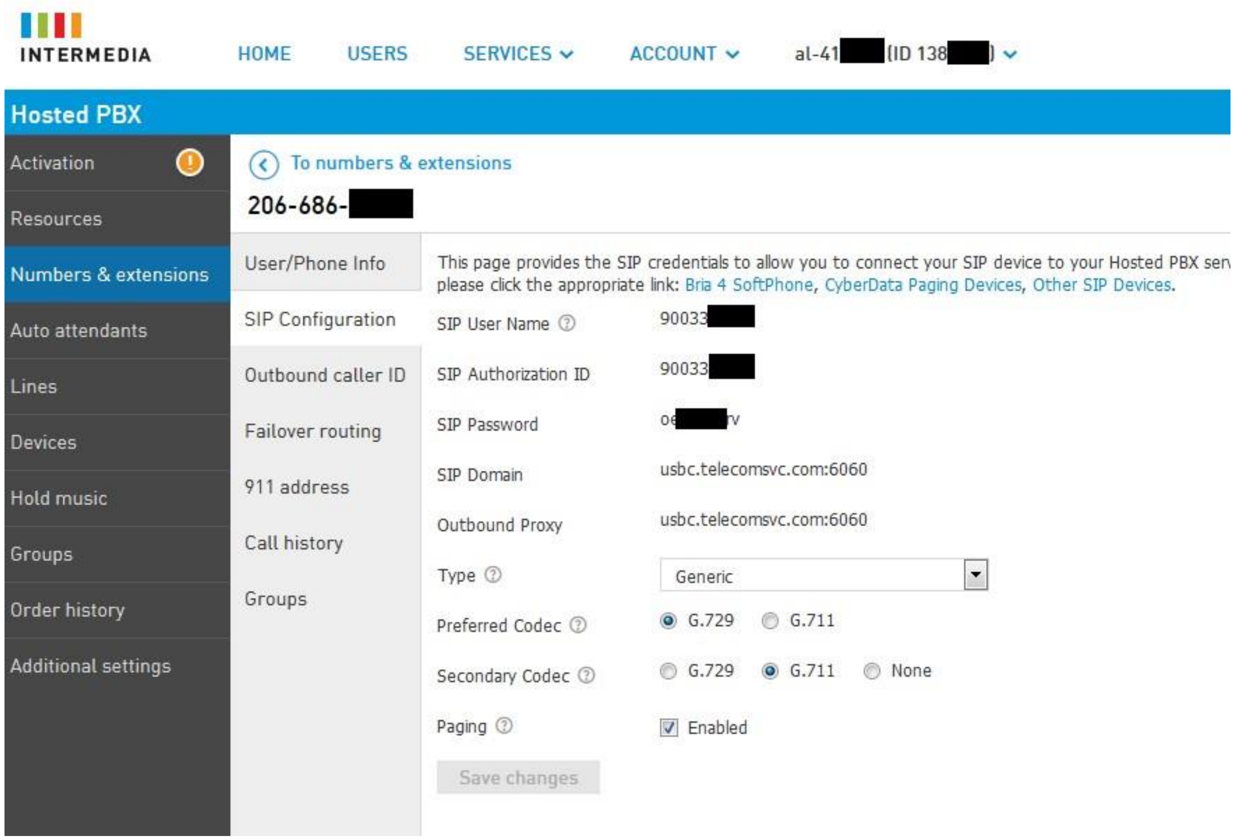
AnyPhone Configuration Guide for the Cisco SPA112
for Analog Phone Endpoints

This documentation will provide you with the steps required to configure your Cisco SPA112 as a Generic AnyPhone device for the purpose of connecting a single analog telephone to the Hosted PBX platform. Hosted PBX also utilizes the SPA112 as a fax-only endpoint. Setup of the SPA112 to serve as a faxing-only endpoint is covered in a different configuration guide.

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.

