Sipura Technology, Inc.

SPA-841 Administration Guide

January 2005

Version 0.9 (DRAFT)

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Table of Contents

1 How to use this document

This Administration Guide provides valuable instructions on managing and operating the SPA-841 IP Telephone. The Guide covers the following topics:

- General Operating Information
- Call, Voice, Signaling Features and Hardware Functionality Descriptions
- Configuration, Performance and Troubleshooting Guidelines
- Provisioning

The SPA-841 User Guide provides more details on how to use the phone's features via the phone's physical interfaces i.e. LED, keys and buttons. The Administrative Guide is very helpful for network administrators that need detailed information on how to configure the phone from the phone's Web Server or are need access to the phone's troubleshooting and performance functionality. Finally, the document also covers provisioning of the SPA-841.

2 Important Operating Information

Please insure the safe operation of the SPA-841 by following the operating instructions in this section.

Compliance and Safety Information

This equipment has been tested and found to comply with the limits for a Class B digital device in accordance with the specifications in part 15 of the FCC rules. This product bears the CE Marking indicating compliance with the 89/336/EEC directive. Standards to which conformity is Declared: EN 61000-4-2:1995, EN 61000-4-3:1997, EN 61000-4-4:1995, EN 61000-4-5:1995, EN 61000-4-6:1996, EN 61000-4-8:1994, EN 61000-4-11:1994, EN 61000-3-2:2001, EN 61000-3-3:1995 & EN 55022:1998 Class B Modifications to this product not authorized by Sipura Technology could void FCC approval, terminating end user authority to use this product.

For indoor use only.
Read installation instructions before connecting to a power source. The electric plug and socket must be accessible at all times as this is the main method to disconnect power from the device.

Shock Hazard:
  Do not operate near water or similar fluid.
  Do not work with this device during periods of lightning activity.
  Do not touch wires at the end of cables or within sockets.

One Year Limited Hardware Warranty
Sipura Technology provides a one (1) year limited hardware warranty. Sipura warrants to customer that this product will conform to its published specifications and will be free from defects in material and workmanship at the time of delivery and for a period of one year thereafter. Without limiting the foregoing, this warranty does not cover any defect resulting from (i) any design or specification supplied by an entity other than Sipura, (ii) non-observance of technical operating parameters (e.g., exceeding limiting values), or (iii) misuse, abuse, abnormal conditions or alteration by anyone other than Sipura.

Replacement, Repair, Refund:
After the receipt of an RMA (Return Materials Authorization) request, Sipura will attempt to refund, repair or replace the SPA. To receive an RMA number contact the party from whom you purchased the SPA.

Network and Service Configuration Settings

The service provider or network administrator who supplied this unit may have provided a pre-configuration of the network and service settings. Depending on the configuration policy, end user access to the local configuration settings may be restricted or inaccessible. If end user configuration is allowed, some of the key network and service configuration setting instructions are describe in the following sections.

3   SPA-841 Overview

The SPA-841 is a SIP based, IP telephone offered by Sipura. The SPA-841 supports up to 4 lines of operation and a pixel-based display. Figure 1.1 shows an illustration of the phone.

Figure 1.1: SPA-841: Sipura IP Phone

SPA-841 Hardware Features:

- 128x48 pixel-based monochrome LCD display
- Four Line Keys. Each Line KEY has a corresponding red/green/amber tricolor LED
- Built-in Microphone and Speaker
- External Microphone and Headset
- Speaker phone on/off button with tricolor LED
- Headset on/off button with tricolor LED
- Handset
- Mute on/off button with tricolor LED
- Hold key
- Up/Down Rocker Key (called Scroll buttons in SPA-841 User Guide)
- Soft Keys and a "i" button
- Volume Key to invoke volume adjustment function
- VM Key to access voice mail
- MWI LED
- Standard 12-Key dialing pad
- Setup Key to invoke the setup menu
- 1 RJ45 for 10BaseT Ethernet Uplink
- 5V dc universal power adaptor

SPA-841 Functionalities

The SPA-841 has the following functionalities:

- Four Call Appearances
- Basic outgoing and incoming Calls
- Call Hold, Call Waiting, Call Transfer, and Conference
- Block Anonymous Calls
- Do Not Disturb
- Call forward unconditional, no answer, and on busy
- Multiple ring tones and default ring tone assignment per Extension
- Distinctive ringing
- Call Logs (Outgoing/Redial, Incoming, Missed)
- Call timer per call
- Personal Directory with 1-digit speed dial
- Caller-ID Name and Number
- Outbound Caller-ID Blocking
- Repeat Dialing on Busy
- Secure RTP
- Provisioning using TFTP/HTTP/HTTPS protocol
- DNS SRV and Multiple A Records for Proxy Lookup and Redundancy (with up to 25 hosts)
- Auto-Dialing From a Directory when off-hook
- On-Hook Dialing
- Speaker phone
- Share Extension and Share Call Appearance[1]
- Configurable Dial Plans with Auto-Completion of dialed number

Navigating the SPA-841 GUI

Note: Although this guide provides an overview of how to navigate the SPA-841 GUI, its primary focus is administrative functions available via the phone's web interface. For more detailed user functionality on the phone's GUI, please refer to the SPA-841 User Guide.

The user can invoke the phone's GUI menu at various times, most notably when pressing the **i** button when trying to view or change some of the phone settings. User can navigate through a menu by using the UP/DOWN keys, and then press one of the soft key choices to select the highlighted item. Pressing the right SoftKey i.e. SK(cancel), quits the current menu and return to the last menu level. Note that pressing the UP key at the 1st item will move the highlight to the last item; pressing the DOWN key at the last item will move the highlight to the first item.

Each phone screen normally has 1 or more virtual soft keys. Where there are more than 2 virtual soft keys, the user can scroll through the virtual soft keys by pressing the "i" button. Pressing "i" button once shows the next 2 virtual soft keys in a round robin fashion. The SPA-841 indicates more virtual soft keys available by showing 2 extra lines on top of the right soft key label (like a deck of cards).

Each item on a menu has a corresponding number on the left hand side. The item number usually starts at 1,2,3,… up to total number of items on the menu. Suppose you have 12 items on the menu and the first item is numbered as 1. You can select the 12th item by pressing the digits 1, 2 with less than 2 seconds of pause between the two digits. You can select the 2nd item by pressing the digit 2 and the selection will be made 2 seconds after you press the digit. You can select the 4th item by pressing the digit 4 and the selection will be made right after you press the digit.

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[1] *Based on Broadsoft specification, "Document number or name"…*

The user can also make a choice on a menu by pressing the corresponding numeric value appears to the left of the menu item. After entering one or more digits and if the number of items in the menu is such that another additional digit may still result in a valid choice, the SPA-841 will wait for the next digit for 2 seconds before processing the selection. Note that the SPA-841 will ignore any leading 0 digit entries. If the user press a digit sequence that is outside the range, the SPA-841 will simply position the highlight to the first or the last item on the menu. Depending on the menu, selecting a menu item by pressing the digit keys may automatically perform the default action on the item, or simply repositioning the highlight on that item (i.e. no default action). For example, on the Speed Dial or Directory menu, the default action is to edit the selected item as if the item is selected by pressing SK(edit). On the other hand, there is no default action on the Redial List or Missed Calls menu.

When a menu is shown on the screen, the status line (between 2 soft key labels) will show the title of the menu. If the menu is invoke as a submenu under the *i* button, the *i* symbol will be prepended to the menu title. This indicates that the user can exit all the way from the setup menu by pressing the *i* button again.

A menu item may have contents that are wider than the LCD can show. When the user highlight the item, the SPA-841 will automatically scroll the contents horizontally from right to left so that the user can view the entire contents.

Definitions for the Four Call Appearances

Some definitions:
- Station: A SPA-841 phone (or similar device) with one or more Extensions and Call appearances provisioned
- Ext: An extension in the SPA-841 is a VoIP account in a voice service provider (VSP)'s network or an IP-PBX system. An extension can be uniquely identified with a User ID (like a phone number) that is unique within the VSP. Up to 4 extensions can be configured in the SPA-841, which we will refer to as Ext 1, Ext 2, Ext 3, and Ext 4 (or simply E1, E2, E3, and E4 respectively). Note that the same extension can be configured on more than one station. These extensions are called *Shared Extensions*. Extension 1 is referred to as the Primary Extension. Some features can only be activated on the Primary Extension such as Call Forwarding and VMWI.
- Call Appearance: Physically a call appearance corresponds to a Line Key on a station. There are 4 Line Keys in SPA-841, which we refer to as Line Keys 1, 2, 3, and 4 (or, simply, L1, L2, L3, and L4 respectively). Functionally a call appearance is an instance of an extension. We call it a *Share Call Appearance* if the associated extension is shared. Multiple instances of the same extension can be configured on SPA-841. In fact all four call appearances can be instances of the same extension. Note that call appearances (on the same station) which are instances of the same extension are shared call appearances if and only if the extension is. Any of the 4 Line Keys can be disabled in the configuration. Each call appearance supports one call at a time.
- VoIP Interface (VI): An Extension and its associated control parameters configured on a particular station. The SPA-841 includes a rich set of configuration parameters to control the operation when calling via the account. When configuring an extension on the SPA-841, we are configuring the core account information as well as the set of VI control parameters. While the account information is usually the same for a shared extension on different stations, the rest of the VI parameters can be very different (such as the dial plan or the preferred codec to use when making a call on this extension could be different for 2 different stations sharing the extension).
- Call Appearance State (CAST): The state of a call appearance which can be one of the following:
  - Disabled: The Line Key is disabled
  - Idle (Ready): The Call Appearance is ready for use

- o Dialing: Collecting digits from the user to be dialed out from this Line
- o Calling: Waiting for the called party to respond
- o Proceeding (a.k.a. Progressing): Called party is ringing
- o Ringing (a.k.a. Alerting): Incoming call ringing
- o Connected: Connected with remote party
- o Held: Remote party is on hold
- o Invalid: Remote party hangs up or error while attempting outbound call
- o Busy: the line is being used by another station (shared line only)
- A Call in a certain state by preceding the term Call with the corresponding state name. For example, a Ringing Call, a Dialing Call
- Active (Call) State: If the state is Dialing, Calling, Proceeding, Connected, or Invalid. When the state of the Call is Active, we refer to it as an Active Call
- Standby (Call) State: If the state is Ringing or Holding. When the state of the Call is Standby, we refer to it as a Standby Call
- Key: Any one of the keys on the SPA-841 keypad. A Key can have 2 states: down (when pressed) or up (when released)
- Button – a Key with an on and off state. The on/off state toggles when the key is pressed down where applicable. There are 3 buttons on the SPA-841: SPEAKER, HEADSET, and MUTE. Each button has an associated LED that indicates the on/off state.
- SK(*label*) – Soft key with the given label, such as SK(select), SK(cancel), SK(dial), SK(conf), SK(xfer)

4   Call Features

Selecting Audio I/O Device and Line

There are 3 sets of audio i/o devices available: a) Handset, b) Built-in microphone and speaker, and c) External microphone and headset. The speaker is also used for ringing. For convenience, we shall simply refer to (b) as the speaker, and (c) as the headset where the context is clear.

