AnyPhone Configuration Guide for the Cisco SPA112
For Fax Machine Endpoints

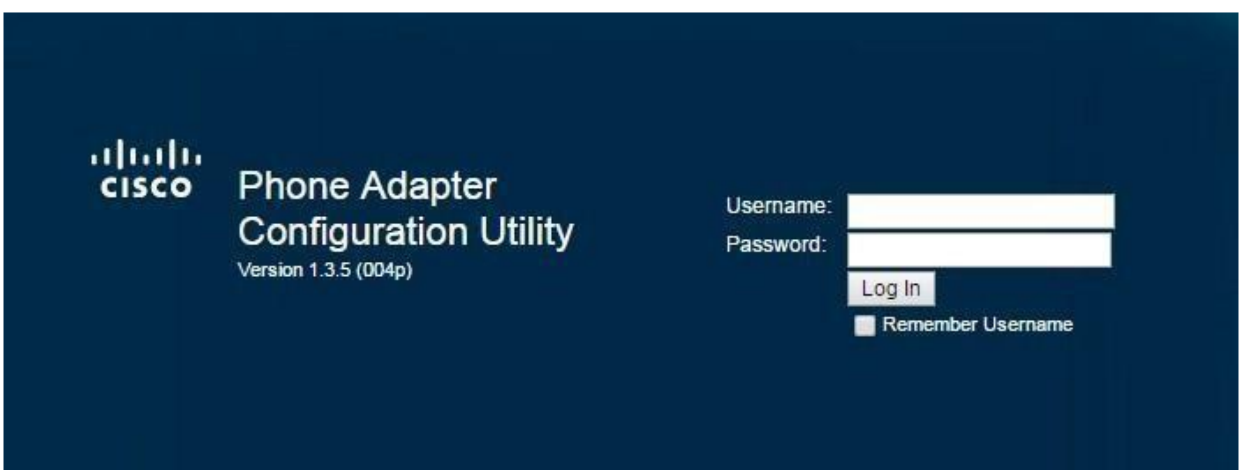
This documentation will provide you with the steps required to configure your Cisco SPA112 as Generic AnyPhone faxing device. Configuring the SPA112 to work with analog telephones is covered in a different document.

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.

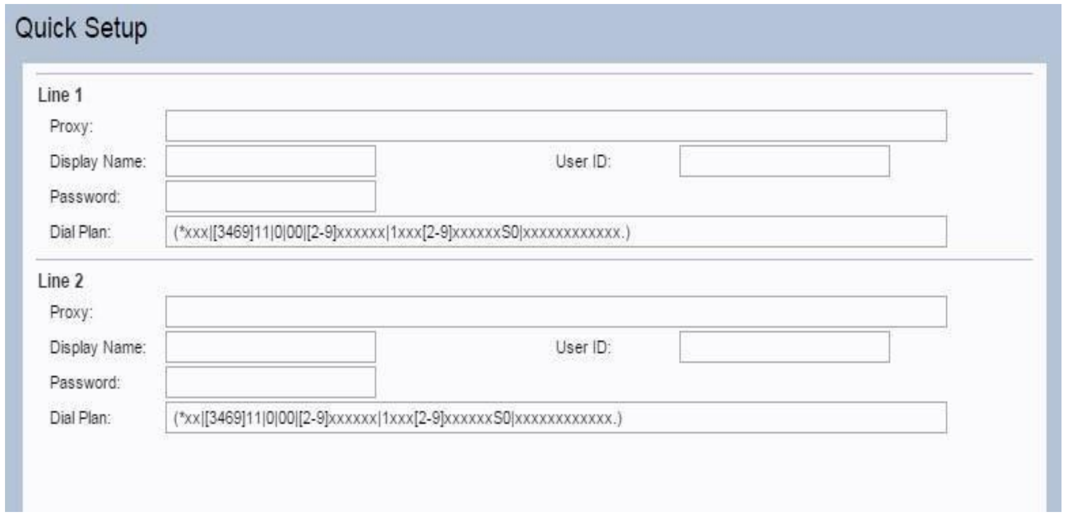
This guide will cover the following steps

- 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 most 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



- 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.

- 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.

