



# Advanced Settings

## User Guide

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### Overview

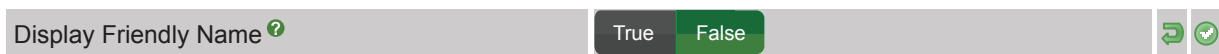
In general, a Schmooze Support Representative should be the only one to modify the Advanced Settings module. When changes are made, they are usually done at the initial installation. For example, if your SIP endpoints are usually external and require Follow Me, then you could set NAT to “Yes” to create Follow Me when creating new extensions. When used correctly this module can save time.

### How To Change A Setting

- The Advanced Settings module is an advanced module. For each setting there can be four states.
  - **Default**- At the far right of each setting lays the indicator of change shown here with the blue dots. The example shows no green U-turn button and is therefore the default setting.



- **Changed But Not Saved**- In this state, a change has been made from defaults, but has not been saved. You will see a green U-turn and checkmark button. Clicking on the green checkmark icon saves it.



- **Changed and Saved**- In this state, you have changed a setting from the default and have saved it. Anywhere you see the U-turn button, the setting is not default. Most changes require you to press the “Apply Config” button.



- **Set Back to Default Setting But Not Saved**- Only the green checkmark button is displayed. Press this button to save the change back to default. Most changes require you to press the “Apply Config” button.



Apply Config

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- Note that while you may use the “Refresh Page” button at the bottom of the page, it is often best to just do a browser refresh.

Refresh Page

### Notice

Use extreme caution when making changes! Some of these settings can render your system inoperable. You are urged to backup before making any changes. Readonly settings are usually more volatile; changing “Override Readonly Settings” to “True” can alter them.

## Logging In

- Log into the Advanced Settings module and you should see a screen like this.

### Advanced Settings Details

Display Friendly Name <sup>?</sup>	<input checked="" type="checkbox"/> True	<input type="checkbox"/> False
Display Hidden Settings <sup>?</sup>	<input checked="" type="checkbox"/> True	<input type="checkbox"/> False
Display Readonly Settings <sup>?</sup>	<input checked="" type="checkbox"/> True	<input type="checkbox"/> False
Override Readonly Settings <sup>?</sup>	<input checked="" type="checkbox"/> True	<input type="checkbox"/> False

## Advanced Settings Details

- These settings control what this module displays and whether or not certain items are editable. Please see Appendix A for an explanation of hidden and readonly items.
  - **Display Friendly Name**- Normally the friendly names will be displayed on this page and the internal freepbx\_conf configuration names are shown in the tooltip. If you prefer to display the configuration variables and show the friendly names in the tooltip, set this to “False.” Default value is **True** and the internal name is AS\_DISPLAY\_FRIENDLY\_NAME.
  - **Display Hidden Settings**- This will display settings normally hidden by the system. These settings are often internally used and are of little interest to most users. Default value is **False** and the internal name is AS\_DISPLAY\_HIDDEN\_SETTINGS.
  - **Display Readonly Settings**- This will display settings that are readonly. These settings are often internally used and are of little interest to most users. Default value is **False** and the internal name is AS\_DISPLAY\_READONLY\_SETTINGS.
  - **Override Readonly Settings**- Setting this to “True” will allow you to override un-hidden readonly settings to change them. Settings that are readonly may be extremely volatile and will most likely break your system if you change them. Take extreme caution if you elect to make any changes to any readonly setting. Default value is **False** and the internal name is AS\_OVERRIDE\_READONLY.

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### Advanced Settings Details

Display Friendly Name ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False
Display Hidden Settings ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False
Display Readonly Settings ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False
Override Readonly Settings ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False

## Asterisk Manager

- These are the credentials for accessing the “Asterisk Manager Interface” or AMI.
  - **Asterisk Manager Password**- Password for AMI. This will be updated automatically in manager.conf. Default value is **amp111** and the internal name is AMPMGRPASS.
  - **Asterisk Manager User**- Password for AMI. This will be updated automatically in manager.conf. Default value is **admin** and the internal name is AMPMGRUSER.