Only one audio device can be selected at anytime. The speaker is selected by pressing the SPEAKER button. The headset is selected by pressing the HEADSET button. When the SPEAKER or HEADSET is selected, the corresponding LED will be steady GREEN. *The handset is selected whenever it goes from the on-hook to the off-hook position.* Hence selecting or turning on of any of the audio device can be thought of as equivalent to an off-hook action, while turning off of any of the audio device while it is selected can be thought of as equivalent to on-hook or hang-up operation.

When there are no active calls, all audio devices are unselected. When any one of the calls becomes active, the handset will be selected automatically as the audio device if it is off-hook, or else the speaker or the headset will be selected according to configuration: the user can configure whether speaker or the headset device has higher preference.

A line is selected by pressing the corresponding Line Key.

To make or receive calls, the user must select a call appearance (Line Key) or an audio device. If the user selects a call appearance, then an audio device is selected automatically according to user's preference setting. If the user selects an audio device, an idle call appearance is selected automatically in the order L1, L2, L3, and L4. The user may switch between audio devices while the call in any of the active states.

Exception: When a new call appearance is selected automatically when user presses SK(conf) or SK(xfer), the SPA-841 will attempt to pick an idle call appearance that is on the same Extension as the last active call. If that cannot be found, the SPA-841 then tries to pick one whose Extension has the same Proxy server as the last active call. Finally, it picks the first idle call available in the order L1, L2, L3, and L4.

Making Calls

There are 2 steps in making a call: 1) Select an audio device, and 2) Dial the number. SPA-841 lets the user perform these 2 steps in either order. That is, the user can select an audio device before or after dialing. SPA-841 also supports 2 types of dialing: explicit or implicit. Explicit dialing refers to one of the following cases:

- enter the target number digits one by one
- enter the speed dial of the target number
- select an entry from a directory and press SK(dial)
- a dedicate hard key is pressed, such as the VM key

Implicit dialing refers to the case where a number or an entry in a directory is highlighted. When a number implicitly dialed, the user can simply select an audio device or call appearance to have the selected dialed.

Only explicit dialing is supported if an audio device is selected prior to dialing. Both explicit and implicit dialing are allowed if audio device is selected after dialing. In particular, explicit dialing before audio device is selected is known as *on-hook dialing*.

By selecting an audio device before dialing, you also indirectly select a call appearance for making the call. SPA-841 will pick the first available call appearance (starting from Line Key 1). If no call appearances are available, no audio devices can be selected.

On the other hand, the user can explicitly select a call appearance by pressing the corresponding Line key, which indirectly selects an audio device. If the handset is off-hook, it will be the default audio device. Otherwise the default device is the one set in <Preferred Audio Device>, which is also configurable from the Phone GUI (**i** → Preferences)

Answering Calls

The Line key LED blinks rapidly in red on incoming calls. The user can answer the call by:
- Selecting an audio device. The SPA-841 will answer the ringing call. If there is more than 1 call ringing at the moment, the call is picked up in ascending order of the Line key number.
- Pressing the corresponding Line Key. The default audio device will be selected automatically.

Ending Calls

A call can be ended by the user if it is in an active state, by turning off the currently selected audio device, or by pressing the corresponding Line key. A call in a standby state cannot be ended; it must be brought back to an active state before it can be ended.

Hold and Resume

The user can place all connected calls on hold by a) explicitly pressing the Hold key, or b) pressing another Line Key (to answer an incoming all, resume a held call, or start a new call). When a call is on

hold, the corresponding Line Key LED blinks slowly in red. To resume a holding call, press the corresponding Line Key.

Pressing the Hold key always apply to all the currently connected calls, but it does not affect calls in other states. Resume operation only applies to the current call appearance. These combinations will allow very flexible call handling.

Note: When the phone is hosting a 3-way conference, two call appearances are in the connected state. Pressing Hold key will hold both calls. Then the user can press a Line Key to bring back one of parties to the connected state.

Call-Waiting

Call-waiting here refers to the case of a new incoming call while there is another active call. The SPA-841 does not support multiple calls on the same Line key. The corresponding Line key blinks quickly in red on any incoming call. If there is no other active calls, the SPA-841 will ring with either the default ring of that extension or the distinctive ring associated with the caller. If there is another active call, however, the SPA-841 will not ring the phone, but plays the call-waiting tone to alert the user.

The SPA-841 supports multiple call-waiting. In fact, all 4 call appearances can ring at the same time.

3-Way Conference

There are two (2) ways to start a 3-way conference on the SPA-841.

- While the user is in a connected call with one or more free call appearances available, SK(conf) will appear. Pressing this soft key will hold the active call and start new dial tone on the next available call appearance. You then dial the 3rd party that you want to conference with. When the 3rd party rings or answers, the user presses SK(conf) again to start the conference.

- While the user is on a connected or proceeding call with one or more call appearances on hold or ringing, SK(confLx) will appear. Pressing this soft key will hold the active call and brings up a screen to prompt to user to press one of the holding or ringing call Line key whom he wants to conference with. If there is one other holding or ringing call, pressing SK(confLx) will start the conference immediately without the need to press another Line key.

Once the conference starts, the SPA-841 plays a special brief tone to all parties to indicate that all 3 parties are now online. The user can terminate the call at any time by hanging up the call. Alternatively the user can drop out of the conference by pressing SK(join) to let the other 2 parties to continue with the conference (this is implemented as a call transfer; so it might not work if the other 2 parties are hosted on different service providers).

Note: If the extension is configured with an external Conference Bridge URL, then conference is not limited to 3-way; it can be as many parties as the conference bridge can handle. Two restrictions apply when using an external conference bridge: a)The conference can only be completed when the 3rd party has answered (not just ringing) using SK(conf), and b)You can only bring a holding call into the conference using SK(confLx), but not a ringing call.

Attended Call Transfer

There are 2 ways:

- While the user is in an active call with one or more idle call appearances, the SK(xfer) will be available. Press SK(xfer) places the current call on hold, and starts a new dial tone on one of available line keys. Dial the transfer target number. When the target starts ringing or answers, pressing SK(xfer) again to complete the transfer.
- While the user is in on a connected or proceeding call with one or more call appearances on hold, the SK(xferLx) will be available. Pressing SK(xferLx) followed by the line key of the holding call that you want to transfer to. If there is only 1 other call on hold, pressing SK(xferLx) immediately completes the call transfer without the need to press another Line Key again.

Notes:
- At the completion of call transfer operation, the holding call peer will be the transferee, and the connected or proceeding call peer will be the transfer target
- We refer to the case where a call transfer is completed when the transfer target is still ringing as semi-attended call transfer. This should not be confused with a blind transfer which is described in the next subsection.

Blind Call Transfer

We define a blind transfer as a call transfer operation where the transferor does not actually call the transfer target. The transferor simply sends the transfer target information to the transferee in a signaling message and then drops out of the loop. With this information, the transferee then calls the transfer target.

To perform a blind transfer, press SK(xfer) or an unused Line Key during an connected call (where the current call peer is the transferee) to start a new dial tone (i.e. you must have a spare call appearance). Then enter *98; the phone will prompt the user to enter the transfer target number. Enter the number to complete the blind transfer. The last call peer placed on hold by the SPA-841 will be the transferee.

Call Back

This service mimics the call back service offered by the PSTN. The user can activate this service on a busy number such that he will be called back as soon as the busy number becomes available. The SPA-841 (and other members in the SPA family) implements this feature by repeatedly dialing the busy number periodically until the called party rings or answers, or until the service order is canceled by the user or expires. For that reason, it is better referred to as the repeat dialing service.

To activate this service, the user selects a Line key and enters *66 (Call back activation code). The SPA-841 then uses this call appearance to call the last called number. The retry period and expiration time are configurable. To cancel the service, the user picks up any line and enter *86. When call back service is active, the corresponding Line key will blink in green. The user can still use this line to make or answer calls; when that happens the call back service is temporarily paused until the call appearance is idle again.

When the call back number rings or answers, the call appearance will ring also like a normal incoming call (the ring to be used is the default ring for the corresponding extension for that Line key). If the call back party answers the call before the local user does, the SPA-841 sends holding tone to the call back party. If the local user picks up the call back Line key first, he will hear ring back tone like a regular outbound call.

Message Waiting Indication (MWI)

The SPA-841 indicates message waiting in 2 ways:

1. Turning on the red MWI light - LED (with an envelope icon right above LED) on the phone.
2. Playing stutter tone instead of regular dial tone. **The MWI light only applies to new message indication received on Extension 1**. It will not be turned on or off due to new message indication received on the other Extensions [2 – 4]. On the other hand, the SPA-841 can be configured to play a stutter tone on the individual call appearance according to the message waiting state of the corresponding Extension "X".

Accessing Voice Mail

The SPA-841 has a VM retrieval key for quick access to voice mail. If a voice mail access number or URL is not specified, pressing the key brings up a screen where the user can enter the appropriate information. This information can also be entered from the web page or downloaded via provisioning.

The SPA-841 will attempt to call the voice mail number or URL if there is an idle call appearance available.

Muting Calls

The user can mute the audio input into the phone when an audio device is switched on, by pressing the MUTE key. To un-mute, press the MUTE key again. If no audio device is on, pressing the MUTE key has no effect. When switching from the speaker phone to the handset, the phone is automatically un-muted.

Shared Call Appearances

The SPA-841 supports shared call appearances in association with a Broadsoft application server. An extension can be shared by two (2) or more stations. Any call appearances on a shared extension is a shared call appearance. At any given time, each station sharing a call appearance can monitor the state of the call appearance. A station can select a share call appearance to make a call only if the call appearance is not being used by another station. All stations will ring on an inbound call to the shared call appearance extension. Whoever picks up the call first will take the call. When a call is placed on hold by one station, it can be resumed from another station sharing the same call appearance.

Additional methods for shared call appearances per vendor application server implementation are being added to the SPA-841 and will be available in future firmware releases. Please contact sales@sipura.com to inquire about roadmap / development schedules.