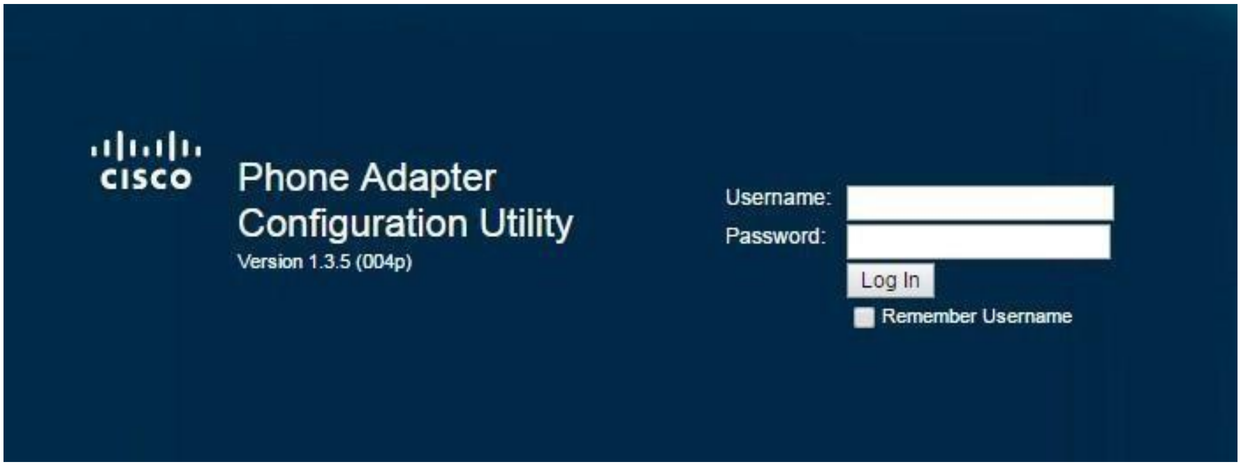
Obtaining information from HostPilot



- Login to HostPilot, locate the device in question, then locate and note the SIP Configuration info:

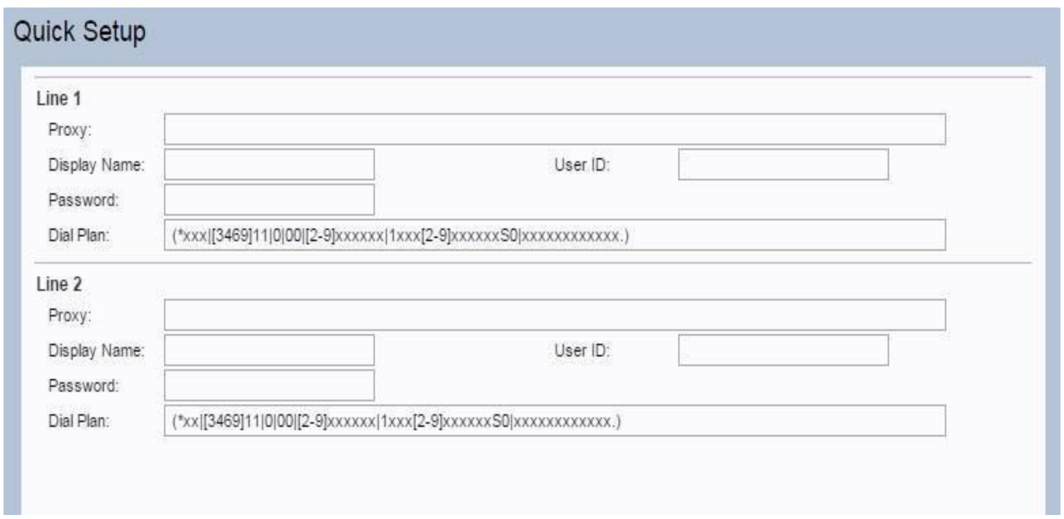
Accessing the web interface

- Connect the adapter to a DHCP enabled network, and locate its assigned IP address, and navigate to it via a web browser.
- If the IP address of the SPA112 is not known, lift the handset and dial "****", followed by "110#" after the prompt. The IP address will be voiced to the user.
- 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*

Configuring the device to register with Intermedia servers



Once logged into the SPA112, select the quick setup tab in the top left corner.