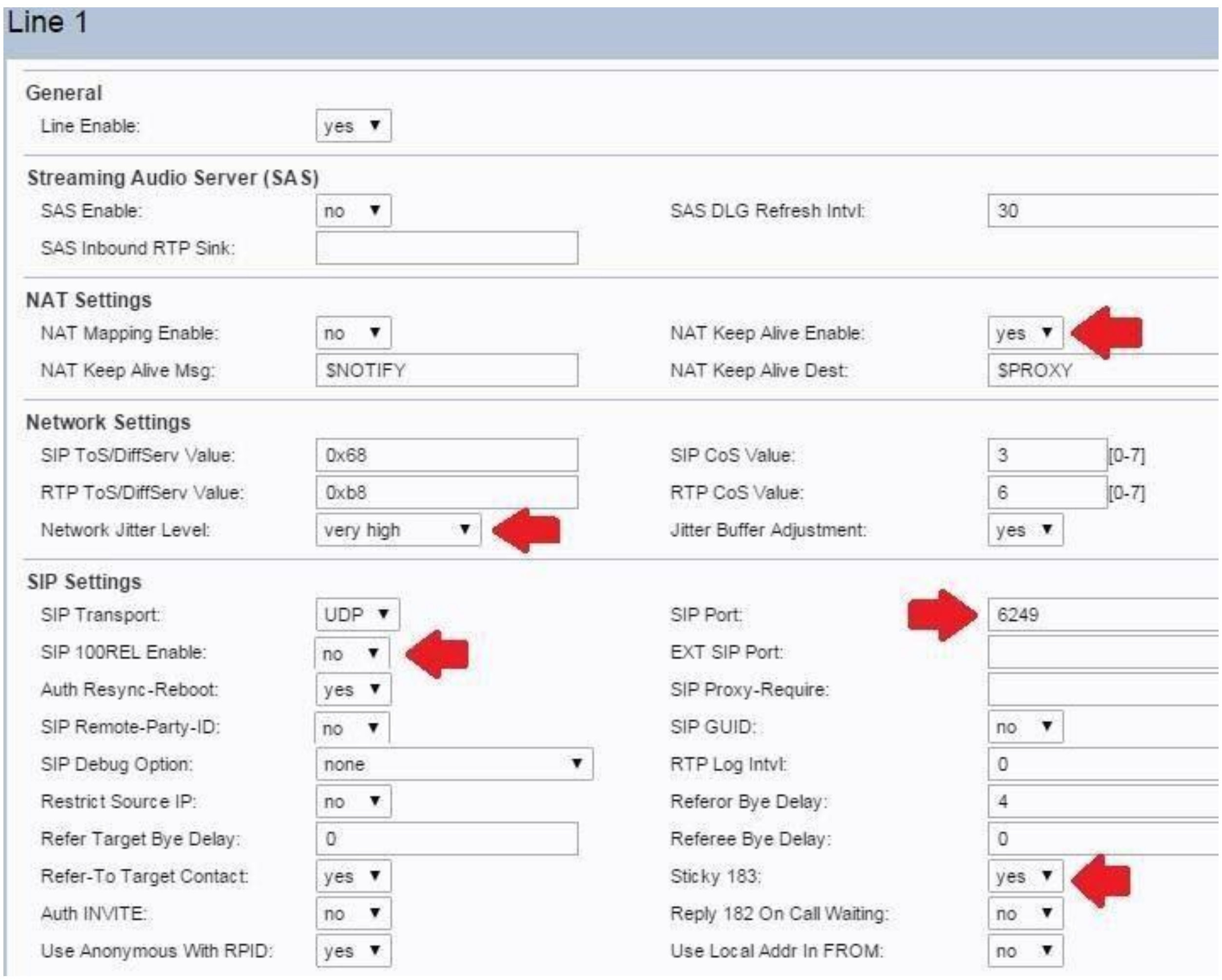
3 digit extension dial plan: **(*xxx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0|xxxxxxxxxxxx.)**

