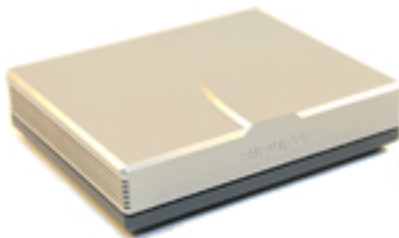




# Configuration Guide

for the

## Sipura SPA-2100



## CONNECT THE SPA-2100

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*Note:*

*Firmware 2.05(c) or 2.0.5(d) is required for service.*

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1. Using a standard telephone cable (RJ-11), connect an analog phone to the **Phone1** port on the SPA-2100.
2. Using a standard network cable (RJ-45), connect the **WAN** port on the SPA-2100 to your network connection (e.g., LAN).
3. Using a standard network cable (RJ-45), connect the **LAN** port on the SPA-2100 to the network card on your PC
4. Power on the SPA-2100 by plugging its power supply into an AC outlet and then plugging the power cable into the **Power** port on the SPA-2100.

## CONFIGURE THE SPA-2100

*To connect to the SPA-2100 Web interface and configure the SPA-2100:*

1. Using the PC connected to the SPA-2100's **LAN** port, launch a Web browser, and enter <http://192.168.0.1> in the **Address** bar.
2. Click the **Admin Login** link, located in the upper right-hand corner of the page.
3. Click the **advanced** link.
4. Click the **Voice** tab at the top of the page.
5. Click the tab for **Line1** or **Line2** at the top of the page, depending on which line you are configuring.
6. In the **NAT Settings** section, change the value of the following field:
  - **NAT Keep Alive Enable** – no
7. In the **Proxy and Registration** section, change the following fields to the value indicated:
  - **Proxy** – sip.primetalker.com
  - **Use Outbound Proxy** – no
  - **Use OB Proxy In Dialog** – no
  - **Make Call Without Reg** – yes
  - **Ans Call Without Reg** – yes
  - **DNS SRV Auto Prefix** – no
8. In the **Subscriber Information**, enter **No\_Number** in the **Display Name** field.
9. Enter the **User ID** (account number) and **Password** (passcode).

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*Note:*

*Line 1 and Line 2 must have a different account number and passcode.*

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10. Change the value of the **Use Auth ID** field to **no**.

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Sipura Phone Adapter Configuration

Router | **Voice** | Info | System | SIP | Provisioning | Regional | **Line 1** | Line 2 | User 1 | User 2 | [User Login](#) | [basic](#) | [advanced](#)

Line Enable:

**Streaming Audio Server (SAS)**  
SAS Enable:  SAS DLG Refresh Intvl:   
SAS Inbound RTP Sink:

**NAT Settings**  
NAT Mapping Enable:  NAT Keep Alive Enable:   
NAT Keep Alive Msg:  NAT Keep Alive Dest:

**Network Settings**  
SIP ToS/DiffServ Value:  SIP CoS Value:  [0-7]  
RTP ToS/DiffServ Value:  RTP CoS Value:  [0-7]  
Network Jitter Level:

**SIP Settings**  
SIP Port:  SIP 100REL Enable:   
EXT SIP Port:  Auth Resync-Reboot:   
SIP Proxy-Require:  SIP Remote-Party-ID:   
SIP Debug Option:  RTP Log Intvl:   
Restrict Source IP:  Referor Bye Delay:   
Refer Target Bye Delay:  Referee Bye Delay:   
Refer-To Target Contact:

**Call Feature Settings**  
Blind Attn-Xfer Enable:  MOH Server:   
Xfer When Hangup Conf:  Conference Bridge URL:   
Conference Bridge Ports:

**Proxy and Registration**  
Proxy:  Use Outbound Proxy:   
Outbound Proxy:  Use OB Proxy In Dialog:   
Register:  Make Call Without Reg:   
Register Expires:  Ans Call Without Reg:   
Use DNS SRV:  DNS SRV Auto Prefix:   
Proxy Fallback Intvl:

**Subscriber Information**  
Display Name:  User ID:   
Password:  Use Auth ID:   
Auth ID:   
Mini Certificate:   
SRTP Private Key:

Line 1 Configuration Page – Top

11. In the **Audio Configuration** section, change the value of the following fields:

- Use Pref Codec Only – no
- G729a Enable – yes
- G723 Enable – yes

- G726-16 Enable – no
- G726-24 Enable – no
- G726-32 Enable – no
- G726-40 Enable – no
- DTMF Process INFO--no
- DTMF Process AVT – yes
- DTMF Tx Method – AVT

**Supplementary Service Subscription**

Call Waiting Serv:	yes	Block CID Serv:	yes
Block ANC Serv:	yes	Dist Ring Serv:	yes
Cfwd All Serv:	yes	Cfwd Busy Serv:	yes
Cfwd No Ans Serv:	yes	Cfwd Sel Serv:	yes
Cfwd Last Serv:	yes	Block Last Serv:	yes
Accept Last Serv:	yes	DND Serv:	yes
CID Serv:	yes	CWCID Serv:	yes
Call Return Serv:	yes	Call Back Serv:	yes
Three Way Call Serv:	yes	Three Way Conf Serv:	yes
Attn Transfer Serv:	yes	Unattn Transfer Serv:	yes
MWI Serv:	yes	VMWI Serv:	yes
Speed Dial Serv:	yes	Secure Call Serv:	yes
Referral Serv:	yes	Feature Dial Serv:	yes

**Audio Configuration**

Preferred Codec:	G726	Silence Supp Enable:	no
Use Pref Codec Only:	no	Silence Threshold:	medium
G729a Enable:	yes	Echo Canc Enable:	yes
G723 Enable:	yes	Echo Canc Adapt Enable:	yes
G726-16 Enable:	no	Echo Supp Enable:	yes
G726-24 Enable:	no	FAX CED Detect Enable:	yes
G726-32 Enable:	no	FAX CNG Detect Enable:	yes
G726-40 Enable:	no	FAX Passthru Codec:	G711u
DTMF Process INFO:	no	FAX Codec Symmetric:	yes
DTMF Process AVT:	yes	FAX Passthru Method:	NSE
DTMF Tx Method:	AVT	FAX Process NSE:	yes
Hook Flash Tx Method:	None	Release Unused Codec:	yes

**Dial Plan**

Dial Plan: (\*xx[[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0]xxxxxxxxxxxxx.)

Enable IP Dialing: no

**FXS Port Polarity Configuration**

Idle Polarity:	Forward	Caller Conn Polarity:	Forward
Callee Conn Polarity:	Forward		

Undo All Changes    Submit All Changes

User Login   basic | advanced

*Line 1 Configuration Page – Bottom*

12. In the **Dial Plan** section, change the value of the **Dial Plan** field to match the dialing pattern used in your country (where you are using the PrimeTalker service).

For US calls, insert “1” as the first digit. For non-US calls, insert “011” before the rest of the phone number. For information on customizing your dial plan, refer to the SPA-2100 manual.

13. To save these settings, click the **Submit All Changes** button.