- Obtain the SIP Domain from HostPilot. Enter this into the Proxy field.
- Enter a Display Name if you wish. Display name can be whatever the name of the user will be. This will only have an effect on calls that terminate to other Intermedia HPBX customers.
- The User ID and Password fields should contain the SIP Username, and SIP password information found in HostPilot.
- Click “Submit” in the lower left corner of the SPA112 admin UI when done.
- Once the equipment is done saving the configuration, select the “Voice” tab at the top of the interface. Line 1 should now show “Registered”.

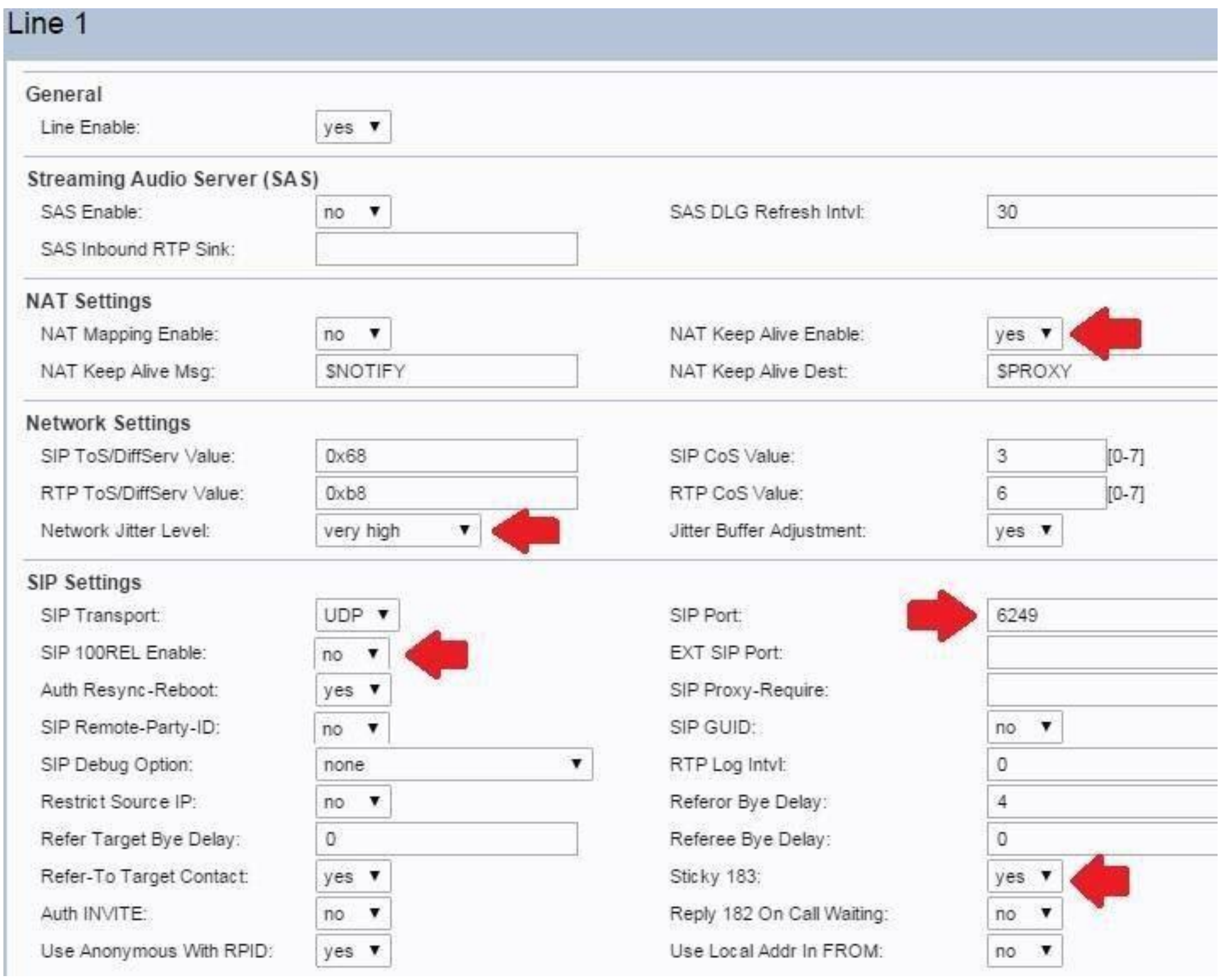
- Be sure to set the emulation from Generic to Cisco in HostPilot.
- The default dial plan does not support 3 or 4 digit extensions. Depending what extension scheme the Hosted PBX account is using, you may need to alter the dial plan.