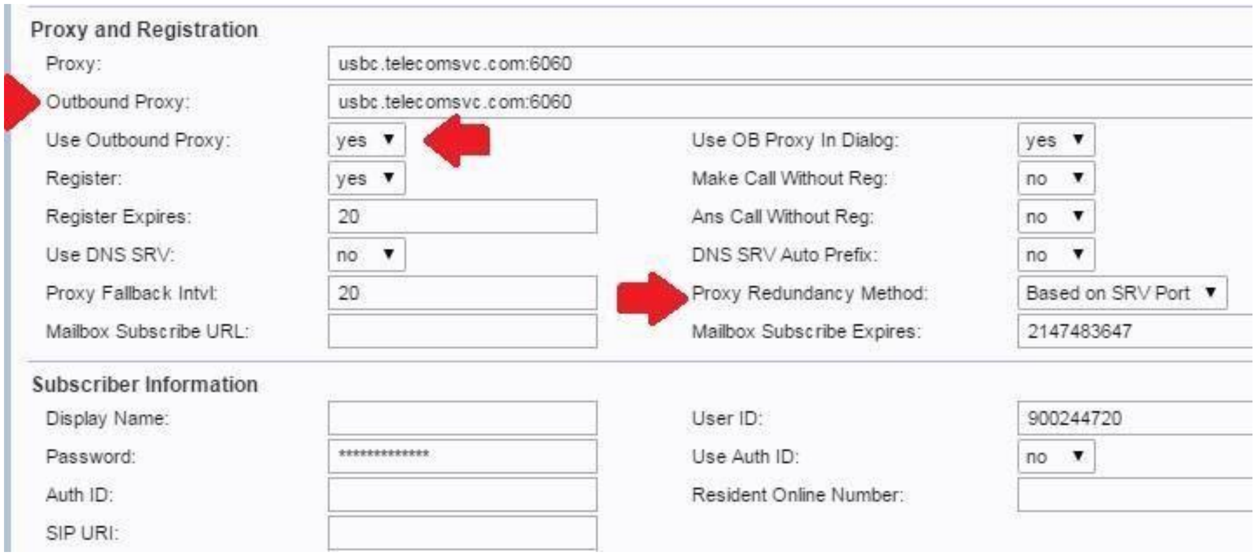
4 digit extension dial plan: **(*xxxx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0|xxxxxxxxxxxx.)**

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**.