Line Key LED Behavior

The following outlines the Line Key LED Behavior:

*   A Line Key corresponds to a call appearance. Call appearances for the same extension are numbered in ascending order of their Line Key position starting at 1. The Line Key LED color and blinking pattern are fully programmable. The administrator can specify a different color and pattern for each well-defined states of the call appearance. These states are described below.
*   Idle: This call appearance is not in use. The user can use it to make outbound call on this station
*   Local Seized: This call appearance has been seized by this station to prepare for an outbound call
*   Local Progressing: This station is making an outbound call that is progressing
*   Local Active: This station is engaged in a connected call on this call appearance
*   Local Ringing: This station is ringing for an incoming call on this call appearance

- Local Held: This station has placed this call appearance on hold
- Remote Seized: This call appearance has been seized by another station to prepare for an outbound call
- Remote Progressing: Another station is making a call on this call appearance and is progressing
- Remote Active: Another station is engaged in a connected call on this call appearance
- Remote Ringing: Another station is ringing for an incoming call to this call appearance
- Remote Held: Another station has placed this call appearance on hold
- Remote Undefined: The share call state is not known (this station is waiting for a notification from the application server)
- Registration Failed: This station has failed to register with the proxy server for the corresponding extension
- Registering: The station is attempting registration with the proxy server for the corresponding extension.
- Disabled: This Line Key on this station is disabled
- Call Back: A call back (repeat dialing) operation is currently active on this call appearance

The Line Key LED supports 4 colors: Red (r), Green (g), Amber (a), and Off (o). The SPA-841 has 4 built-in blinking patterns defined:

- No Blink (nb): Steady on (or off)
- Slow Blink (sb): 500ms on, 500ms off
- Fast Blink (fb): 100ms on, 100ms off
- Double Blink (db): 100ms on, 100ms off, 100ms on, 700ms off

The administrator can also define arbitrary blinking pattern by following the syntax of a LedScript. The default LedScript for each call appearance state is described in section 3.

Other Supplementary Services

The SPA-841 supports the following supplementary Services:

Block Caller ID

If enabled, the SPA-841 will attempt to hide the caller-ID on all outbound calls by default. The user can also enable or disable this feature on a per call basis by pre dialing a *code before making the call.

Block Anonymous Call

Reject all inbound calls with no caller-ID with a 406 response

Do Not Disturb (DND)

Reject all inbound calls as if the phone is busy. That is, the phone either responds a 302 if call forward on busy if enabled, or 486 otherwise.

Secure Call

If enabled, the SPA-841 will try to negotiate on all outbound calls to use encrypted media (SRTP) by default. The user can also enable or disable this feature on a per call basis by pre dialing a *code before

making the call. Note that in order to use Secure Call on an extension, <Mini Certificate> and <SRTP Private Key> must be configured for that extension.

The enabling and disabling of these services applies to calls on all configured extensions on the SPA-841

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. The phone screen will be updated with the CID information extracted from the Mini-Certificate received from the other end. The call state label on the phone screen will also be prepended with a '$' symbol (such as $Connected instead of Connected).

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 partyThe 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+EBZmi3AmcqE9U1LxEOGwopaGyGOh3VyhKgi6JaVtQZt87PiJINKW8XQj3B9Qqe3V
gYxWCQNa335YCnDsenASeBxuMIEaBCYd1I1fVEodJZOGwXwfAde0MhcbD0kj7LVlzcsTyk2TZYTccnZ7
5TuTjj13qvYs=
5nEtOrkCa84/mEwl3D9tSvVLyliwQ+u/Hd+C8u5SNk7hsAUZaA9TqH8Iw0J/IqSrsf6scsmundY5j7Z5mK5J
9uBxSB8t8vamFGD0pF4zhNtbrVvIXKI9kmp4vph1C5jzO9gDfs3MF+zjyYrVUFdM+pXtDBxmM+fGUfrpAu
Xb7/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>
Sm9lIFNtaXRoAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAxNDA4NTU1MTIzNAAAAAAAMDAwMDAw
MDEwMTM00OvJakde2vVMF3Rw4pPXL7lAglagMpbLSAG2+++YlSqt198Cp9rP/xMGFfoPmDKGx6JFtk
Q5sxLcuwgxpxpxkeXvpZKlYlpsb28L4Rhg5qZA+Gqj1hDFCmG6dffZ9SJhxES767G0JIS+N8lQBLr0Auem
otknSjjjOy8c+1lTCd2t44Mh0vmwNg4fDck2YdmTMBR516xJt4/uQ/LJQlni2kwqlm7scDvll5k232EvvvVtCK0
AYa4eWd6fQOpiESCO9CC9aYU1X5lJuU+EBZmi3AmcqE9U1LxEOGwopaGyGOh3VyhKgi6JaVtQZt87P
iJINKW8XQj3B9Qqe3VgYxWCQNa335YCnDsenASeBxuMIEaBCYd1I1fVEodJZOGwXwfAde0MhcbD0kj
7LVlzcsTyk2TZYTccnZ75TuTjj13qvYs=

<SRTP Private Key>
b/DWc96X4YQraCnYzl5en1CIUhVQQqrvcr6Qd/8R52IEvJjOw/e+Klm4XiiFEPaKmU8UbooxKG36SEdKus
p0AQ==

5   Memory Features and other User Accessible Settings

Call Logs

There are 3 call logs maintained for each VoIP interface:
- Redial List – Each Redial List entry is added when dialing is completed, regardless the number is correct or not, or the call is successful or not
- Answered Calls – An Answered Call is logged when the coming call is answered.
- Missed Calls – A Missed Call is logged for each incoming call that causes the phone to ring but not answered.

From the Phone GUI, the user can:

View a call log:
1. Press SETUP
2. Select "Call History"
3. Select the log you want to view

Delete a log,
1. High light the entry
2. Press SK(del)

Edit a log,
1. High light the entry
2. Press SK(edit)
3. Press SK(save) to save the changes, or SK(cancel) to abort the changes

Call a log entry
1. High light the entry
2. Press SK(dial), or Off-Hook, or Turn on Speaker, or Turn on Headset, or Press a Line Key

---

Save a log entry into Personal Directory (if the directory is not already full)
1. High light the entry
2. Press SK(save)
3. Modify the entry as needed
4. Press SK(save) to save the entry in the personal directory

The SPA-841 keeps up to 60 entries per log in reverse chronological order. The logs are saved in the phone's non-volatile memory.

All call logs can be viewed on the SPA-841 web page also using the link:
http://<ip-address./calllog.htm
Each log is shown on the web page with the syntax: [*ext-id*]*name*, *phone*, *time-stamp*

- *ext-id* is the extension on which the call is made from or received at (E1, E2, E3, or E4). If [E*n*] is not present, it implies [E1].
- *name* is the call peer's name. If this field is missing, the peer's name is not available
- *phone* is the call peer's phone number (user-id). If this field is missing, the peer's phone number is not available
- *t* is the time-stamp at which the call log is entered. It has the format mm/dd hh/mm<a|p> (where a = am and p = pm

---

Example 1: [E2]Joe Smith,14089991234,t=10/12 11:12a

Example 2: 14089991234,10/12 01:12p
(On E1; name not available)

Personal Directory

A directory entry consists of the following information:
- Name
- Phone number or (SIP) URL
- Ring Tone

Up to 100 entries can be stored in the SPA-841. An entry can be added or edited from the phone GUI or on the web page or through remote provisioning. To do this from the web page or through remote provisioning, you specify the value as a DirEntry script. To view or edit the personal directory via the phone web interface, use the link http://<ip-address>/pdir.htm

The user can invoke the Directory menu on the phone in 2 ways: a) Press the **i** button and select "Directory", or b) Press SK(dir) when you're not already in a SETUP screen. The user can add a new entry to the Directory by selecting the "New Entry" option (the first item) on the Directory menu and press SK(add). To modify or delete an existing item, highlight the corresponding entry and press SK(edit) or SK(del). When trying to delete an entry, the phone will prompt the user for confirmation before proceeding.

The user can copy an existing directory entry to create a new entry. To do this, high light the entry you want to copy and press SK(copy). Then high light the "New Entry" option and press SK(paste). The user can edit the new entry and then press SK(save) to save the entry.

The user can also save a call log entry into the directory. To do this, select a call log entry and press SK(save). Then edit the entry if needed, and then press SK(save) to save the new entry in the directory.

Entering and Saving Settings

The SPA-841 GUI allows the user to enter alphanumeric and numeric inputs will ease. The following data types may be entered into the phone:

- Numeric String, such as a phone number when dialing
- Alphanumeric String, such a name when adding a Directory entry
- IP Address, such as a DNS server IP address
- Boolean (y/n), such as enabling or disabling DHCP
- Option, such as selecting a preferred audio device or a ring tone

When any two or more of the numeric string, alphanumeric string, and IP address types can apply to an entry, the phone will show a soft key for switching input modes among the applicable types. The soft keys will change when pressed to allow the user to scroll through the available modes, such as SK(num) → SK(alpha) → SK(IP). The cursor will also change according to the following table, to remind the user which input mode is being used:

Input Mode	Soft Key	Cursor (appending)	Cursor (inserting)
Numeric	num	Blinking underscore _	Blinking vertical line	
Alphanumeric	alpha	Blinking full-height block ■	Blinking open rectangle ⊏
IP Address	IP	Blinking half-height block ▪	Blinking half rectangle ∟

When entering digits during numeric mode, just enter the corresponding digits from the key pad.

When entering letters or symbols during alphanumeric mode, the phone will show a template of 2 or more choices of symbols as the user presses each digit key; the user can scroll through the choices by pressing the same key multiple times. To accept a symbol as input, the user can a)stop pressing the digit key for 1.5s, b)pressing another digit key, c)navigate to another input field (if applicable) by pressing the UP/DOWN key, or d)save the value by pressing the proper soft key, such SK(save) or SK(ok)

When entering IP address, pressing the * digit will insert a dot (.), or a colon (:) if there are 3 dots entered already.

When entering numeric, alphanumeric, or IP address values, the following additional soft keys will be available for easier editing:

- SK(<<) – Move cursor to the left by 1 character
- SK(>>) – Move cursor to the right by 1 character
- SK(erase<) – Erase the character to the left of the cursor
- SK(clear) – Clear the entire entry

Note that SK(<<), SK(>>), SK(erase<), and the digit key presses will automatically repeat by keep pressing the corresponding key down without releasing it. If input value is wider than what the LCD display can show, the SPA-841 automatically scroll the input to the left so that the user can continue to enter input. On the other hand, the SPA-841 will stop accepting further input for a give field if the maximum allowed input length has been reached for that input field.

For Boolean input, the SPA-841 will show SK(y/n) pressing which will toggle the current input between yes and no.

For Option input, the SPA-841 will show SK(option) pressing which will either a) change the input field to the next available choice in round-robin fashion, or b) bring up a menu of choices that the user can select from.

For most settings, the input value is saved once the user press SK(save) or SK(ok) before exiting the edit screen for that setting. However, there are a few exceptions where the user must press SK(save) again at the upper level when all the required changes have been made to the settings so that they are saved into the configuration at the same time. These exceptions are:

Preferences
- Call Forward
- Network

Speed Dials

The user can configure up to 8 speed dials in the SPA-841. To configure a speed dial, select the Speed Dial option under the *i* menu. The user can configure a speed dial for digit 2, 3, …, and 9. If a slot is not configured, it will show <Not Assigned> on the entry. High light the speed dial entry you want to add or modify, press SK(edit) to make the changes, and press SK(ok) to store the value.

A speed dial entry can be a phone number or URL. The user can also enter a name that matches one of the directory entries. As you enter the value, the SPA-841 will attempt to match the entry to the directory entries and brings up a list of potential matches on the screen. User can scroll down to select one of the suggested entries.