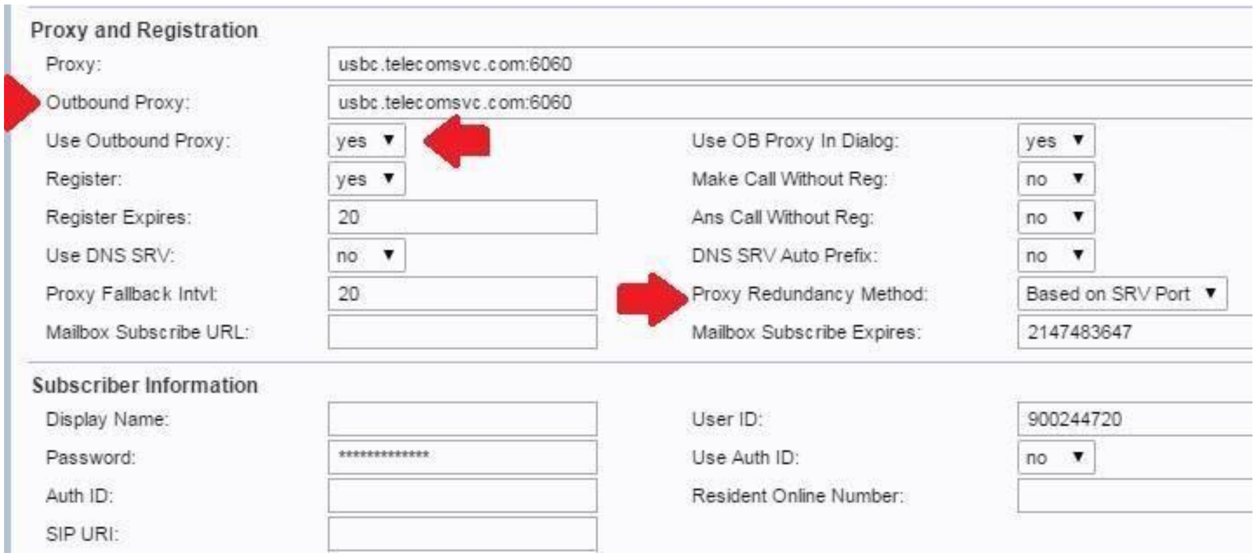
3-digit extension dial plan: **(*xxx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0|xxxxxxxxxxxx.)** 4-digit extension dial plan: **(*xxxx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0|xxxxxxxxxxxx.)**

Configuring the Voice Tab

- Next you will need to select Line 1 under the Voice tab, and make the following changes to the ATA's configuration as indicated in the image below.
- Make note of the red arrows, and confirm your adapter's configuration matches the changes marked.
- Optionally enable G729a and/or G711u. These are the only two supported codecs at this time.