### Asterisk Manager

Asterisk Manager Password ?	<input type="text" value="amp111"/>
Asterisk Manager User ?	<input type="text" value="admin"/>

## Backup Module

- This section allows you to specify an ‘Email “From:” Address’ and enable advanced options.
  - **Email "From:" Address**- The field for emails when using the backup to email feature. **No Default** provided, field can be left blank and the Internal name is AMPBACKUPEMAILFROM.
  - **Enable Advanced Options**- In it's current form, this setting exposes “sudo” on a give backup set when saving/restoring. This has serious security risks and you are strongly discouraged unless you are very comfortable with the implications. This could be necessary if backing up certain system files on your PBX that apache normally would not have access to. It also requires that your “sudo” permissions be setup for the apache user and be restricted to the minimum commands required. Default value is **False** and the internal name is AMPBACKUPADVANCED.

### Backup Module

Email “From:” Address ?	<input type="text"/>
Enable Advanced Options ?	<input type="checkbox"/> True <input checked="" type="checkbox"/> False

### Call Flow Control Module

- By default, the Call Flow Control module will not hook time conditions, allowing one to associate a call flow toggle feature code with a time condition since time conditions have their own feature code as of version 2.9. If there is already an association configured on an upgraded system, then this setting will have no affect for the time conditions that are affected. Setting this to “True” reverts to the 2.8 and prior behavior by allowing for the use of a call flow toggle to be associated with a time condition. This can be useful for two scenarios. First, to override a time condition without the automatic resetting that occurs with the built-in time condition overrides. The second use is the ability to associate a single call flow toggle with multiple time conditions, thus creating a “master switch” that can be used to override several possible call flows through different time conditions.
- **Hook Time Conditions Module**- Enable or disable. Default value is **False** and the internal name is DAYNIGHTTHOOK.

Call Flow Control Module

Hook Time Conditions Module ?

True

False

### Camp-On Module

- Camp-On is the function of retrying a busy extension, at some interval, until that extension answers. In general, the default settings are fine for most users. However, if you are using phones with channel specific capabilities, then be aware of the **Caller Policy** and **Callee Policy** settings in the Extensions module and how it relates to the Camp-On module section in advanced settings. See the Extensions module user guide for info on “Call Camp-On” configuration. For reference, the “Call Camp-On Services” section of the Extension module is shown below.

- Call Camp-On Services

Forcing default settings: ?

Caller Policy: ?

Callee Policy: ?

Disable Camp-On

✓ Generic Device

Native Technology Support

- By default, the Call Camp-On is set to “Generic Device” in newly created extensions. Here, you may “Disable Camp-On” for new extension creation and make changes to the operation of the Camp-On feature.
- **Maximum Active Camp-On Requests**- System wide maximum number of outstanding

Camp-On requests that can be active. This limit is useful on a system that might have memory constraints since the internal state machine takes up system resources relative to the number of active requests it has to track. Restart Asterisk for changes to take effect. Default value is **20** with acceptable values of **1-1000**. Internal name is `CC_MAX_REQUESTS_GLOBAL`.

- **Non Extensions Callee Policy**- If this is set to “generic” or “always,” then it will be possible to attempt call completion requests when dialing non-extensions, such as ring groups or other possible destinations that could work with call completion. Setting this will bypass and callee policies can result in inconsistent behavior. If set, “generic” is the most common and “always” will attempt to use technology specific capabilities if the called extension uses a channel that supports said tech. Default value is **Never** and the internal name is `CC_NON_EXTENSION_POLICY`.
- **Only Use Default Camp-On Settings**- You can force all extensions on a system to only use the default Camp-On settings. When in this mode, the individual settings will not be shown on the extension page. If there were unique settings previously configured, the data will be retained and used until you switch this back to “False.” With this set, the **Caller Policy** (`cc_agent_policy`) and **Callee Policy** (`cc_monitor_policy`) settings will still be configurable for each user so you can still enable or disable the call camping ability on select extensions. Default is **True** and the internal name is `CC_FORCE_DEFAULTS`.
- **Caller Policy Default**- Used to enable Camp-On for a user and set the technology mode that will be used when engaging the feature. In most cases “generic” should be chosen unless you have phones designed to work with channel specific capabilities. Default value is **generic** and the internal name is `CC_AGENT_POLICY_DEFAULT`.
- **Callee Policy Default**- Used to control if other phones are allowed to Camp-On to an extension. If so, it sets the technology mode used to monitor the availability of the extension. If no specific technology support is available, then it should be set to “generic.” In this mode, a callback will be initiated to the extension when it changes from “InUse” state to “NotInUse.” If it was busy when first attempted, this will be when the current call has ended. If it simply did not answer, then this will be the next time this phone is used to make or answer a call and then hangs up. It is possible to set this to take advantage of “native” technology support, if available, and automatically fallback to “generic” by setting this to “always.” Default value is **generic** and the internal name is `CC_MONITOR_POLICY_DEFAULT`.
- **Caller Timeout to Request Default**- This is how many seconds after dialing an extension that a user has to make a call completion request. Default is **30** with an internal name of `CC_OFFER_TIMER_DEFAULT`.
- **Max Camp-On Life Busy Default**- This is how long a call completion request will remain active, in seconds, before expiring if the phone rang busy when first attempting the call. Default is **4800** and the internal name is `CCBS_AVAILABLE_TIMER_DEFAULT`.
- **Max Camp-On Life No Answer Default**- This is how long a call completion request will remain active, in seconds, before expiring if the phone was simply unanswered when first attempting the call. Default is **7200** and the internal name is `CCNR_AVAILABLE_TIMER_DEFAULT`.
- **Default Time to Ring Back Caller**- This is how long, in seconds, to ring back to a caller who has a **Caller Policy** set to “Generic Device.” This has no affect if set to any other setting. Default is **15** and the internal name is `CC_RECALL_TIMER_DEFAULT`.
- **Default Caller Callback Mode**- If not set, a callback request will be dialed straight to the