To dial a speed dial, dial the corresponding single digit like a 1-digit phone number. The user can enter "#" after the speed dial number but it is not necessary.

Caller and Called Name Matching

When making an outgoing call, SPA-841 will try to find the dialed number in the personal directory first, and then from the missed call log, and finally from the answered call log. If a match is found and the name field is present in the matched entry, it will be shown on the call screen as the called peer's name.

For incoming calls, SPA-841 will try to find the caller's phone number in the personal directory. If a match is found and the name field is present in the matched entry, it will replace the current caller ID name and will be shown on the call screen as the calling peer's name. The same name will also go into the incoming call log. If a match is not found, or the name field is not present in the matched entry, the current caller ID name will be used if it exists.

Dialing Assistance

Dialing assistance is an option the user can enable or disable under the Preferences option of the *i* menu. If the option is enabled, the phone will show up to 10 potential matches from the Redial List (most recent first) and from the Personal Directory as the user dials the target number. This feature applies to both off-hook (with dial tone) and on-hook (without dial tone) dialing. When the suggestion list appears, the user can scroll down the list using the Up/Down key to select the desired target number from the list.

When looking for matches from the Redial List and the Directory, the SPA-841 will skip any leading *codes, so that the match is purely based on the target number. The *code that will be skipped are (if configured in the phone):

- All call forward activation and deactivation codes
- Secure call activation and deactivation codes
- Block CID activation and deactivation codes
- Block anonymous call activation and deactivation codes
- DND activation and deactivation codes
- All the Prefer codec and Force codec codes
- Referral services codes and Feature Dial Services codes

Time/Date

The SPA-841 obtains the current time information from one of the following ways:

- NTP Server – One or two NTP servers can be configured to the phone. When it first boots up, the phone will try to contact the NTP server to get the current time. Then the phone periodically synchronizes the current time with the NTP server. The synchronization period is fixed at 1 hour. In between the update, the phone tracks the time with its own internal clock.
- SIP Messages – Each SIP message (request or response) sent to the SPA may contain a Date header with the current time information. If the header is present, the SPA will use it to update its current time.
- Manual Setup – SPA-841 also allows the user to manually enter the current time and date from the phone GUI or from the web page. However, this value will be overridden by the NTP time or SIP Message Date whenever they are presented to the phone. Manual setup requires the user to enter the time in 24 hour format only.

Note that the time served by the NTP Server and the SIP Date Header are GMT time. The local time is obtained by offsetting the GMT according to the time zone of the region. <Time Zone> can be configured from the web page or provisioning. This time can be further offset by <Time Offset (HH/mm)> parameter, which must be entered in 24 hour format only. This parameter can be configured from the Phone GUI also.

The <Time Zone> and <Time Offset (HH/mm)> offset values are not applied to the manual time/date setup.

Daylight Saving Time

The SPA-841 supports auto adjustment for daylight saving time. To enable this feature, the administrator must configure the <Daylight Saving Time Rule> (regional) parameter. This parameter is a rule with three (3) fields. Each field is separated by semicolon (;) as shown below:

```
start=<start-time>;end=<end-time>;save=<save-time>
```

where

       `<start-time>` and `<end-time>` are of the form: `<month>/<day>/<weekday>[/HH:[mm[:ss]]]`

       `<save-time>` is of form: `[/[+|-]HH:[mm[:ss]]]`

       `<month>` = 1,2,3,…,12 (for Jan, Feb, …, Dec)

```
<day> = [+|-]1,2,3,...,31
```

`<weedday>` = 1, 2, 3, …, 7 (for Mon, Tue, …, Sun), or 0 which has special meaning

`HH` = hour (0-23)

`mm` = minute (0-59)

`SS` = second (0-59)

<start-time> and <end-time> specify the start and end time-date of daylight saving time and <save-time> is the amount of hour/min/sec to add to the current time during daylight saving period. The <save-time> value can be preceded by a negative (-) sign if subtraction is desired instead of addition.

If <weekday> is 0, it means the date to start or end daylight saving is at exactly the given date. In that case, the <day> value must not be negative. If <weekday> is not zero, then the daylight saving starts or ends on the <weekday> on or after the given date if <day> is positive, or on or before the given date if <day> is negative. If <day> is −1, it means the <weekday> on or before the end-of-the-month (in other words the last occurrence of < weekday> in that month).

Optional values inside [ ] are assumed to be 0 if they are not specified. Midnight means 0:0:0 of the given date.

Example 1:
Starts at midnight on 1st Sunday of April; ends at midnight on the last Sunday of October; add 1 hour (USA, N. America). Below are all equivalent rules:

```
start=4/1/7/0:0:0;end=10/31/7/0:0:0;save=1
start=4/1/7;end=10/-1/7;save=1
start=4/1/7/0;end=10/-1/7/0;save=1
```


Example 2:
Starts at midnight on last Sunday of April; ends at midnight on the last Sunday of September; add 1 hour (Egypt):

```
start=4/-1/5;end=9/-1/4;save=1 (Egypt)
```


Example 3:
Starts at midnight on first Sunday of October; ends at midnight on the 3rd Sunday of March; add 1 hour (New Zealand):

```
start=10/1/7;3/22/7;save=1  (New Zealand)
```

Checking Phone Status

Besides the Info group on the web page, the Phone GUI has a Status menu where one can check the current status of the phone.

Reboot and Restart

Reboot will cold boot the phone and restart DHCP. Restart will warm boot the phone, without restarting DHCP. Both options require the user to confirm before proceeding.

Factory Reset

Factory reset will reset all parameters to default value. Personal directory and call logs will be cleared also. This option require the user to confirm before proceeding.

Password Protection

The user can configure a password to the phone to protect against access to the following options:

- Directory
- Call History (Redial List, Missed Calls, and Answered Calls)
- Call Forward
- Speed Dial
- Ring Tone
- Preferences
- Time/Date
- Voice Mail
- Factory Reset
- Set Password

Audio Volume Adjustment

There are 4 kinds of audio volume to be adjusted:
- Ringer Volume (on the speaker)
- Speaker
- Handset
- Headset

To adjust the volume, press the VOL key to invoke the volume adjustment screen. Then use the UP/DOWN keys to increase or decrease the volume. The kind of volume to adjust is by context. When no audio device is selected at the moment, the Ringer volume will be adjusted. Each UP/DOWN key press will play a short burst of the current ring tone. If one of the audio devices is selected, then the volume adjustment applies to that device. The screen will show why kind of volume is being adjusted with a pictorial representation of the volume level.

Each volume has 16 steps in nonlinear logarithmic scale. Each setting is a 16-bit multiplier in the range 41 to 32764 in Q3.12 format (or 0.01 to 7.999 or –40 to 18 dB). Each step is equal to 58/(16-1) or 3.87 dB.

Additional volume settings can be added and accessed via the phone setup screens. These volumes are:
- DTMF Echo Volume (use <DTMF Playback Level>)
- Call Progress Tone Volume (setting the ToneScript for each tone)

Ring Tone

The SPA-841 offers 10 ring tones. The characteristics of each ring tone is configurable using a RingScript (see Section 3). In a RingTone script, the user can assign a name for the ring tone, and specifies 2 components for it:

- Waveform – 1, 2, 3, or 4
- Cadence – 1, 2, 3, …, or 8

The user can assign a default ring tone for each extension. The user can assign a ring tone for each personal directory entry (which overrides the default ring tone). Ring tone can be selected from the GUI.

In addition, the SPA-841 allows 2 user-downloadable ring tones. Therefore a total of 12 ring tone choices are available to the user to be used as default ring for an extension or assigned to individual callers in the personal directory. The user downloaded ring tone as labeled as "User 1" and "User 2" in the choices for the <Default Ring>. On the phone ring tone menu, the User 1 and 2 choices are replaced by the corresponding name of the ring tone or "Not Installed" if the user ring tone slots have not been downloaded into.

The user ring tone (1 or 2) can be downloaded using the phone web interface with the link:

> http://<phone-ip-addr>/ringtone1[?<url>]

where <url> syntax is [tftp://][host[:port]]/<path>. Only tftp is supported. If host is not specified, the TFTP host is the web client; default port is 69.To remove user ringtone 1 from the phone, set the <path> to "delete", such as:

> http://<phone-ip-addr>/ringtone1?/delete

The link is case sensitive.

Notes:
- For User Ring Tone 1 and 2, the cadence is fixed with the on-time equals to the duration of the ring tone file and off-time equals to 4s. The total ring duration is fixed at 60s.
- The user ring tone names displayed on the phone GUI are extracted from the ring tone file header file
- No need to reboot phone after a ring tone download.
- However, if your phone is already in ring tone GUI menu while the new ring tone is downloaded, you will need to exit and re-enter the ring tone menu in order to see the new ring tone name on the menu.

Star Code to Activate/Deactivate Certain Services

THE SPA-841 accepts star codes to activate or deactivate certain services on the phone. Below is a list of services that are accessible with a star code, with the default star code shown in parenthesis:

- Call Return (*69) – call the last caller, regardless which extension
- Call Back(*66) – Periodically redial the last busy number (every 30s by default) until it rings or until the trial expires (30 min by default), regardless which extension. Only 1 call back operation can be ordered at a time. A new order will automatically cancel the last order.
- Cancel Call Back(*86) – Cancel the last call back operation
- Call Forward All (*72) – Call forward all inbound calls. Applicable to primary extension only
- Cancel Call Forward All (*73) – Cancel call forward all. Applicable to primary extension only
- Call Forward Busy (*90) – Call forward on busy. Applicable to primary extension only

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- Cancel Call Forward Busy (*91) – Cancel call forward on busy. Applicable to primary extension only
- Call Forward No Answer (*92) – Call forward if no answer. Applicable to primary extension only
- Cancel Call Forward No Answer (*93) – Cancel call forward no answer. Applicable to primary extension only
- Block Caller ID Always (*67) – Block caller ID on all outbound calls. Apply to all extensions
- Block Caller ID Per Call (*81) – Block caller ID on the next outbound call (on the current call appearance only).
- Unblock Caller ID Always (*68) – Unblock caller ID on all outbound calls. Apply to all extensions
- Unblock Caller ID Per Call (*82) – Unblock caller ID on the next outbound call (on the current call appearance only).
- Secure All Calls (*16) – Default to prefer to use encrypted media for all outbound calls. Apply to all extensions.
- Secure No Calls (*17) – Default to prefer to use unencrypted media for all outbound calls. Apply to all extensions.
- Secure Next Call (*18) – Prefer to use encrypted media for the next outbound call (on this call appearance only)
- Do Not Secure Next Call (*19) – Prefer to use unencrypted media for the next outbound call (on this call appearance only).
- Do Not Disturb (*78) – Apply to all extensions
- Cancel Do Not Disturb (*79) – Apply to all extensions
- Block Anonymous Calls – Apply to all extensions
- Cancel Block Anonymous Calls – Apply to all extensions
- Blind Transfer (*98)
- Prefer G711u (*017110) – Set preferred codec to G711u for next outbound call
- Prefer G711a (*017111) – Set preferred codec to G711a for next outbound call
- Prefer G723 (*01723) – Set preferred codec to G723 for next outbound call
- Prefer G726r16 (*0172616) – Set preferred codec to G726r16 for next outbound call
- Prefer G726r24 (*0172624 – Set preferred codec to G726r24 for next outbound call
- Prefer G726r32 (*0172632) – Set preferred codec to G726r32 for next outbound call
- Prefer G726r40 (*0172640) – Set preferred codec to G726r40 for next outbound call
- Prefer G729a (*01729) – Set preferred codec to G729a for next outbound call
- Force G711u (*027110) – Force to use G711u for next outbound call
- Force G711a (*027111) – Force to use G711a for next outbound call
- Force G723 (*02723) – Force to use G723 for next outbound call
- Force G726r16 (*0272616) – Force to use G726r16 for next outbound call
- Force G726r24 (*0272624 – Force to use G726r24 for next outbound call
- Force G726r32 (*0272632) – Force to use G726r32 for next outbound call
- Force G726r40 (*0272640) – Force to use G726r40 for next outbound call
- Force G729a (*02729) – Force to use G729a for next outbound call

When off-hook dialing, the star code is processed by the phone as soon as they are entered (and recognized by the phone). When on-hook dialing, the user can pre-dial multiple star code before the target number, such as *67*017110…. ; the star codes will be handled one by one in the order they are entered, before the target number is dialed out. There is SK(*code) in the dialing soft key to assist the user on what *code is available.