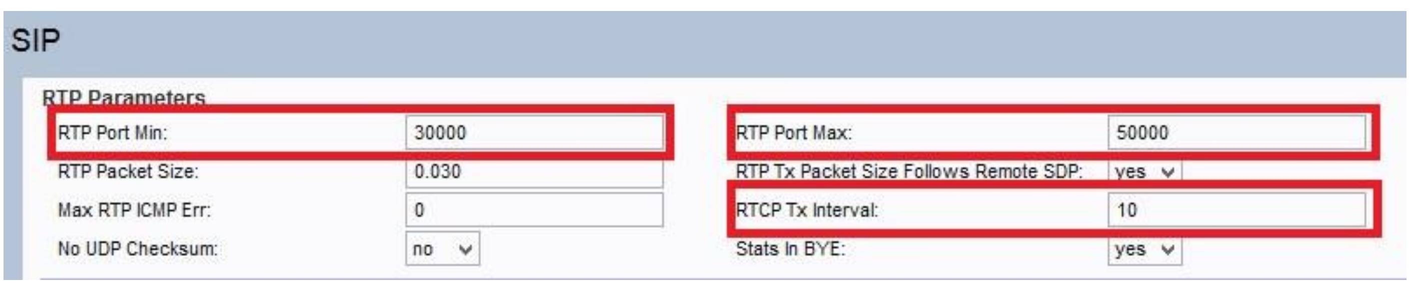
*** Note: The "SIP Port" will need to be 6xxx (xxx will be the Ext number for this phone). In the Example above you will find Ext 249 is using SIP Port: **6249**. If you do not configure this correctly, it may cause audio issues or inbound call failure issues. For sites using 4 digits extensions, use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





SIP configuration of the SPA112

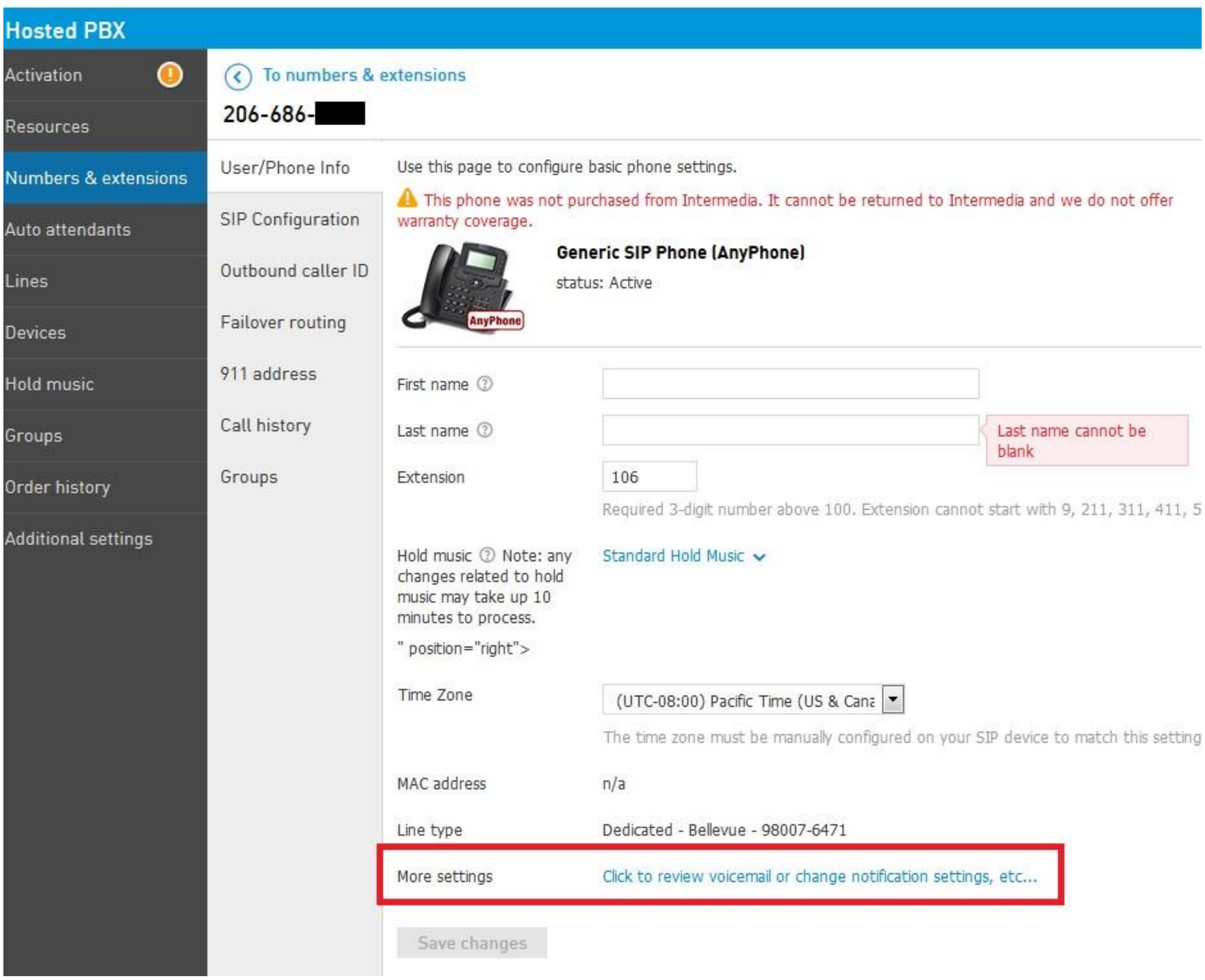
- Under SIP/Voice within the SPA112 configuration UI, ensure that the RTP parameters of your configuration match below.
- RTP port Min should be **30000**, and RTP Port Max should be **50000** Set the RTCP Tx Interval to **10**