specific device that made the call. If using “native” technology support, this may be the preferred mode. The “internal” (Callback Standard) option will initiate a call back to the caller just as if someone else on the system placed the call, which means the call can pursue Follow-Me. To avoid Follow-Me settings, choose “extension” (Callback Extension). Default value is **extension** and the internal name is `CC_AGENT_DIALSTRING_DEFAULT`.

- **Default Max Camped-On Extensions**- Only valid when using “native” technology support for **Caller Policy**. This is the number of outstanding call completion requests that can be pending to different extensions. With “generic” device mode, you can only have a single request outstanding and this will be ignored. Default value is **5** and the internal name is `CC_MAX_AGENTS_DEFAULT`.
- **Default Callback Alert-Info**- This is an optional alert-info setting that can be used when initiating a callback. Only valid when **Caller Policy** is set to “Generic Device.” **No Default** provided, the field can be left blank. Internal name is `CC_AGENT_ALERT_INFO_DEFAULT`.
- **Default Callback CID Prepend**- This is an optional CID Prepend setting that can be used when initiating a callback. Only valid when **Caller Policy** is set to “Generic Device.” **No Default** provided, the field can be left blank. Internal name is `CC_AGENT_CID_PREPEND_DEFAULT`.
- **Announce the Callee Extension**- When set to “True,” the target extension being called will be announced upon answering the callback and prior to ringing the extension. Setting this to “False” will go directly to the ringing extension. The CID information will still reflect who is being called back. Default value is **True** and the internal name is `CC_ANNOUNCE_MONITOR_DEFAULT`.
- **Default Callee Alert-Info**- An optional alert-info setting that can be used to send to the extension being called back. **No Default** provided, the field can be left blank. Internal name is `CC_MONITOR_ALERT_INFO_DEFAULT`.
- **Default Callee CID Prepend**- This is an optional CID Prepend setting that can be used to send to the extension being called back. **No Default** provided, the field can be left blank. Internal name is `CC_MONITOR_CID_PREPEND_DEFAULT`.
- **Default Max Queued Callers**- This is the maximum number of callers that are allowed to queue up call completion requests against this extension. Default value is **5** and the internal name is `CC_MAX_MONITORS_DEFAULT`.
- **BLF Camp-On Available State**- This is the state that will be set for BLF subscriptions after attempting a call. It is still possible to Camp-On to the last called number, prior to the offer\_timer expiring. Restart Asterisk for this change to take effect. Default value is **NOT\_INUSE** and the internal name is `CC_BLF_OFFERED`.
- **BLF Camp-On Pending State**- This is the state that will be set for BLF subscriptions upon a successful Camp-On request, pending a callback when the party becomes available. Restart Asterisk for this change to take effect. Default value is **INUSE** and the internal name is `CC_BLF_PENDING`.
- **BLF Camp-On Busy Caller State**- This is the state that will be set for BLF subscriptions once the callee becomes available if the caller is not busy. Restart Asterisk for this change to take effect. Default value is **ONHOLD** and the internal name is `CC_BLF_CALLER_BUSY`.
- **BLF Camp-On Recalling State**- This is the state that will be set for BLF subscriptions once the callee becomes available if the caller is not busy. Restart Asterisk for this change to take effect. Default value is **RINGING** and the internal name is `CC_BLF_RECALL`.

