

Cisco SPA 112 2-Port Phone Adapter

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The SPA 112 connects 2 analog phones to a VoIP service provider.



SPA 112 Setup

1. Connect the SPA 112 to your network according to the instructions included with the device. When the IP address is discovered, enter it in the address bar of a web browser to go to the Cisco Phone Adapter Device Configuration Utility.

NOTE: Attaching an analog phone to one of the ports, you can use the integrated IVR to obtain the IP address as follows:

- Go off-hook and press **** in quick succession.
- When prompted to enter the option, dial 110#; the IVR will then play the current IP address.

2. Choose Quick Setup on the menu bar:

The screenshot shows the Cisco Phone Adapter Configuration Utility web interface. The 'Quick Setup' tab is selected. The interface displays configuration fields for Line 1 and Line 2. Line 1 fields include Proxy (66.171.194.10), Display Name (27743), Password (masked), User ID (69000327743), and Dial Plan ([2-9]11[03]011xxxx.T[9]1[2-9]xxxxxxxxx[1-8]xxxx). Line 2 fields include Proxy, Display Name, Password, User ID, and Dial Plan (*xxx[3469]11[0]00[2-9]xxxxxx[1xxx[2-9]xxxxxxS0]xxxxxxxxxxxxxx).

For **Line 1**, enter:

- **Proxy:** IP address of SBC **66.171.194.10**
- **Display Name:** Extension number used for the SPA 112.
- For the values to enter for the following user settings, [launch the iPECS Cloud Manager Console](#) and choose User > User Setup. Select the user and scroll down to view the user's Device tab:

Device	Feature	Service	Information	DN Based CID Routing
Assigned Device				
- CISCO SPA112 (User ID : 69000327743)		Authentication ID	<input type="text" value="27743"/>	Authentication Password <input type="password" value="*****"/>

- **User ID:** Enter the User ID displayed on the Device tab in parentheses.
 - **Password:** Password entered when this user was created in User Setup.
 - **Dial Plan:** Using [2-9]11|03|011xxx.T|91[2-9]xxxxxxxx|[1-8]xxxx will configure the SPA 112 with the correct dial plan format.
3. Choose Network Setup on the menu bar.
 4. Expand Basic Setup in the left pane and click Internet Settings.

Enter:

- **Connection Type:** Static IP or DHCP.
- **Internet Address:** If connection type is set to Static IP, enter the IP address of the SPA 112 on the network.
- **Subnet Mask:** If connection type is set to Static IP, enter the subnet mask of the network.
- **Default Gateway:** If connection type is set to Static IP, enter the IP address of the default gateway.

- **MTU:** Select Auto.
- **Host Name:** Name of the SPA 112 on the network.
- **Domain Name:** Name of the network domain if needed.
- **DNS Server Order:** If connection type is set to Static IP, enter the IP address of the DNS server.

If connection type is set to Static IP, click Submit to save your changes.

5. Choose Voice on the menu bar.
6. Select SIP in the left pane.

The screenshot shows the Cisco Phone Adapter Configuration Utility interface. The 'Voice' tab is selected, and the 'SIP' configuration page is displayed. The left sidebar shows the navigation menu with 'SIP' highlighted. The main content area is divided into several sections:

- SIP Timer Values (sec):** A grid of input fields for various SIP timers. SIP T1 is set to .5, SIP T4 to 5, SIP Timer F to 16, SIP Timer D to 32, INVITE Expires to 240, Reg Min Expires to 1, Reg Retry Intvl to 30, Reg Retry Random Delay to 0, and Reg Retry Intvl Cap to 0. SIP T2 is set to 4, SIP Timer B to 32, SIP Timer H to 32, SIP Timer J to 32, ReINVITE Expires to 30, Reg Max Expires to 7200, Reg Retry Long Intvl to 1200, and Reg Retry Long Random Delay to 0.
- Response Status Code Handling:** Input fields for SIT1 RSC, SIT3 RSC, Try Backup RSC, SIT2 RSC, SIT4 RSC, and Retry Reg RSC.
- RTP Parameters:** RTP Port Min is set to 16384, RTP Packet Size to 0.030, Max RTP ICMP Err to 0, and No UDP Checksum to no. RTP Port Max is set to 24383, RTP Tx Packet Size Follows Remote SDP to yes, RTCP Tx Interval to 0, and Stats In BYE to yes.

For RTP Parameters, adjust the RTP port range:

- **RTP Port Min:** 16384
- **RTP Port Max:** 24383

7. Select Line 1 in the left pane:

Phone Adapter Configuration Utility

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:	yes
Streaming Audio Server (SAS)	
SAS Enable:	no
SAS Inbound RTP Sink:	
SAS DLG Refresh Intvl:	30
NAT Settings	
NAT Mapping Enable:	no
NAT Keep Alive Msg:	\$NOTIFY
NAT Keep Alive Enable:	yes
NAT Keep Alive Dest:	\$PROXY
Network Settings	
SIP ToS/DiffServ Value:	0x68
RTP ToS/DiffServ Value:	0xb8
Network Jitter Level:	high
SIP CoS Value:	3 [0-7]
RTP CoS Value:	6 [0-7]
Jitter Buffer Adjustment:	yes
SIP Settings	
SIP Transport:	UDP
SIP 100REL Enable:	no
Auth Resync-Reboot:	yes
SIP Remote-Party-ID:	yes
SIP Debug Option:	none
Restrict Source IP:	no
Refer Target Bye Delay:	0
Refer-To Target Contact:	no
Auth INVITE:	no
Use Anonymous With RPID:	yes
SIP Port:	5060
EXT SIP Port:	5060
SIP Proxy-Require:	
SIP GUID:	no
RTP Log Intvl:	0
Referor Bye Delay:	4
Referee Bye Delay:	0
Sticky 183:	no
Reply 182 On Call Waiting:	no
Use Local Addr In FROM:	no
Call Feature Settings	
Blind Attn-Xfer Enable:	no
Xfer When Hangup Conf:	yes
Conference Bridge Ports:	3
Emergency Number:	
Feature Key Sync:	no
MDH Server:	
Conference Bridge URL:	
Enable IP Dialing:	no
Mailbox ID:	

For General, enter:

- **Line enable:** yes

For Network Settings, enter:

- **NAT Keep Alive Enable:** yes

For SIP Settings, enter:

- **SIP Transport:** UDP
- **SIP Port:** 5060
- **EXT SIP Port:** 5060

Scroll down to Proxy and Registration:

CISCO Phone Adapter Configuration Utility

Quick Setup Network Setup **Voice** Administration Status

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

Line 1

Proxy and Registration

Proxy:

Outbound Proxy:

Use Outbound Proxy:

Register:

Register Expires:

Use DNS SRV:

Proxy Fallback Intvl:

Mailbox Subscribe URL:

Auto Register When Fallover:

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:

Password:

Auth ID:

SIP URI:

User ID:

Use Auth ID:

Resident Online Number:

Supplementary Service Subscription

Call Waiting Serv:	<input type="text" value="yes"/>	Block CID Serv:	<input type="text" value="yes"/>
Block ANC Serv:	<input type="text" value="yes"/>	Dist Ring Serv:	<input type="text" value="yes"/>
Cfwd All Serv:	<input type="text" value="yes"/>	Cfwd Busy Serv:	<input type="text" value="yes"/>
Cfwd No Ans Serv:	<input type="text" value="yes"/>	Cfwd Sel Serv:	<input type="text" value="yes"/>
Cfwd Last Serv:	<input type="text" value="yes"/>	Block Last Serv:	<input type="text" value="yes"/>
Accept Last Serv:	<input type="text" value="yes"/>	DND Serv:	<input type="text" value="yes"/>
CID Serv:	<input type="text" value="yes"/>	CWCID Serv:	<input type="text" value="yes"/>
Call Return Serv:	<input type="text" value="yes"/>	Call Redial Serv:	<input type="text" value="yes"/>
Call Back Serv:	<input type="text" value="yes"/>	Three Way Call Serv:	<input type="text" value="yes"/>
Three Way Conf Serv:	<input type="text" value="yes"/>	Attn Transfer Serv:	<input type="text" value="yes"/>
Unattn Transfer Serv:	<input type="text" value="yes"/>	MWI Serv:	<input type="text" value="yes"/>
VMWI Serv:	<input type="text" value="yes"/>	Speed Dial Serv:	<input type="text" value="yes"/>
Secure Call Serv:	<input type="text" value="yes"/>	Referral Serv:	<input type="text" value="yes"/>
Feature Dial Serv:	<input type="text" value="yes"/>	Service Announcement Serv:	<input type="text" value="no"/>
Reuse CID Number As Name:	<input type="text" value="yes"/>	CONFID Serv:	<input type="text" value="yes"/>

Enter:

- **Proxy:** IP of SBC 66.171.194.10
- **Register:** yes

For Subscriber Information, enter:

- **Display Name:** Extension number used for the SPA 112.
- **Password:** Password entered when this user was created in User Setup.
- **Auth ID:** Authentication ID displayed on the user's Device tab in the Manager Console.
- **User ID:** User ID displayed on the user's Device tab in the Manager Console.

Device Feature Service Information DN Based CID Routing

Assigned Device

- CISCO SPA112 (User ID : 69000327743) Authentication ID Authentication Password

Scroll down to Audio Configuration:

The screenshot displays the Cisco Phone Adapter Configuration Utility interface for 'Line 1'. The 'Audio Configuration' section is highlighted, showing the following settings:

- Preferred Codec: G711u
- DTMF Tx Method: INFO

The 'Dial Plan' section shows the following dial plan format:

```
([2-9]11|03|011xxx.T|91[2-9]xxxxxxxx[4-8]xx|[1-3]xx)
```

Enter:

- Preferred Codec: G711u
- DTMF Tx Method: INFO

For Dial Plan, using:

- [2-9]11|03|011xxx.T|91[2-9]xxxxxxxx[4-8]xx|[1-3]xx will configure the SPA with the correct dial plan format for most customers with 3 digit extensions
- [2-9]11|03|011xxx.T|91[2-9]xxxxxxxx[4-8]xx|[1-3]xxx will configure the SPA with the correct dial plan format for most customers with 4 digit extensions
- [2-9]11|03|011xxx.T|91[2-9]xxxxxxxx[4-8]xx|[1-3]xxxx will configure the SPA with the correct dial plan format for most customers with 5 digit extensions

For other configurations, please reference your Cisco SPA documentation or contact iPECS Support.