

AnyPhone Configuration Guide for the Cisco SPA112 For Fax Machine Endpoints

This documentation will provide you with the steps required to configure your Cisco SPA112 as Generic AnyPhone faxing device. Configuring the SPA112 to work with analog telephones is covered in a different document.

Note: A single Anyphone line has a maximum concurrent call capacity of 4 calls. Hardware capacity should also be taken into consideration. Cisco SPA112 can handle up to 2 concurrent calls.

This guide will cover the following steps

- Logging into the ATA
- Configuring equipment for service with Intermedia.

1. Accessing the web interface.

- Connect the adapter to a DHCP enabled network, and locate its assigned IP address, and navigate to it via a web browser.
- The default username, and password should both be “admin”
Note, if entering in the adapters IP address does not provide you with a web GUI the adapter most likely has its web interface disabled, and must be logged into via the LAN port on the back of the adapter to have remote administration enabled



2. Configuring device to register with Intermedia servers.

- Once logged into the adapter select the quick setup tab in the top left corner

Quick Setup

Line 1

Proxy:

Display Name: User ID:

Password:

Dial Plan:

Line 2

Proxy:

Display Name: User ID:

Password:

Dial Plan:

- The proxy listed will be SIP Domain defined in HostPilot
- Display name can be whatever the name of the user will be this is not critical to functionality.
- User ID, and Password will be the SIP Username, and SIP password information found under the user in HostPilot.
- Once information has been entered in click “submit” in the lower left corner.
- Once the equipment is done saving the configuration Select the “Voice” tab at the top of the interface, you should now see under line 1 it says “Registered” • Be sure to set the emulation from Generic to Cisco in hostpilot.

Line 1 Status			
Hook State:	On	Registration State:	Registered
Last Registration At:	11/7/2015 06:04:40	Next Registration In:	147 s

- The default dialplan does not support 3, or 4 digit extensions. Depending what extension scheme the customer is using you will need to alter the dial plan.

3 digit extension dial plan: (*xxx[[3469]11|0|00][2-9]xxxxxx|1xxx[2-9]xxxxxS0|xxxxxxxxxxxxx.)








4 digit extension dial plan: (*xxxx[[3469]11|0|00][2-9]xxxxxx|1xxx[2-9]xxxxxS0|xxxxxxxxxxxxx.)

3. Configuring Voice Tab

- Next you will need to select line 1 under the voice tab, and make the following changes to the ATA's configuration
- Make note of the red arrows, and confirm your adapter's configuration matches the changes marked.

***** Note: The "SIP Port" will need to be 6xxx (xxx will be the Ext number for this phone).** In the Example above you will find Ext 249 is using SIP Port: **6249**. If you do not configure this correctly it will cause audio issues or inbound call failure issues. For sites using 4 digits extensions use the last 3 digits of the extensions. E.G. 4001 would be **6001**.

Line 1

General	
Line Enable:	yes ▼
Streaming Audio Server (SAS)	
SAS Enable:	no ▼
SAS DLG Refresh Intvl:	30
SAS Inbound RTP Sink:	
NAT Settings	
NAT Mapping Enable:	no ▼
NAT Keep Alive Enable:	yes ▼ 
NAT Keep Alive Msg:	SNOTIFY
NAT Keep Alive Dest:	SPROXY
Network Settings	
SIP ToS/DiffServ Value:	0x68
SIP CoS Value:	3 [0-7]
RTP ToS/DiffServ Value:	0xb8
RTP CoS Value:	6 [0-7]
Network Jitter Level:	very high ▼ 
Jitter Buffer Adjustment:	yes ▼
SIP Settings	
SIP Transport:	UDP ▼ 
SIP 100REL Enable:	no ▼ 
Auth Resync-Reboot:	yes ▼
SIP Proxy-Require:	
SIP Remote-Party-ID:	no ▼
SIP GUID:	no ▼
SIP Debug Option:	none ▼
RTP Log Intvl:	0
Referor Bye Delay:	4
Restrict Source IP:	no ▼
Refer Target Bye Delay:	0
Refer-To Target Contact:	yes ▼
Sticky 183:	yes ▼ 
Auth INVITE:	no ▼
Reply 182 On Call Waiting:	no ▼ 
Use Anonymous With RPID:	yes ▼
Use Local Addr In FROM:	no ▼
SIP Port:	6249 
EXT SIP Port:	