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### Camp-On Module

Maximum Active Camp-On Requests ?	<input type="text" value="20"/>
Non Extensions Callee Policy ?	<input type="text" value="never"/>
Only Use Default Camp-On Settings ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False
Caller Policy Default ?	<input type="text" value="generic"/>
Callee Policy Default ?	<input type="text" value="generic"/>
Caller Timeout to Request Default ?	<input type="text" value="30"/>
Max Camp-On Life Busy Default ?	<input type="text" value="4800"/>
Max Camp-On Life No Answer Default ?	<input type="text" value="7200"/>
Default Time to Ring Back Caller ?	<input type="text" value="15"/>
Default Caller Callback Mode ?	<input type="text" value="extension"/>
Default Max Camped-On Extensions ?	<input type="text" value="5"/>
Default Callback Alert-Info ?	<input type="text"/>
Default Callback CID Prepend ?	<input type="text"/>
Announce the Callee Extension ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False
Default Callee Alert-Info ?	<input type="text"/>
Default Callee CID Prepend ?	<input type="text"/>
Default Max Queued Callers ?	<input type="text" value="5"/>
BLF Camp-On Available State ?	<input type="text" value="NOT_INUSE"/>
BLF Camp-On Pending State ?	<input type="text" value="INUSE"/>
BLF Camp-On Busy Caller State ?	<input type="text" value="ONHOLD"/>
BLF Camp-On Recalling State ?	<input type="text" value="RINGING"/>

## Developer and Customization

- The Developer and Customization settings are for use in the development process only and are of little interest to end users.
  - **Always Download Web Assets**- The PBX appends versioning tags on the CSS and javascript files and some of the main logo images. The versioning will help force browsers to load new versions of the files when module versions are upgraded. Setting this value to "True" will try to force these to be loaded to the browser for every page load by appending an additional timestamp in the version information. This is useful during development and debugging when changes are being made to javascript and CSS files. Default value is **False** and the internal name is FORCE\_JS\_CSS\_IMG\_DOWNLOAD.
  - **Debug File**- Full path and name of the PBX debug file. Used by the dbug() function by developers. Default value is `/var/log/asterisk/freepbx_dbug` and the internal name is FPBXDEBUGFILE.
  - **Developer Mode**- This enables several debug features geared towards developers, including

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some page load timing information, some debug information in the Module Admin and use of the original CSS files. Other future capabilities will also be enabled. Default value is **False** and the internal name is DEVEL.

- **Disable PBX debug Logging**- Set to “True” to stop all debug() calls from writing to the debug file (FPBXDEBUGFILE). Default value is **True** and the internal name is FPBXDEBUGDISABLE.
- **Disable Mainstyle CSS Compression**- Stops the automatic generation of a stripped CSS file that replaces the primary sheet, usually mainstyle.css. Default value is **False** and the internal name is DISABLE\_CSS\_AUTOGEN.
- **Leave Reload Bar Up**- Forces the “Apply Configuration Changes” reload bar to always be present even when not necessary. Default value is **False** and the internal name is DEVELRELOAD.
- **POST\_RELOAD Debug Mode**- Display debug output for script used if POST\_RELOAD is used. Default value is **False** and the internal name is POST\_RELOAD\_DEBUG.
- **Provide Verbose Tracebacks**- Provides a very verbose traceback when die\_freepbx() is called, including extensive object details if present in the traceback. Default value is **False** and the internal name is DIE\_FREEPBX\_VERBOSE.
- **Use Packaged Javascript Library**- The PBX packages several javascript and components into a compressed file called libfreepbx.javascript.js. By default, this will be loaded instead of the individual uncompressed files. This is useful during development and debugging. Default value is **True** and the internal name is USE\_PACKAGED\_JS.

### Developer and Customization

Always Download Web Assets ?	True	False
Debug File ?	<input type="text" value="/var/log/asterisk/freepbx_debug"/>	
Developer Mode ?	True	False
Disable PBX debug Logging ?	True	False
Disable Mainstyle CSS Compression ?	True	False
Leave Reload Bar Up ?	True	False
POST_RELOAD Debug Mode ?	True	False
Provide Verbose Tracebacks ?	True	False
Use Packaged Javascript Library ?	True	False

## Device Settings

- This section of Advanced Settings is very useful on fresh systems when most of the extensions are going to be “not default” and you want to speed up the build process and/or expose additional settings at extension creation time. For instance, setting SIP NAT to “Yes” as all your devices are remote or implicitly specifying voice codecs.
- **Show all Device Setting on Add**- When adding a new extension/device, setting this to “True”

will show most available device settings, as seen when editing a previously created extension/device, vs. the default simplified extension/device creation menu. Default value is **False** and the internal name is `ALWAYS_SHOW_DEVICE_DETAILS`.

