

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

The screenshot shows the HostPilot web interface. At the top, there is a navigation bar with the INTERMEDIA logo and menu items: HOME, USERS, SERVICES, and ACCOUNT. A user profile is shown as 'al-41 [redacted] (ID 138 [redacted])'. Below the navigation bar is a blue header for 'Hosted PBX'. On the left is a sidebar menu with options: Activation, Resources, Numbers & extensions (highlighted), Auto attendants, Lines, Devices, Hold music, Groups, Order history, and Additional settings. The main content area shows a breadcrumb 'To numbers & extensions' and a phone number '206-686-[redacted]'. Below this is a table of settings for 'SIP Configuration':

| Setting | Value |
|----------------------|---|
| SIP User Name | 90033-[redacted] |
| SIP Authorization ID | 90033-[redacted] |
| SIP Password | 06-[redacted] |
| SIP Domain | usbc.telecomsvc.com:6060 |
| Outbound Proxy | usbc.telecomsvc.com:6060 |
| Type | Generic |
| Preferred Codec | <input checked="" type="radio"/> G.729 <input type="radio"/> G.711 |
| Secondary Codec | <input type="radio"/> G.729 <input checked="" type="radio"/> G.711 <input type="radio"/> None |
| Paging | <input checked="" type="checkbox"/> Enabled |

A 'Save changes' button is located at the bottom of the configuration table.

□ 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”.

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

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

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**.

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 Enable: | yes ▼ |
| NAT Keep Alive Msg: | SNOTIFY |
| NAT Keep Alive Dest: | SPROXY |
| Network Settings | |
| SIP ToS/DiffServ Value: | 0x68 |
| RTP ToS/DiffServ Value: | 0xb8 |
| Network Jitter Level: | very 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: | no ▼ |
| SIP Debug Option: | none ▼ |
| Restrict Source IP: | no ▼ |
| Refer Target Bye Delay: | 0 |
| Refer-To Target Contact: | yes ▼ |
| Auth INVITE: | no ▼ |
| Use Anonymous With RPID: | yes ▼ |
| SIP Port: | 6249 |
| EXT SIP Port: | |
| SIP Proxy-Require: | |
| SIP GUID: | no ▼ |
| RTP Log Intvl: | 0 |
| Referor Bye Delay: | 4 |
| Referee Bye Delay: | 0 |
| Sticky 183: | yes ▼ |
| Reply 182 On Call Waiting: | no ▼ |
| Use Local Addr In FROM: | no ▼ |

| 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: | | | |

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**

| SIP | | | |
|-------------------|-------|--|-------|
| RTP Parameters | | | |
| 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 ▼ |

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:

| Proxy and Registration | |
|-------------------------------|---------------------------------|
| Proxy: | usbc.accessline.com:6060 |
| Outbound Proxy: | usbc.accessline.com:6060 |
| Use Outbound Proxy: | no ▾ |
| Register: | yes ▾ |
| Register Expires: | 3600 |
| Use DNS SRV: | no ▾ |
| Proxy Fallback Intvl: | 3600 |
| Mailbox Subscribe URL: | usbc.accessline.com:6060 |

- 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.

Hosted PBX

Activation ⓘ
 Resources
Numbers & extensions
 Auto attendants
 Lines
 Devices
 Hold music
 Groups
 Order history
 Additional settings

← To numbers & extensions

206-686-████

User/Phone Info Use this page to configure basic phone settings.

⚠ This phone was not purchased from Intermedia. It cannot be returned to Intermedia and we do not offer warranty coverage.

Generic SIP Phone (AnyPhone)

status: Active

First name ⓘ

Last name ⓘ Last name cannot be blank

Extension
Required 3-digit number above 100. Extension cannot start with 9, 211, 311, 411, 5

Hold music ⓘ Note: any changes related to hold music may take up 10 minutes to process.
" position="right"> Standard Hold Music ▾

Time Zone
The time zone must be manually configured on your SIP device to match this setting

MAC address n/a

Line type Dedicated - Bellevue - 98007-6471

More settings
Click to review voicemail or change notification settings, etc...

- 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.

The screenshot shows the Intermedia Hosted PBX Phone settings interface. The top navigation bar includes the Intermedia logo and the account number [2066866793]. The left sidebar contains menu items: Voicemails (0), Call Recordings (0), FollowMe Forwarding (highlighted), Queue Availability (In), Settings, Tools, and Help. The main content area displays a yellow notification: "Your changes have been saved." Below this, a text block explains: "FollowMe Forwarding allows you to forward calls directly to another phone number, or forward calls to several numbers at the same time. To enable FollowMe Forwarding, select 'Forward Calls to:', and then select the desired forwarding option. Click choose 'Calls will ring at your Office', and then click 'Save Changes'." The settings are configured as follows:

- Forward Calls to:
 - Specific phone number
 - Multiple phone numbers at the same time (then Voicemail):
 - + Add Additional Number
 - Office (dropdown) or [] (input) [X] Delete
- If none of the numbers are answered after 3 (dropdown) rings, then send caller to Voicemail.
- Calls will ring at your Office or use forwarding setup through your phone

 A blue "Save Changes" button is located at the bottom of the settings area.

- 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/csbpvgg/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. |