Star code can be included in a directory entry and redial list.

In addition there are two special codes:

- Referral Services Codes – This is a list of '|' separated star codes (such as "*79|*199|*09"), where entering each member triggers a blind transfer operation except the transfer target number in this case will be prepended by the corresponding star code (whereas blind transfer itself contains the target number only without the star code). Like blind transfer, a referral service code can only be entered when the last call is placed on hold.
- Feature Dial Services Codes – this is a list of '|' separated star codes where entering each member will cause the next outbound call target number prepended by the corresponding star code.

Disabling Services

Services handled locally by the phone can be disabled by the administrator in one or two ways: a) by directly disabling the service in the configurable profile, or b) by emptying out the *code associated with the service.

Services that can be disabled (or enabled) by method (a) are:
- Block Caller ID (including blocking and unblocking, per call and always)
- Block Anonymous Calls
- Do Not Disturb
- Secure Call (including enabling and disabling, per call and always)
- Call Forward All
- Call Forward Busy
- Call Forward No Answer
- Call Back
- Conference Call (no star code)
- Attended Call Transfer (no star code)
- Blind Call Transfer

Services that can be disabled (or enabled) by method (b) are any of the services that have a star code assignment in the SPA-841 configuration profile (see Section 21 on a complete list). Note that if service is enabled, but the corresponding *code is emptied, the service can still be enabled/disabled by the end-user from the phone GUI or web page.

If a service is disabled, any soft key associated with that service will be hidden from the GUI. These are: SK(dnd), SK(-dnd), SK(cfwd) and SK(-cfwd). Any menu item associated with a disabled service will be preceded with an exclamation mart (!) and no soft keys will be visible to select that entry.

6   Voice and Signaling Features

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

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.

SIP Proxy 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.

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.

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.

Voice Algorithms:

The SPA-841 supports the following voice algorithms.

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.

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.

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.

G.723.1

The SPA supports the use of ITU G.723.1 audio codec at 6.4 kbps.

Codec Selection

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

Dynamic Payload

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

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.

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.

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.

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.

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.
.

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.

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.

Please see Section 10 for complete Dial Plan implementation details.

Network Address Translation (NAT) Traversal

The SPA-841 supports NAT traversal for VoIP signaling and media packets. This section overviews the basics around NAT.   The SPA-841 NAT features are reviewed in the configuration section.

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.

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.

Note: NAT Mapping does not have to applied globally on the phone. It can be enabled or disabled per Extension. Some service providers offer their own solution of NAT traversal; NAT mapping should be disabled on the SPA-841 for extensions configured for these service providers.

7 Data Networking Features Supported

The SPA-841 supports the following Data Networking Features:
MAC Address (IEEE 802.3)
IPv4 – Internet Protocol Version 4 (RFC 791) upgradeable to v6 (RFC 188
ARP – Address Resolution Protocol
DNS – A Record (RFC 1706), SRV Record (RFC 2782)
DiffServ (RFC 2475) and ToS – Type of Service (RFC 791/1349)
DHCP Client – Dynamic Host Configuration Protocol (RFC 2131)
ICMP – Internet Control Message Protocol (RFC792)
TCP – Transmission Control Protocol (RFC793)
UDP – User Datagram Protocol (RFC768)
RTP – Real Time Protocol (RFC 1889) (RFC 1890)
RTCP – Real Time Control Protocol (RFC 1889)
SRTP - Secure Real Time Control Protocol (RFC XXXX)

8 Configuring and Provisioning Overview

The SPA-841 requires the administrator to ensure that two files are loaded correctly onto the device. These files are:

- A firmware file (e.g. SPA841.bin). This file provides the operating system and call processing functions for the device. Firmware upgrades are explained in Section 13.

- A configuration file (e.g. SPA841.cfg).   This file provides the specific parameter values for the administrator's network. Many of the remaining sections explain the details on the configuration file i.e. the configurable parameters, their values, how to edit the parameters, how to load the configuration file securely and remotely.

Both files can be loaded to the device remotely using TFTP, HTTP or HTTPS.   The configuration file can be manually accessed and changed using the phone's web server and that web interface is explained in the next section.

Note: In this document, Configuration refers to the parameter values of the configuration file. Provisioning refers to preparing the device with the correct configuration and firmware files for the network implementation.

9   Web Interface Administration and Security

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

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/.
- 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 0 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.

Web Interface 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 the user 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.

Web Interface 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.

10   SPA-841 Configuration Parameters

This section tabulates all the SPA-841 parameters. The parameters can be configured in 2 ways: a) phone configuration web page, and b) remote provisioning. In addition, a small subset of "user options" can be set via the phone's GUI under various setup menus.

On the configuration web page, the parameters are organized into several groups with each group hidden under a "tab" on the top of the web page. To see the group of parameters, the user click on the corresponding "tab" on the web page. The following groups are defined at present:
* Info – Status Information (read only). This is the default "tab" when the web page is first loaded.
* System – System level parameters, including network and debug parameters
* SIP – Parameters to adjust the SIP stack behavior
* Provisioning – Parameters that control remote provisioning
* Regional – Parameters that depends on country or region, such as call progress tones and * codes
* Phone – Parameters that apply to all the extensions configured for the phone
* Ext 1 – Parameters that apply to Extension 1
* Ext 2 – Parameters that apply to Extension 2
* Ext 3 – Parameters that apply to Extension 3

- Ext 4 – Parameters that apply to Extension 4
- User – User level parameters (other parameters are presumably administrator level)

Each parameter has a name and corresponding value. A parameter-name is of the form: seg[_seg][[*n*]], where seg is text segment composed of ascii characters only excluding space and underscore, [ ] indicates optional components, and *n* = 1, 2, 3, 4. Parameter names are case sensitive. Parameter values may be case sensitive or case insensitive, depending on the parameter. Examples are: Proxy[1], RTP_Packet_Size, DND_Act_Code.

When a parameter name is displayed on the web page, the underscore is replaced by a single space and the [*n*] component, if exists, will be hidden. Examples are: Proxy, RTP Packet Size, DND Act Code.

Notations

- *var* (in italics) represents a variable or value to be substituted by the actual value it represents. *var* can appear standalone or anywhere within a longer string.
- a|b|c represents choices. The value can be one of the given choices.
- [optional value] represents an optional value
- <Par Name> represents a configuration parameter name, such as <Proxy>. In a profile, the corresponding tag is formed by replacing the space with an underscore "_", such as <RTP Packet Size> is replaced with RTP_Packet_Size. Some parameters carry an implicit index value if they apply to multiple extensions, such as <Proxy> which exists for Extensions 1, 2, 3, and 4. In a profile, a "[*n*]" must be appended to the parameter to make it unambiguous. In this example, n = 1, 2, 3, 4 for Proxy[1], Proxy[2], Proxy[3], and Proxy[4] respectively.
- An empty default value field implies an empty string ("")

Notes:
- 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.

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
Int – A generic integer value. The range depends on the parameter.
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 11 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

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 $f_{i,j} = n_1[+n_2]+n_3[+n_4[+n_5[+n_6]]]]$ 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

RingScript – A script that describes a ring tone.

Syntax: n=*ring-tone-name*;w=*waveform-id-or-path*;c=*cadence-id*;b=*break-time*;t=*total-time*

ring-tone-name is a name to identify this ring tone specification. This name will appear on the Ring Tone menu of the phone. The same name can be used in a SIP Alert-Info header in an inbound INVITE request to tell the phone to play the corresponding ring tone specification. Because of this, the name should contain characters allowed in a URL only.

Waveform-id is the index of the desired waveform to use in this ring tone specification. There are 4 built-in waveforms at present:
1 = A classic phone with mechanical bell
2 = Typical phone ring
3 = A classic ring tone
4 = A wide-band frequency sweep signal
This field can also be a network path (url) to download a ring tone data file from a server on-the-fly. In this case, the syntax of the field is

w=[tftp://]hostname[:port]/path.

cadence-id is the index of the desired cadence to play the given waveform. 8 cadences (1–8) as defined in <Cadence 1> through <Cadence 8>. *Cadence-id* can be 0 If w=3,4, or an url. Setting c=0 implies the on-time is the natural length of the ring tone file.

break-time specifies the number of seconds to break between two bursts of ring tone, such as b=2.5

total-time specifies the total number of seconds to play the ring tone before it times out

Example 1: n=Classic-1;w=3;c=1

Example 2: n=Simple-1;w=2;c=1

Example 3: (TBD)

LedScript – A script that describes the color and blinking pattern of a Line Key LED. Each script contains a number of fields separated by a semicolon(;). White spaces are ignored. Each field has the syntax *<field-name> = <field-value>*. The allowed *field-name* and corresponding *field-values* are listed below:
c=o|r|g|a
This field sets the color of the LED. The 4 choices are:
o = off
r = red
g = green
a = amber

p=n[b]|s[b]|f[b]|d[b]|u[d]

This field sets the blinking pattern of the LED. The 4 choices are:
nb = no blink (steady on or off)
sb = slow blink (1s on and 1s off)
fb = fast blink (100ms on and 100ms off)
ud = user-defined (according to the contents of the u field)
u=on/off/on/off...
This is a user-defined blinking pattern used only when p = ud. It consists of up to 4 pairs of on/off duration in seconds with up to 2 deciaml places; each value is separated by a forward slash (/).