- **Require Strong Secrets**- Requires a strong secret on SIP and IAX devices requiring at least two numeric and non-numeric characters and no fewer than 6 total characters. This can be disabled if using devices that cannot meet these needs, or you prefer to put more rigid constraints including that this rule actually considers weak when they may not be. Default value is **True** and the internal name is `DEVICE_STRONG_SECRETS`.
- **Remove mailbox Setting when no Voicemail**- If set to “True,” any fixed device associated with a user that has no voicemail configured will have the “mailbox=” setting removed in the generated technology configuration file, such as the `sip_additional.conf`. This will not affect the value in the GUI. Default value is **False** and the internal name is `DEVICE_REMOVE_MAILBOX`.
- **SIP canrenvite (directmedia)**- Default setting for SIP canreinvite (same as directmedia). See Asterisk documentation for details. Default value is **no** and the internal name is `DEVICE_SIP_CANREINVITE`.
- **SIP trustpid**- Default setting for SIP trustpid. See Asterisk documentation for details. Default value is **yes** and the internal name is `DEVICE_SIPTRUSTPID`.
- **SIP sendrpid**- Default setting for SIP sendrpid. A value of “yes” is equivalent to “rpid” and will send the “Remote-Party-ID” header. A value of “pai” is only valid starting with Asterisk 1.8 and will send the “P-Asserted-Identity” header. See Asterisk documentation for details. Default value is **no** and the internal name is `DEVICE_SIP_SENDRPID`.
- **SIP nat**- Default setting for SIP NAT. A “no” follows the protocol. A “yes” will attempt to handle NAT. This also works for the LAN or local (uses the network address:port instead of the reported ports). A “never” tries to block it, no RFC3581. A “route” ignores the rport information. See Asterisk documentation for details. Default value is **no** and the internal name is `DEVICE_SIP_NAT`.
- **SIP encryption**- Default setting for SIP encryption. Whether to offer SRTP encrypted media (and only SRTP encrypted media) on outgoing calls to a peer. Calls will fail with `HANGUPCAUSE=58` if the peer does not support SRTP. See Asterisk documentation for details. Default value is **no** and the internal name is `DEVICE_SIP_ENCRYPTION`.
- **SIP qualifyfreq**- Default setting for SIP qualifyfreq. This specifies the frequency of which “qualify” OPTIONS messages will be sent to the device. This can help keep NAT holes open, but not dependable for remote client firewalls. Only valid for Asterisk 1.6 and above. See Asterisk documentation for details. Default value is **60** with an acceptable range of **15-86400**. The internal name is `DEVICE_SIP_QUALIFYFREQ`.
- **SIP and IAX qualify**- Default setting for SIP and IAX qualify. Whether to send periodic OPTIONS messages (for SIP) or otherwise monitor the channel and at what point to consider the channel unavailable. A value of “yes” is equivalent to 2000 msec. This can help keep NAT holes open with SIP, but not dependable for remote client firewalls. See Asterisk documentation for details. Default value is **yes** and the internal name is `DEVICE_QUALIFY`.
- **SIP and IAX allow**- Default settings for SIP and IAX allow (allow codecs). Codecs to allow in addition to those set in general settings unless explicitly “disallowed” for the device. Values can be separated with “&,” for example “ulaw&g722&g729,” where the preference order is preserved. See Asterisk documentation for details. **No** default provided and this field can be left blank. The internal name is `DEVICE_ALLOW`.