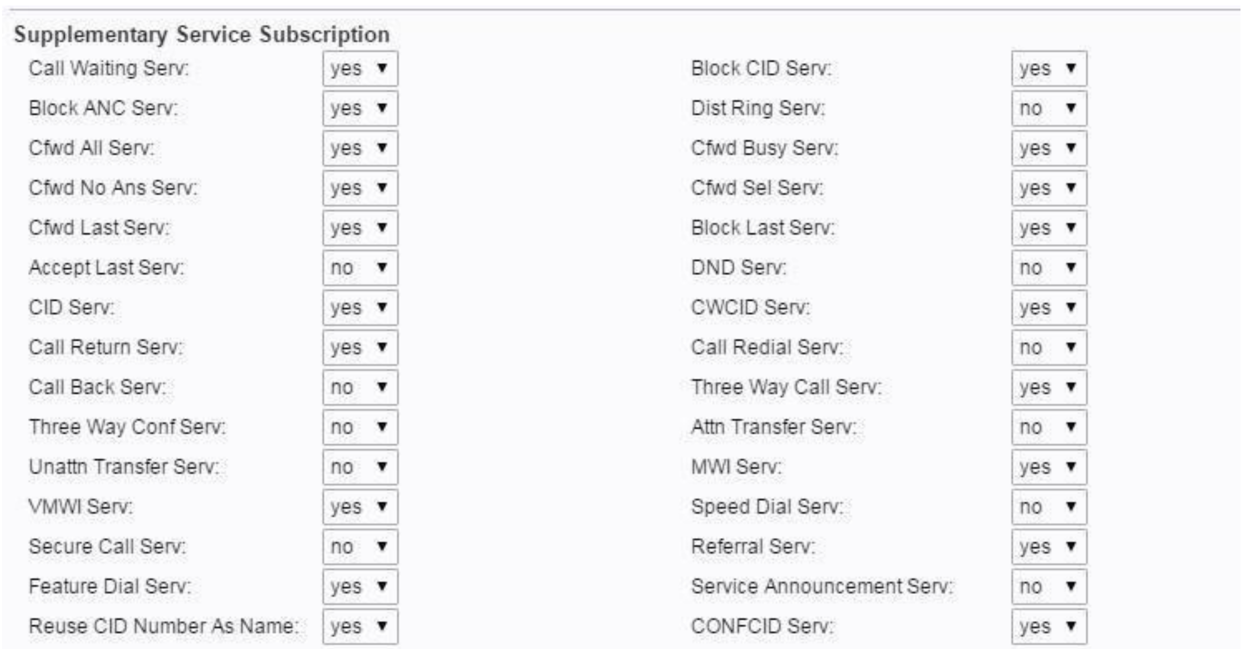
Proxy and Registration

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

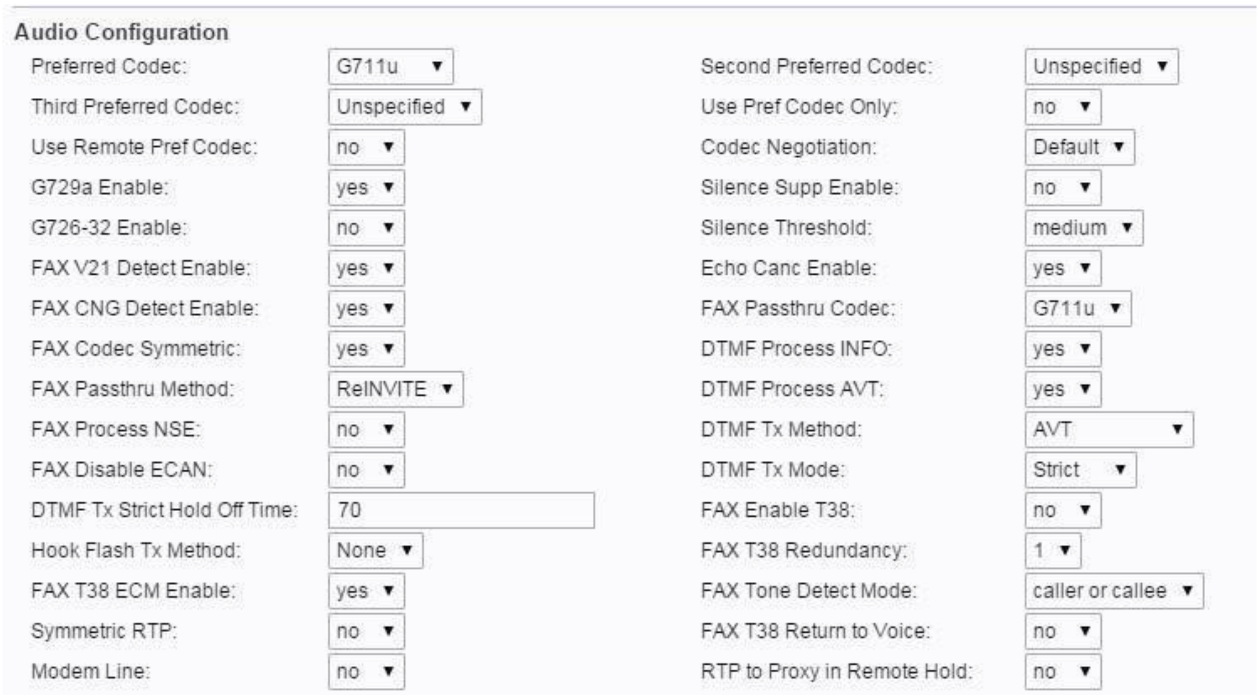
Display Name:		User ID:	900244720
Password:	**************	Use Auth ID:	no
Auth ID:		Resident Online Number:	
SIP URI:			

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



Supplementary Service Subscription

Call Waiting Serv:	yes	Block CID Serv:	yes
Block ANC Serv:	yes	Dist Ring Serv:	no
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:	no	DND Serv:	no
CID Serv:	yes	CWCID Serv:	yes
Call Return Serv:	yes	Call Redial Serv:	no
Call Back Serv:	no	Three Way Call Serv:	yes
Three Way Conf Serv:	no	Attn Transfer Serv:	no
Unattn Transfer Serv:	no	MWI Serv:	yes
VMWI Serv:	yes	Speed Dial Serv:	no
Secure Call Serv:	no	Referral Serv:	yes
Feature Dial Serv:	yes	Service Announcement Serv:	no
Reuse CID Number As Name:	yes	CONFCID Serv:	yes



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**