> **Example 1:** c=r;p=sb
> Color is red and slow blink
>
> **Example 2**: c=o
> LED is off
>
> **Example 3**: c=g
> Color is green and steady on
>
> **Example 4**: c=a;p=ud;u=.1/.1/.1/.1/.1/.9
> Color is amber and blink with the pattern (100ms on, 100ms off, 100ms on, 100ms off, 100ms on, 900ms off).

DirEntry – A script that describes a directory entry

Syntax: n=*name*;p=*phone*;r=*ring-tone-id*

name is the name of the entry
phone is the phone number or URL of the entry
ring-tone-id is the ID of the ring tone (1–10) that should be played when the entry is called.

> **Example 1:** n=Joe Smith;p=14089991234
> (ring-tone = default)
>
> **Example 2**: p=123456@192.168.2.12:5061;r=5
> (unknown name)

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.

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 .	Bool	Yes
Web Server Port	TCP port through which the SPA web server will communicate	Uns8	80
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 those supplied by DHCP server if DHCP is enabled (See "DNS Server Order" for more information); 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 those DHCP server supplied if DHCP is enabled (See "DNS Server Order" for more information); when DHCP is disabled, this will be the secondary DNS server.	IP	0.0.0.0
DNS Server Order	Controls how to organize the DNS servers supplied by DHCP server and the ones statically configured; it is used only when DHCP is enabled. Manual: Use statically configured DNS servers if there is any; otherwise, use the ones supplied by DHCP server. "Manual,DHCP", "DHCP,Manual": Both DNS servers supplied by DHCP server and the ones statically configured, but in different order.	Choice	Manual
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	

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

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
Forced Resync Delay	Maximum time SPA will wait before initiating a resync operation if it is not idle.	Time0	14400
Resync From SIP	Enables resync of configuration profile from a SIP command	Bool	Yes
Resync After Upgrade Attempt	Whether or not to resync following an upgrade attempt	Bool	Yes
Resync Trigger 1	Scriptable resync condition	Script	empty
Resync Trigger 2	Scriptable resync condition	Script	empty
Resync Fails On FNF	Whether file-not-found response from provisioning server should be considered a failed resync	Bool	Yes
Profile Rule	Primary configuration profile URL script.	ProfileScript	/spa841.cfg
Profile Rule B	Secondary configuration profile URL script	ProfileScript	empty
Profile Rule C	Secondary configuration profile URL script	ProfileScript	empty
Profile Rule D	Secondary configuration profile URL script	ProfileScript	empty
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	ProfileMsg	See

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	resync		provisioning discussion section
Log Resync Failure Msg	Syslog message generated after a failed resync	ProfileMsg	See provisioning discussion section

Upgrade Parameters

Parameter Name	Description	Type	Default
Upgrade Enable	Master enable for firmware upgrade operations	Bool	Yes
Upgrade Error Retry Delay	Retry interval following upgrade failure	Time0	3600
Upgrade Rev Limit	Restricts downgrades to versions greater than a specified minimum version	RevNum	empty
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
License Keys	Premium features license keys. The key need only be entered once. (write-only parameter)	String	empty

SIP Parameters

SIP Parameters

Parameter Name	Description	Type	Default
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	This is the MIME Type to be used in a SIP INFO	Str31	application/dtmf-relay

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Type	message used to signal DTMF event.		
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)

Parameter Name	Description	Type	Default
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

Parameter Name	Description	Type	Default
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

Parameter Name	Description	Type	Default
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	0.03
Max RTP ICMP Err	Number of successive ICMP errors allowed when transmitting RTP packets to the peer before the SPA will terminate the call. If value is set to 0, the SPA will ignore the limit on ICMP errors.	Uns32	0
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.

Dynamic Payload Types

Parameter Name	Description	Type	Default
AVT Dynamic Payload[1,2]	AVT (DTMF tones) dynamic payload type	Uns8	101
INFOREQ Dynamic Payload	If this value is not blank, the SPA will include X-nt-inforeq in the SDP to indicate support of INFO method for out-of-band DTMF transmission (per Nortel MCS specification). The recommended value to use is 111	Uns8	
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:

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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

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.

NAT Support Parameters

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 UAS in a response. Else the parameter is ignored.	Bool	No
Insert VIA received	If set to "yes", the SPA will insert received parameter in VIA header in SIP responses if received from IP and VIA sent-by IP differ	Bool	No
Insert VIA rport	If set to "yes", the SPA will insert rport parameter in VIA header in SIP responses if received-from port and VIA sent-by port differ	Bool	No
Substitute VIA addr	If set to "yes", the SPA will use nat-mapped IP:port values in VIA header	Bool	No
Send Resp To Src Port	If set to "yes", SPA will send a response to the source port where the corresponding request is received from. Else the SPA will send the response as indicated in the VIA header by the UAC	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	Bool	No

	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:
	    "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	If this parameter is specified, the SPA will assume this IP address when generating SIP messages and SDP (if NAT Mapping is enabled for that Line). However, results of STUN and VIA received parameter processing, if available, will supersede this statically configured value	IP	
Ext RTP Port Min	External port mapping of <RTP Port Min>. If the parameter is specified and non-zero, the SPA will use this value to compute the corresponding external RTP port of a connection and use the result in the SDP for establishing the connection. However, results of STUN, if available, will supersede this statically configured value	Port	
NAT Keep Alive Intvl	This is the interval at which keep alive messages are sent by the SPA to preserve a NAT mapping. It controls the interval for sending keep alive messages for the SIP signaling ports and also for the RTP ports.	Uns16	15

Regional Parameters

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

Distinctive Ring Patterns

8 cadence scripts can be specified to be used as building blocking to construct a ring tone in a RingScript.

Parameter Name	Description	Type	Default
Cadence 1	Cadence 1 script	CadScript	60(2/4)"
Cadence 2	Cadence 2 script	CadScript	60(.3/.2, 1/.2,.3/4)"
Cadence 3	Cadence 3 script	CadScript	60(.8/.4,.8/4)
Cadence 4	Cadence 4 script	CadScript	60(.4/.2,.3/.2,.8/4)
Cadence 5	Cadence 5 script	CadScript	60(.4/.2,.3/.2,.8/4)
Cadence 6	Cadence 6 script	CadScript	60(.4/.2,.3/.2,.8/4)
Cadence 7	Cadence 7 script	CadScript	60(.4/.2,.3/.2,.8/4)
Cadence 8	Cadence 8 script	CadScript	60(.4/.2,.3/.2,.8/4)

Control Timer Values (sec)

Parameter Name	Description	Type	Default
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

Notes:
1. 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.

Vertical Service Activation Codes

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
Call Back Act Code	Callback when the last outbound call is not busy	ActCode	*66
Call Back Deact Code	Cancel callback	ActCode	*86
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
Block 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
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" a call to an application server to perform further processing, such as call park.
	
	Notes:
		Str79	

	- 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 *code in the dial plan and the SPA will send INVITE *73@..... as usual when user dials *73.
		Str79	

Notes:
1. These codes automatically appended to the dial-plan. It is therefore not necessary to explicitly include them in dial-plan.

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. It is therefore not necessary to explicitly include them in dial-plan.

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	
Set 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
Time Offset (HH/mm/ss)	Additional time offset to apply to form the local time. A – sign can be added at the beginning to indicate the offset should be subtracted from the	Str9	

	current time to form the local time; otherwise the offset is added.		
Daylight Saving Time Rule	If a rule is specified, the SPA will automatically adjust for daylight saving time. See "Daylight Saving Time" Section on syntax of this rule	Daylight Saving Time Rule	
DTMF Playback Level	Local DTMF playback level in dBm (up to 1 decimal place)	PwrLevel	-26
DTMF Playback Length	Local DTMF playback duration in seconds (up to 3 decimal places)	Time3	.1

Phone Parameters

General

Parameter Name	Description	Type	Default
Station Name	A name to identify this station (reserved for future use).	Str31	
Preferred Audio Device	Specifies whether to use the headset or the speaker when pressing a Line key to make a new call or to answer an incoming call. Choices = {Speaker, Headset}.	Choice	Speaker
Voice Mail Number	A phone number or URL to check voice mail.	Str79	
Text Logo	To set the text logo to display when the phone boots up. Default is (blank) which will show "Sipura Technology". Up to 2 lines of text can be setup; each line should contain less than 32 characters. A newline character (\n) must be inserted between the 2 lines and escaped with %0a. For example, "Super%0aTelecom".	Str32	

Line Key *n*, *n* = 1–4
Parameters in this section must be appended with [*n*] in the configuration profile

Parameter Name	Description	Type	Default
Extension	Select an extension to be used for this Line key. Choices are {1,2,3,4} for the 4-Line version and {1,2} for the 2-Line version.	Choice	1
Short Name	A short label to be shown on the LCD display for the Line Key *n*	Str15	
Share Call Appearance	Yes indicates Line Key *n* is a shared call appearance. Otherwise this call appearance is not shared (a.k.a. private)	Bool	no

Line Key LED Pattern
Each parameter in this section represents the call appearance state where the specified LED pattern should apply

Parameter Name	Description	Type	Default
Idle	LED pattern during the Idle state, where the call appearance is not is in use and is available to make a new call. Leaving this entry blank indicates the default value of "c=g"	LedScript	
Local Seized	LED pattern during the Local Seized state, where this station has seized the call appearance to prepare for a new outbound call. Leaving this entry blank indicates the default value of "c=r"	LedScript	

Local Progressing	LED pattern during the Local Progressing state, where this station is attempting on this call appearance an outgoing call that is in proceeding (i.e. the called number is ringing). Leaving this entry blank indicates the default value of "c=r"	LedScript	
Local Ringing	LED pattern during the Local Ringing state, when the call appearance is ringing. Leaving this entry blank indicates the default value of "c=r;p=f"	LedScript	
Local Active	LED pattern during the Local Active state, where the call appearance is engaged in an active call. Leaving this entry blank indicates the default value of "c=r"	LedScript	
Local Held	LED pattern during the Local Held state, where the call appearance is held by this station. Leaving this entry blank indicates the default value of "c=r;p=s"	LedScript	
Remote Undefined	LED pattern during the Remote Undefined state, where the shared call state is undefined (the station is still waiting for the state information from the application server). Not applicable if the call appearance is not shared. Leaving this entry blank indicates the default value of "c=r;p=d"	LedScript	
Remote Seized	LED pattern during the Remote Seized state, where the shared call appearance is seized by another station. Not application if the call appearance is not shared. Leaving this entry blank indicates the default value of "c=r;p=d"	LedScript	
Remote Progressing	LED pattern during the Remote Progressing state, where another station is attempting on this shared call appearance an outbound call that is progressing. Not applicable if the call appearance is not shared. Leaving this entry blank indicates the default value of "c=r;p=d"	LedScript	
Remote Ringing	LED pattern during the Remote Ringing state, where the shared call appearance is in ringing on another station. Not applicable if the call appearance is not shared. Leaving this entry blank indicates the default value of "c=r;p=d"	LedScript	
Remote Active	LED pattern during the Remote Active state, where another station is engaged in an active call on this shared call appearance. Not applicable is this call appearance is not shared. Leaving this entry blank indicates the default value of "c=r;p=d"	LedScript	
Remote Held	LED pattern during the Remote Held state, where another station has placed this call appearance on hold. Not applicable if the call appearance is not shared. Leaving this entry blank indicates the default value of "c=r;p=s"	LedScript	
Disabled	LED pattern when the Call Appearance is disabled (not available for any incoming or outgoing call). Leaving this entry blank indicates the default value of "c=o"	LedScript	
Register Failed	LED pattern when the corresponding extension has failed to register with the proxy server. Leaving this entry blank indicates the default value of "c=a"	LedScript	
Registering	LED Pattern when the corresponding extension is trying to register with the proxy server. Leaving this entry blank indicates the default value of "c=r;p=s"	LedScript	
Call Back Active	Call Back operation is currently active on this call appearance	LedScript	

Supplementary Services

Enable or disable the corresponding supplementary services on the phone.

