

Cisco SPA112 Adapter Setup Guide

Login to the Cisco Phone Adapter Configuration Utility with the below credentials.

- Username: admin
- Password: admin

Quick Setup Tab

Line 1

1. Proxy: 'virtualpbx.net'
2. Display Name: label the phone as needed
3. User ID: 4081231234 (example)
4. Password: contact VirtualPBX Support for assistance

Line 2

Contact Support for assistance with Line 2

The screenshot shows the Cisco Phone Adapter Configuration Utility interface. The top navigation bar includes 'Quick Setup', 'Network Setup', 'Voice', 'Administration', and 'Status'. The 'Quick Setup' tab is active. On the left, a sidebar contains a red-bordered box with the text 'Enter Proxy, User ID, and Password information here' and an arrow pointing to the 'Proxy' field of the 'Line 1' configuration. The 'Line 1' configuration fields are: Proxy (virtualpbx.net), Display Name (VPBX), User ID (4081231234), Password (masked with asterisks), and Dial Plan ((*xx[[3469]11]0[00][2-9]xxxxxx[1xxx[2-9]xxxxxxS0]xxxxxxxxxxxx.)). The 'Line 2' configuration fields are empty.

Save when finished

Voice Tab

Click on Line1 from the left to configure

1. Enable NAT Keep Alive
2. Disable SIP Remote-Party-ID

Phone Adapter Configuration Utility

admin(Admin) Log Out About Help

Quick Setup Network Setup **Voice** Administration Status

Information System SIP Provisioning Regional **Line 1** User 1 Line 2 User 2

Line 1

General	
Line Enable:	<input type="button" value="yes"/>
Streaming Audio Server (SAS)	
SAS Enable:	<input type="button" value="no"/>
SAS Inbound RTP Sink:	<input type="text"/>
SAS DLG Refresh Intvl:	<input type="text" value="30"/>
NAT Settings	
NAT Mapping Enable:	<input type="button" value="no"/>
NAT Keep Alive Enable:	<input type="button" value="yes"/>
NAT Keep Alive Msg:	<input type="text" value="NOTIFY"/>
NAT Keep Alive Dest:	<input type="text" value="SPROXY"/>
Network Settings	
SIP ToS/DiffServ Value:	<input type="text" value="0x68"/>
SIP CoS Value:	<input type="text" value="3"/> [0-7]
RTP ToS/DiffServ Value:	<input type="text" value="0xb8"/>
RTP CoS Value:	<input type="text" value="6"/> [0-7]
Network Jitter Level:	<input type="button" value="high"/>
Jitter Buffer Adjustment:	<input type="button" value="yes"/>

SIP Settings

SIP Transport:	<input type="button" value="UDP"/>	SIP Port:	<input type="text" value="5060"/>
SIP 100REL Enable:	<input type="button" value="no"/>	EXT SIP Port:	<input type="text"/>
Auth Resync-Reboot:	<input type="button" value="yes"/>	SIP Proxy-Require:	<input type="text"/>
SIP Remote-Party-ID:	<input type="button" value="no"/>	SIP GUID:	<input type="button" value="no"/>
SIP Debug Option:	<input type="button" value="none"/>	RTP Log Intvl:	<input type="text" value="0"/>
Restrict Source IP:	<input type="button" value="no"/>	Referor Bye Delay:	<input type="text" value="4"/>
Refer Target Bye Delay:	<input type="text" value="0"/>	Referee Bye Delay:	<input type="text" value="0"/>
Refer-To Target Contact:	<input type="button" value="no"/>	Sticky 183:	<input type="button" value="no"/>
Auth INVITE:	<input type="button" value="no"/>	Reply 182 On Call Waiting:	<input type="button" value="no"/>
Use Anonymous With RPID:	<input type="button" value="yes"/>	Use Local Addr In FROM:	<input type="button" value="no"/>

Proxy and Registration

Proxy:

Outbound Proxy:

Use Outbound Proxy:

Register:

Register Expires:

Use DNS SRV:

Proxy Fallback Intvl:

Mailbox Subscribe URL:

Use OB Proxy In Dialog:

Make Call Without Reg:

Ans Call Without Reg:

DNS SRV Auto Prefix:

Proxy Redundancy Method:

Mailbox Subscribe Expires:

Subscriber Information

Display Name:

User ID:

Password:

Use Auth ID:

Auth ID:

Resident Online Number:

SIP URI:

Proxy and Subscriber Information should be auto-populated from the first tab.

May define Auth ID as the User ID

Audio Configuration

Preferred Codec:

Second Preferred Codec:

Third Preferred Codec:

Use Pref Codec Only:

Use Remote Pref Codec:

Codec Negotiation:

G729a Enable:

Silence Supp Enable:

G726-32 Enable:

Silence Threshold:

FAX V21 Detect Enable:

Echo Canc Enable:

FAX CNG Detect Enable:

FAX Passthru Codec:

FAX Codec Symmetric:

DTMF Process INFO:

FAX Passthru Method:

DTMF Process AVT:

FAX Process NSE:

DTMF Tx Method:

FAX Disable ECAN:

DTMF Tx Mode:

DTMF Tx Strict Hold Off Time:

FAX Enable T38:

Hook Flash Tx Method:

FAX T38 Redundancy:

FAX T38 ECM Enable:

FAX Tone Detect Mode:

Symmetric RTP:

FAX T38 Return to Voice:

Modem Line:

Dial Plan

Dial Plan:

FXS Port Polarity Configuration

Idle Polarity:

Caller Conn Polarity:

Callee Conn Polarity:

Disable codec G726-32

Set Codec to G729

Save when finished

Connect the analog phone or fax machine to this device and the configuration is complete. Contact VirtualPBX Support for any questions or suggestions.