Voice Mail

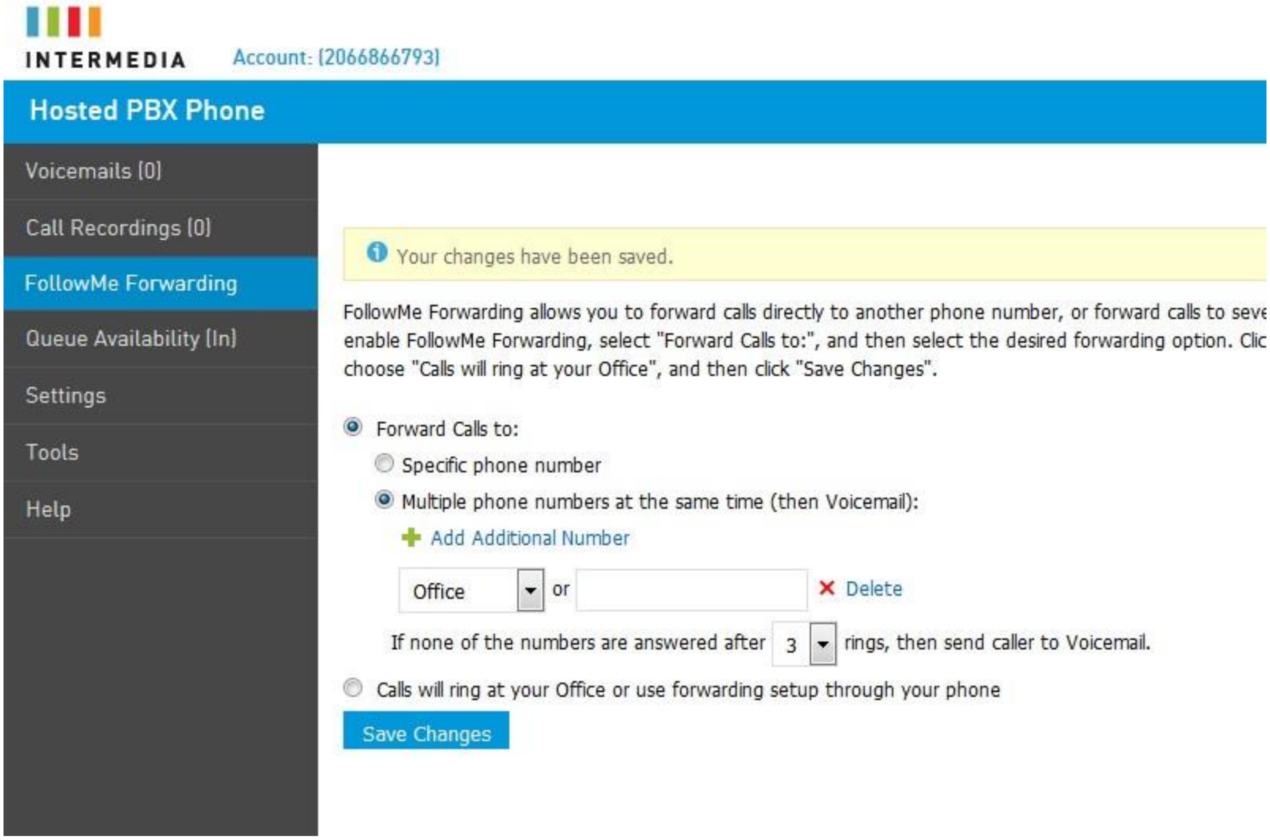
- Within the SPA112 configuration UI, under Voice/Line 1 configuration, copy the same value in the Proxy field and paste it within the Mailbox Subscribe URL field:

- Log in to TeleWeb next to route calls to Voicemail if unanswered. To login to TeleWeb: ○ Log in to HostPilot. ○ Search for the DID of your device under “Numbers & extensions”. ○ Click on the phone number of your device in the results and you will now see the “User/Phone info” tab for your phone info.
  - Click the link next to “More settings” to open TeleWeb.



- Within TeleWeb:
  - Select the “FollowMe Forwarding” tab ○ Next select the option to “Forward Calls to:”

- Next select the option for “Multiple phone numbers at the same time (then Voicemail):” ○ Set the drop down to Office and hit “Save Changes”. This will cause calls to route to voicemail if unanswered by the user.



- If there are new voicemails, the user will hear a “stutter” dial tone upon lifting the handset.
- Additionally, “ring splash” can be enabled on the SPA112 to briefly ring the handset upon a new voicemail. This setting can be found under Voice/User 1/Ring Settings/VMWI Ring Splash Len (a value of .5 would ring the phone for .5 seconds). The default value for ring splash is 0, which effectively disables it.
- To listen to voicemail, the user must dial either their extension or full 10 digit DID, then enter their PIN once prompted. It may be possible to automate this process if the user’s phone has a configurable voicemail button.

Call Transfer

To transfer a call, the user will need to perform the following:

- Have a call already established
- Press the flash key on the phone
- User should now hear dial tone
- Dial the number they wish to transfer to
- Once the call connects, the user may announce the transfer to the destination party
- Pressing flash again will join the two calls, allowing all three parties to speak as if conferenced
- The user may now hang up to complete the transfer

Star Codes

- The SPA-112 has several star codes which are accessible by the user. Several have been tested and have been documented below. For more information on star codes, see Cisco's website:

  http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/csbpvga/spa100200/admin_guide_SPA100/spa100_ag/voice.html?bookSearch=true#55636

Code	Code Name	Works?	Description
*07	Call Redial Code	Yes	Redials the last number called.
*56	CW Act Code	Yes	Enables call waiting on all calls.
*57	CW Deact Code	Yes	Disables call waiting on all calls.
*67	Block CID Act Code	Yes	Blocks caller ID on all outbound calls.
*68	Block CID Deact Code	Yes	Removes caller ID blocking on all outbound calls.
*69	Call Return Code	Yes	This code calls the last caller.
*70	CW Per Call Deact Code	No	Disables call waiting for the next call.
*71	CW Per Call Act Code	No	Enables call waiting for the next call.
*72	Cfwd All Act Code	Yes	Forwards all calls to the extension specified after the activation code.
*73	Cfwd All Deact Code	Yes	Cancels call forwarding of all calls.
*77	Block ANC Act Code	Unknown	Blocks all anonymous calls.
*78	DND Act Code	Yes	Enables the do not disturb feature. (Ring splash still notifies in this state)
*79	DND Deact Code	Yes	Disables the do not disturb feature.
*81	Block CID Per Call Act Code	Yes, but only if destination is immediately dialed	Blocks caller ID on the next outbound call.
*82	Block CID Per Call Deact Code	Does not seem to have a use	Removes caller ID blocking on the next inbound call.
*87	Block ANC Deact Code	Unknown	Removes blocking of all anonymous calls.
*98	Blind Transfer Code	No, at least not with a single trunk. Possibly with another	Begins a blind transfer of the current call to the extension specified after the activation code.