Parameter Name	Description	Type	Default
Conference	Enable/disable 3-way conference service	Bool	Yes
Attn Transfer	Enable/disable attended-call-transfer service	Bool	Yes

Blind Transfer	Enable/disable blind-call-transfer service	Bool	Yes
Do Not Disturb	Enable/disable do-not-disturb service	Bool	Yes
Block Anonymous Call	Enable/disable block-anonymous-call service	Bool	Yes
Call Back	Enable/disable call-back (a.k.a. repeating dialing) service	Bool	Yes
Block Caller ID	Enable/disable blocking outbound Caller-ID service	Bool	Yes
Secure Call	Enable/disable secure-call service	Bool	Yes
CFWD All	Enable/disable call-forward-all service	Bool	Yes
CFWD On Busy	Enable/disable call-forward-on-busy service	Bool	Yes
CFWD On No Answer	Enable/disable call-forward-on-no-answer service	Bool	Yes

Ring Tone

Each entry defines a ring tone to be used on the phone, with an ID between 1 and 10. The ID can be used in a DirEntry to indicate which ring tone to use when the corresponding caller calls.

Parameter Name	Description	Type	Default
Ring 1	Ring tone script for Ring 1	RingScript	n=Classic-1;w=3;c=1
Ring 2	Ring tone script for Ring 2	RingScript	n=Classic-2;w=3;c=2
Ring 3	Ring tone script for Ring 3	RingScript	n=Classic-3;w=3;c=3
Ring 4	Ring tone script for Ring 4	RingScript	n=Classic-4;w=3;c=4
Ring 5	Ring tone script for Ring 5	RingScript	n=Simple-1;w=2;c=1
Ring 6	Ring tone script for Ring 6	RingScript	n=Simple-2;w=2;c=2
Ring 7	Ring tone script for Ring 7	RingScript	n=Simple-3;w=2;c=3
Ring 8	Ring tone script for Ring 8	RingScript	n=Simple-4;w=2;c=4
Ring 9	Ring tone script for Ring 9	RingScript	n=Simple-5;w=2;c=5
Ring 10	Ring tone script for Ring 10	RingScript	n=THx;w=4;c=7

Extension 1 – 4 Parameters

Each Extension has its own set of Ext configuration parameters. In a configuration profile, the parameters listed in this section must be appended with [n], where n = 1,2,3,4, to indicate which extension the parameter belongs to.

General

Parameter Name	Description	Type	Default
Line Enable	Enable this extension for service	Bool	Yes
Share Ext	Indicates whether this extension is to be shared with other stations. If the extension is not shared, then a call appearance assigned to this extension is not shared, regardless the setting of <Share Call Appearance> for that call appearance. If the extension is shared, then whether or not a call appearance assigned to this extension is shared follows the setting of <Share Call Appearance> for that call appearance.	Bool	Yes

NAT Settings

Parameter Name	Description	Type	Default
NAT Mapping Enable	If set to yes, the SPA will substitute private address/port with external address/port where appropriate. Otherwise, the SPA will use the private address/port only.	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. If value is $PING, a PING message is sent. For all other values, the string is sent as specified.	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

Network Settings

Parameter Name	Description	Type	Default
SIP TOS/DiffServ Value	TOS/DiffServ field value in UDP IP Packets carrying a SIP Message	Byte	0xb8
RTP TOS/DiffServ Value	TOS/DiffServ field value in UDP IP Packets carrying a RTP data	Byte	0xb8
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 minimum. If the jitter level is set to high, then the rate of buffer size decrement is slower (more conservative), else faster (more aggressive).	Choice	High

Call Feature Settings

Parameter Name	Description	Type	Default
Blind Attn-Xfer Enable			
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	
Message Waiting			
Default Ring			
Conference Bridge URL			

SIP Settings

Parameter Name	Description	Type	Default
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 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
SIP Proxy-Require			
Auth Resync-Reboot			
SIP Remote-Party-ID			
SIP Debug Option	None, 1-line, full, exclude OPTIONS, exclude REGISTER, exclude NOTIFY, …	Choice	

Proxy and Registration

Parameter Name	Description	Type	Default
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

Parameter Name	Description	Type	Default
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

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
Silence Supp Enable	Enable silence suppression so that silent audio frames are not transmitted	Bool	No
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
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 or INFO based on outcome of codec negotiation	{InBand, AVT, INFO Auto}	Auto
DTMF Process AVT	If set to yes, SPA will include AVT payload type in its SDP and will process AVT (DTMF) packets sent by the peer. Else, it will not include AVT payload type in the SDP.	Bool	Yes
Release Unused Codec	If set to yes, SPA will relinquish all the low bit rate codecs included in the original SDP that are not supported by the called party; hence other concurrent calls can reuse these	Bool	Yes

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	low bit rate codecs (such as g729a). Otherwise, the SPA will not relinquish these resources until the end of the call even if the called party did not indicate support for these codecs		

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.

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.

User Parameters

Call Forward

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

Speed Dial

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	

Supplementary Service Settings

Parameter Name	Description	Type	Default
Block CID	Block Caller ID on/off for all calls	Bool	No
Block ANC	Block Anonymous Calls on or off	Bool	No
DND	DND on or off	Bool	No
Secure Call Setting	If set to yes, all outbound calls are secure calls by default	Bool	No
Dial Assistance	Enable/disable dialing assistance. If enabled, the phone will show up to 10 potential matches from the Redial List (most recent first) and from the Personal Directory as the user is dialing the target number. This feature applies to both off-hook (with dial tone) and on-hook (without dial tone) dialing. The user can scroll down the list using the Up/Down key to select the desired target number from the list	Bool	No

Audio Volume

Parameter Name	Description	Type	Default
Ringer	Higher = Louder. Range is 1-16	Uns8	8
Speaker	Higher = Louder. Range is 1-16	Uns8	8
Handset	Higher = Louder. Range is 1-16	Uns8	8
Headset	Higher = Louder. Range is 1-16	Uns8	

Info Parameters (Read Only)

System Information	
DHCP	Enabled / Disabled
Current IP	Current IP address of 841
Host Name	Hostname of the 841
Domain	Domain name used by 841
Current Netmask	Current Network Mask
Current Gateway	Current Gateway
Primary DNS	Primary DNS server
Secondary DNS	Secondary DNS servers
Product Information	
Product Name	Production Name: SPA-841
Serial Number	Serial number of this device
Software Version	Firmware version running on 841
Hardware Version	Hardware version of this device
MAC Address	MAC address
Client Certificate	Installed / Not Installed
Phone 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 Recv	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 Recv	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
Ext 1 – 4 Status (One Section Per Extension)	
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
Mapped SIP Port	NAT Mapped SIP Port
Call 1 – 4 Status (One Section Per Call; each Call corresponds to a Line Key)	
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
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
Downloaded Ring Tone	
Status	Indicates whether the phone is downloading a ring tone (and from where) or idle
Ring Tone 1	Information about the user downloaded ring tone 1: name, size, and time-stamp of the tone
Ring Tone 2	Information about the user downloaded ring tone 2: name, size, and time-stamp of the tone

11   Dial Plan

The SPA allows each extension 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 (unique per extension)

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 (independent for each extension):

ParName:	Dial_Plan[1], Dial_Plan[2], Dial_Plan[3], Dial_Plan[4]
Default:	(*xx | [3469]11 | 0 | 00 | [2-9]xxxxxx | 1xxx[2-9]xxxxxxS0 | xxxxxxxxxxx.)

The Dial_Plan parameters contain the actual dial plan scripts for each extension.

