casual Access is when customers choose not to use their primary carriers to process the long-distance call being made. The customer dials the carrier's 101XXXX number.

CO (Central Office): Switching center for the local exchange carrier.

Centrex: This service is offered by the LEC to the end user. The feature-rich Centrex line offers the same features and benefits as a PBX to a customer without the capital investment or maintenance charges. The LEC charges a monthly fee to the customer, who must agree to sign a term agreement.

Circuits: The communication path(s) that carry calls between two points on a network.

Customer Premise Equipment: The only part of the telecommunications system that the customer comes into direct contact with. Example of such pieces of equipment are: telephones, key systems, PBXs, voicemail systems and call accounting systems as well as wiring telephone jacks. The standard for this equipment is set by the FCC, and the equipment is supplied by an interconnect company.

Dedicated Access: Customers have direct access to the long-distance provider via a special circuit (T1 or private lines). The circuit is hardwired from the customer site to the POP and does not pass through the LEC switch. The dial tone is provided from the long-distance carrier.

Dedicated Access Line (DAL): Provided by the local exchange carrier. An access line from the customer's telephone equipment directly to the long-distance company's switch or POP.

Demarcation Point: This is where the LEC's ownership and responsibility (wiring, equipment) ends and the customer's responsibilities begin.

Direct Inward Dialing (DID): Allows an incoming call to bypass the attendant and ring directly to an extension. Available on most PBX systems and a feature of Centrex service.

Dual Tone Multifrequency (DTMF): Better known as the push button keypad. DTMF replaces dial pulses with electronically produced tones for network signaling.

Enhanced Service: Services that are provided in addition to basic long distance and accessed by way of a touchtone phone through a series of menus.

Exchange Code (NXX): The first three digits of a phone number.

Flat-rate Pricing: The customer is charged one rate (sometimes two rates, one for peak and one for off-peak) rather than a mileage-sensitive program rate.

IXC (Interexchange Carrier): A long-distance provider that maintains its own switching equipment.

IVR (Interactive Voice Response): Provides mechanism for information to be stored and retrieved using voice and a touchtone telephone.

Local Loop: The local telephone company provides the transmission facility from the customer to the telephone company's office, which is engineered to carry voice and/or data.

North American Numbering Plan (NANP): How we identify telephone numbers in North America. We can identify the telephone number based on their three separate components (NPA) (NXX) (XXXX).

PIN (Personal Identification Code): A customer calling/billing code for prepaid and pay-as-you-go calling cards.

Private Branch Exchange: Advanced phone system commonly used by the medium to larger customer. It allows the customer to perform a variety of in-house routing (inside calling). The dial tone that is heard when the customer picks up the phone is an internal dial tone.

SS7 (System Signaling Number 7): Technology used by large carriers to increase the reliability and speed of transmission between switches.

Switch (Switching): Equipment that connects and routes calls and provides other interim functions such as least cost routing, IVR, and voicemail. It performs the "traffic cop" function of telecommunications via automated management decisions.

Touchtone (DTMF): The tone recognized by a push button (touchtone) telephone.

Unified Messaging: Platform that lets users send, receive, and manage all email, voice, and fax messages from any telephone, PC, or information device.

Voice Mail: A system that allows storage and retrieval of voice messages through voicemail boxes.

[i] "Carrier Grade Voice Over IP", 2nd Edition, Daniel Collins, McGraw-Hill, 2003