Proxy and Registration			
Proxy:	usbc.telecomsvc.com:6060		
Outbound Proxy:	usbc.telecomsvc.com:6060		
Use Outbound Proxy:	yes ▼	Use OB Proxy In Dialog:	yes ▼
Register:	yes ▼	Make Call Without Reg:	no ▼
Register Expires:	20	Ans Call Without Reg:	no ▼
Use DNS SRV:	no ▼	DNS SRV Auto Prefix:	no ▼
Proxy Fallback Intvl:	20	Proxy Redundancy Method:	Based on SRV Port ▼
Mailbox Subscribe URL:		Mailbox Subscribe Expires:	2147483647
Subscriber Information			
Display Name:		User ID:	900244720
Password:	*****	Use Auth ID:	no ▼
Auth ID:		Resident Online Number:	
SIP URI:			

For sites using these adapters for faxing be sure to disable call waiting.

Supplementary Service Subscription	
Call Waiting Serv:	yes ▼
Block ANC Serv:	yes ▼
Cfwd All Serv:	yes ▼
Cfwd No Ans Serv:	yes ▼
Cfwd Last Serv:	yes ▼
Accept Last Serv:	no ▼
CID Serv:	yes ▼
Call Return Serv:	yes ▼
Call Back Serv:	no ▼
Three Way Conf Serv:	no ▼
Unattn Transfer Serv:	no ▼
VMWI Serv:	yes ▼
Secure Call Serv:	no ▼
Feature Dial Serv:	yes ▼
Reuse CID Number As Name:	yes ▼
Block CID Serv:	yes ▼
Dist Ring Serv:	no ▼
Cfwd Busy Serv:	yes ▼
Cfwd Sel Serv:	yes ▼
Block Last Serv:	yes ▼
DND Serv:	no ▼
CWCID Serv:	yes ▼
Call Redial Serv:	no ▼
Three Way Call Serv:	yes ▼
Attn Transfer Serv:	no ▼
MWI Serv:	yes ▼
Speed Dial Serv:	no ▼
Referral Serv:	yes ▼
Service Announcement Serv:	no ▼
CONFCID Serv:	yes ▼

Audio Configuration

Preferred Codec:	G711u	Second Preferred Codec:	Unspecified
Third Preferred Codec:	Unspecified	Use Pref Codec Only:	no
Use Remote Pref Codec:	no	Codec Negotiation:	Default
G729a Enable:	yes	Silence Supp Enable:	no
G726-32 Enable:	no	Silence Threshold:	medium
FAX V21 Detect Enable:	yes	Echo Canc Enable:	yes
FAX CNG Detect Enable:	yes	FAX Passthru Codec:	G711u
FAX Codec Symmetric:	yes	DTMF Process INFO:	yes
FAX Passthru Method:	RelINVITE	DTMF Process AVT:	yes
FAX Process NSE:	no	DTMF Tx Method:	AVT
FAX Disable ECAN:	no	DTMF Tx Mode:	Strict
DTMF Tx Strict Hold Off Time:	70	FAX Enable T38:	no
Hook Flash Tx Method:	None	FAX T38 Redundancy:	1
FAX T38 ECM Enable:	yes	FAX Tone Detect Mode:	caller or callee
Symmetric RTP:	no	FAX T38 Return to Voice:	no
Modem Line:	no	RTP to Proxy in Remote Hold:	no

3. Sip configuration

- Under the Sip sub option insure that the RTP parameters of your configuration match below.
- RTP port Min should be **30000**, and RTP Port Max should be **50000** • Set the RTCP Tx Interval to **10**

The screenshot shows the Cisco Phone Adapter Configuration Utility interface. The 'SIP' tab is selected, and the 'RTP Parameters' section is expanded. The following parameters are highlighted with red boxes:

RTP Port Min:	30000	RTP Port Max:	50000
RTP Packet Size:	0.030	RTP Tx Packet Size Follows Remote SDP:	yes
Max RTP ICMP Err:	0	RTCP Tx Interval:	10
No UDP Checksum:	no	Stats In BYE:	yes