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 )
```

12   Provisioning Overview

Provisioning

This section contains information regarding the steps that a network administration should take when setting up a provisioning system for large numbers of IP telephones and/or terminal adaptors.   An additional reference document is the Sipura SPA Provisioning Guide, which provides detailed information on provisioning requirements.

Provisioning Capabilities

The SPA-841 provides for secure provisioning and remote upgrades. The following provides an overview of the basic functionality and requirements for provisioning IP telephones:

- Provisioning is achieved through configuration profiles requested by the device from a provisioning server, via TFTP, HTTP or HTTPS.

- No end-user intervention is required to initiate or complete a profile update or firmware upgrade. The SPA-841 is programmed to resync with the server on power-up and periodically thereafter.

- Remote upgrade is also achieved via TFTP, HTTP or HTTPS.

- General purpose parameters are provided as an additional aid to service providers in managing the provisioning process. These parameters are often used to hold encryption keys, file path locations, or provisioning state.

- The SPA can be configured to resync its internal configuration state to a remote profile periodically and on power up. Additional triggers are available, including SIP NOTIFY messages, registration state, and provisioning state.

- 256-bit symmetric key encryption of profiles.

- Supports targeted (end-point unique) key-less profile encryption.

- The SPA 841 supports a secure first-time provisioning mechanism using SSL functionality. For this purpose, each SPA-841 is loaded with a unique client certificate at manufacturing time. The certificate identifies each device by MAC address, Serial Number, and Product Name. This feature can be used in conjunction with a properly configured HTTPS provisioning server, to provide for an additional level of security in provisioning units.

- Profile resyncs and firmware upgrades are attempted only when the SPA is idle, as they may trigger a software reboot, as a result of changes in parameter values. A configurable override delay is available to force a resync after a predetermined grace period.

Configuration Profile

The SPA-841 configuration profile can take one of two forms: a compact binary file, or an XML text file. By convention, the profile is named with the extension ".cfg" (e.g. spa841.cfg). In the case of a binary profile, the Sipura Profile Compiler tool (SPC) is provided by Sipura for compiling a plain-text file containing parameter-value pairs into a properly formatted (and optionally encrypted) CFG file. The spc tool is available from Sipura for the Win32 environment (spc.exe), and the Linux-i386-elf environment (spc-linux-i386-static). In the case of an XML text profile, the XML file itself is fed to the SPA-841 directly, without prior compilation into a binary form. The XML text profile function is an advanced feature, described in the SPA Provisioning Guide.

The syntax of the source plain-text file accepted by the Sipura Profile Compiler is a series of parameter-value pairs, with the value enclosed in double quotes. Each parameter-value pair is followed by a semicolon, thus:

 Parameter_Name "parameter value";

For example:

  Proxy[1] "sip.service.com";
  Register_Expires[1] "900";

A sample source plain-text profile can be generated from the SPC tool with the following command-line invocation:

  spc.exe --sample-profile sample.txt

The sample.txt can then be edited as appropriate for the administrator's network, and compiled into a binary profile with this SPC command-line invocation:

  spc.exe sample.txt sample.cfg

If no quoted value is specified for a parameter (or if a parameter specification is missing entirely from the source plain-text file) the value of the parameter will remain unchanged in the SPA-841 when the SPA-841 resyncs to the compiled binary profile.

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.

Typical usage would be as in the following example:

Par1 "123" ;
Par2 ! ;

Here, Par1 is managed by the administrator through remote provisioning, and is only visible in the admin-login webUI (http://spa-ip-addr/admin/advanced). On the other hand, Par2 is managed by the end-user, is visible in the user-login webUI (http://spa-ip-addr/advanced), and is unaffected by periodic resyncs to the provisioning server.

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

In its most basic configuration, the SPA-841 resyncs to the provisioning server periodically, downloading the configuration profile specified as a URL in the Profile_Rule parameter.

Eample:

Profile_Rule "http://my.local.server.net/profile-path/spa841.cfg" ;

Please refer to the Sipura SPA Provisioning Guide for a more exhaustive description of the available provisioning features in the SPA-841.

13   Provisioning Flow

Provisioning a pool of SPA841 phones consists of two main steps:
- Configuring all devices to their common provisioning environment, and
- Configuring each device to their unique network attributes.

The first step (also known as pre-provisioning) includes programming initial parameters such as a resync URL, a resync period, and an administrator password.

The second step (provisioning proper) manages the SPA841 under normal provisioned conditions, configuring all pertinent parameters for normal operation unique to each device.

Pre-Provisioning

New units need to be told where to find their configuration within the customer's network. This step can be automated within a local network. By default, each SPA841 will resync out-of-the-box to the configuration profile spa841.cfg located in the virtual root directory of the TFTP boot server specified by the Local Area Network's DHCP parameters (DHCP option 66). This shared configuration profile can redirect the SPA841 to its eventual home URL.

This pre-provisioning step is typically performed in a controlled environment. For sufficiently large volume orders, Sipura can customize the pre-provisioning parameters into each unit at the time of manufacture.

For instance, pre-provisioning can be as simple as configuring only the Profile Rule parameter with the final URL of the phone's profile.

For example:

Profile_Rule "http://prov.voip.one.com/spa841/profile/$MA.cfg" ;

Here, the macro variable $MA expands to the MAC address of each specific SPA841 (e.g. $MA.cfg would expand to 000e08dabcde.cfg for a unit with that MAC address).

Provisioning Proper

Once the SPA841 is pre-provisioned it will resync periodically to the specified profile URL, maintaining its internal configuration synchronized with the provisioning server. Typically, a separate configuration file is maintained for each SPA841 in the network.

The periodic profile would contain individual information about each phone, including account identifiers and credentials.

Please refer to the Sipura SPA Provisioning Guide for a more exhaustive description of the available provisioning features in the SPA-841.

14   Firmware Upgrade

The SPA is firmware upgradeable via TFTP or 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. spa841.bin)

Firmware upgrades are attempted only when the SPA is idle, since they trigger a software reboot. During a firmware upgrade the phone is out of service and cannot be used to initiate or receive calls. The LEDs flash during the course of an upgrade, providing progress indication.

To initiate a (remotely provisioned) upgrade, program the URL of the firmware BIN file into the Upgrade_Rule parameter. The SPA841 will respond by downloading the file and performing the upgrade. Subsequently, a new upgrade will not be attempted again until the contents of Upgrade_Rule are modified to a different URL (such as by changing the firmware file name).

Example:

Upgrade_Rule "http://prov.service.com/spa841/spa841-00-09-01.bin" ;

Using the above strategy, the SPA841 might attempt to upgrade again if it is redirected to a different file (for example, if the DNS entries change and the server name resolves to a new IP address). To prevent spurious upgrades, it is better to condition the upgrade on a specific target firmware version number. The SPA-841 can be directed to upgrade to a specific version, by indicating the version number before the URL. The following examples illustrate.

This will upgrade to 1.0.3:

Upgrade_Rule "(!1.0.3)? http://call.me.com/firmware/spa841-01-00-03.bin" ;

This will upgrade to 2.1.4(d):

Upgrade_Rule "(!2.1.4(d))? Tftp://192.168.20.100/phones/spa841-02-01-04-d.bin" ;

Note that the downloaded file at the specified URL must match the version specified in the conditional expression.

Please refer to the Sipura SPA Provisioning Guide for a more exhaustive description of the available provisioning features in the SPA-841.

Premium Features

Premium functionality in the SPA-841 is enabled via license keys, that can be obtained from Sipura Technology. Specifically, the 4 independently configurable extension capability can be enabled with such a key. This is an "upgrade" from the 2 independently configurable extension capability that comes with the default SPA-841.

The keys consist of a sequence of letters and digits, organized in groups of 4, separated by the '-' character. Each key is unique to a specific phone. To enable the premium feature in a phone, program the key supplied for that phone into the License_Keys parameter, either via the phone's webUI, or via remote provisioning. This operation need only be performed once. Thereafter, the premium feature is enabled on that phone. The License_Keys parameter is write-only, and will revert to the empty state after every submit or resync operation. If the key is programmed via a remote profile resync, it is advisable that the License_Keys parameter be emptied in the profile stored on the server once the key has been installed in the phone.

To obtain premium feature license keys for the SPA-841, please contact Sipura Technology: sales@sipura.com.

15   Functional URLs for Upgrades, Reboot and Resynch

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

Upgrade URL

Through upgrade URL the user can upgrade the SPA to a firmware specified by the URL. Note: If the value of "upgrade enable" parameter in Provisioning tab is no, the user cannot upgrade the SPA even if the web page tells you that the upgrade will be done when it is not in use. See **14** 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**

Resync URL

Through Resync URL the user 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.

- 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

Reboot URL

Through the Reboot URL, the user 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

16   Performance Reporting Features

Call Statistics and 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

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.

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.

- 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.

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.

17   Troubleshooting

Phone does not "turn on" or "boot up".

If the phone does not turn-on or boot-up, please execute the following steps
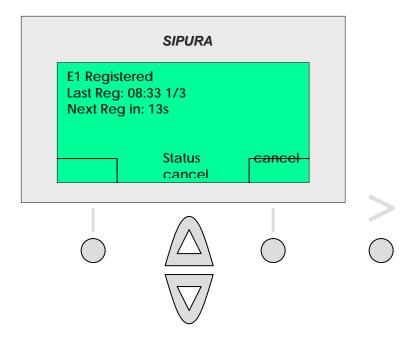
Step 1 - Check all the connections into the phone (5 Volt Power supply, RJ 45 Ethernet connection, and Handset cord.)

Step 2 - Check the Wall socket and confirm that AC power is available.   Try plugging another electrical device into the socket to confirm power.

Step 3 - Contact Network Administrator or Service Provider. Tell them that the phone does not boot up and that all the connections including the AC have been checked.   Please have the Model: SPA 841, Serial Number and telephone number ready for the support personnel. The Serial number is printed on the bottom of the phone.

Phone will not make or receive calls.

If the phone will not make or receive calls, please execute the following steps.

Step 1 - Please press "i" button, then "11", then "2". This will provide the registration information for Extension 1 of the phone. The screen should display "E1 Registered" and other information in the format provided below. There may be a problem and the screen will display "Not Registered".

SIPURA

E1 Registered
Last Reg: 08:33 1/3
Next Reg in: 13s

Status
cancel   cancel

>

Step 2 – Write down if the phone is "Registered" or "Not Registered". To exit from this menu option, press the "i".

Step 3 – Get the Current IP Address of the phone. To get the Current IP Address, please press "i", "9", "2". The phone will display the Current IP Address and please write it down. To exit from this menu option, press the "i" button.

Step 4 - Contact the network administrator or service provider. Please have the Model: SPA 841, Serial Number, telephone number, Current IP Address and Registration Status (Registered, Not Registered) ready for the support personnel. The Serial number is printed on the bottom of the phone.

Calls with Poor Voice Quality.

The source of this issue often is one of two sources:

- The phone or call controller may have be configured improperly.
- The transport network has performance issues.

Please try to gather the following information

Step 1 - Identify when and where the issue occurred. Please look at the Redial List (calls that the user made) by pressing "i", "3", "1" and write down the calls that were affected. Then look at the Answered Calls by pressing "i", "3", "2" and write down the calls that were affected.

Step 2 - Identify the type of voice quality issue i.e. part of conversations dropped out or were lost, excessive echo, unclear or garbled speech, one way conversation, low volume

Step 3 – Get the Current IP Address of the phone. To get the Current IP Address, please press "i", "9", "2". The phone will display the Current IP Address and please write it down. To exit from this menu option, press the "i" button.

Step 4 - Contact the network administrator or service provider. Be prepared to provide the call log information and the voice quality issue type. Additionally, please have the Model: SPA 841, Serial Number, telephone number, Current IP Address and Registration Status (Registered, Not Registered) ready for the support personnel. The Serial number is printed on the bottom of the phone.

18   Customer Support Methods

For support on the SPA-841 IP Telephone, please use the following methods:

First, contact the VAR dealer from whom you purchased your phone(s). Authorized service providers and VAR dealers may access the Sipura web site at http://www.sipura.com/support/index.htm. This site contains FAQs and other technical documentation and soft tools for supporting the Sipura IP Telephone and other products.

Please see the SPA-841 User Guide for information about the configuration and settings that a typical end user will encounter. The SPA-841 User Guide can be found on the Sipura Technology web site.

http://www.sipura.com/support

Service Provider Support:
Service providers may contact Sipura Technology and receive direct support by sending e-mail to support@sipura.com.

19   Care for the SPA-841 Phone

Do not expose the phone to heat, sun, cold, water.

The SPA-8421 phone is an electronic device and should not be exposed to heat, sun, cold or water. A rule of thumb is that if its too hot or cold for you to be comfortable, its not a good place for the phone either. Do not expose the phone to water, as this will create a shock hazard.

Cleaning the phone
To clean the phone, please use a lightly moistened paper towel. Do not spray or pour cleaning solution directly onto the phone.

20 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
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
FQDN	Fully Qualified Domain Name
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
MWI	Message Waiting Indication
OSI	Open Switching Interval
PCB	Printed Circuit Board
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
SDP   Session Description Protocol
SDRAM   Synchronous DRAM
sec   seconds
SIP   Session Initiation Protocol
SP   Service Provider
SPA   Sipura IP Telephone
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

21 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.