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- **SIP and IAX disallow**- Default settings for SIP and IAX disallow (disallow codecs). Specify which codecs to disallow. It can help to reset codecs from the general settings by setting a value of “all” and then specifically including allowed codecs with the “allow” directive. Values can be separated with “&,” for example “g722&g729.” See Asterisk documentation for details. **No** default provided and this field can be left blank. The internal name is DEVICE\_DISALLOW.
- **SIP and DAHDi callgroup**- Default setting for SIP, DAHDi and ZAP callgroup. Callgroup(s) that the device is part of. This can be one or more groups, for example, “1,3-5” would place the device in the “1,3,4,5” callgroups. See Asterisk documentation for details. **No** default provided and this field can be left blank. The internal name is DEVICE\_CALLGROUP.
- **SIP and DAHDi pickupgroup**- Default setting for SIP, DAHDi and ZAP pickupgroup. This is the pickupgroup that a device can pickup calls from. This can be one or more groups, for example, “1,3-5” would place the device in the “1,3,4,5” pickupgroups. Device does not have to be in a group to be able to pickup calls from that group. See Asterisk documentation for details. **No** default provided and this field can be left blank. The internal name is DEVICE\_PICKUPGROUP.

### Device Settings

Show all Device Setting on Add?	<input type="checkbox"/> True <input checked="" type="checkbox"/> False
Require Strong Secrets?	<input type="checkbox"/> True <input checked="" type="checkbox"/> False
Remove mailbox Setting when no Voicemail?	<input type="checkbox"/> True <input checked="" type="checkbox"/> False
SIP canrenivate (directmedia)?	<input type="text" value="no"/>
SIP trustpid?	<input type="text" value="yes"/>
SIP sendrpip?	<input type="text" value="no"/>
SIP nat?	<input type="text" value="no"/>
SIP encryption?	<input type="text" value="no"/>
SIP qualifyfreq?	<input type="text" value="60"/>
SIP and IAX qualify?	<input type="text" value="yes"/>
SIP and IAX allow?	<input type="text"/>
SIP and IAX disallow?	<input type="text"/>
SIP and DAHDi callgroup?	<input type="text"/>
SIP and DAHDi pickupgroup?	<input type="text"/>

## Dialplan and Operational

- The Dialplan and Operational section can, like Device Settings, be useful when initially building a system and you have certain behaviors in mind for the overall application.
- **Block CNAM on External Trunks**- Some carriers will reject a call if a caller ID (CNAM) is presented. This occurs in several areas when configuring CID on the PBX using the format of “CNAM.” To remove the CNAM part of the CID on all external trunks, set this value to “True.”

This WILL NOT remove CNAM when a trunk is called from an intra-company route. This can be done on individual trunks in addition to globally if there are trunks where it is desirable to retain CNAM information. Most carriers seem to ignore CNAM. Default value is **False** and the internal name is BLOCK\_OUTBOUND\_TRUNK\_CID.

- **Call Forward Ringtimer Default**- This is the default time, in seconds, to try and connect a call that has been forwarded by the server side CF, CFU or CFB options (if your phone uses client side CF, such as SIP redirects, this will not have any affect). If set to the default of “0,” it will use the standard ring timer. If set to “-1,” it will ring the forwarded number with no limit, which is consistent with the behavior of existing PBX systems. If set to any other value, it will ring that value before diverting to the users voicemail, if one exists. This can be overwritten for each extension. Default value is **0** and the internal name is CFRINGTIMERDEFAULT.
- **CW Enabled by Default**- Enable or disable call waiting, by default, on extension creation. Set to “no” if you don’t want phones to be commissioned with call waiting already enabled. The user would then be required to dial the CW feature code (\*70 by default) to enable CW. Most installations should be left at default settings as this allows multi-line phones to receive multiple calls on their line appearances. Default value is **True** and the internal name is ENABLECW.
- **Disable -custom Context Includes**- Normally the PBX auto-generates a custom context that may be usable for adding custom dialplan to modify the normal behavior of FreePBX. It takes a good understanding of how the Asterisk processes these “Includes” to use this and in many of the cases, there is no useful application. All “Includes” will result in a WARNING in the Asterisk log if there is no context found to include, though it results in no errors. If you know that you will want the “Includes,” you can set this to “True.” If you comment it out, the PBX will revert to legacy behavior and include the contexts. Default value is **False** and the internal name is DISBALECUSTOMCONTEXTS.
- **Ditech VQA Inbound Setting**- If Ditech’s VQA (Voice Quality Application) is installed, then this setting will be used for all inbound calls. For more information, in the Asterisk CLI, issue the “core show application VQA” command. Default value is **7** and the internal name is DITECH\_VQA\_INBOUND.
- **Ditech VQA Outbound Setting**- If Ditech’s VQA (Voice Quality Application) is installed, then this setting will be used for all outbound calls. For more information, in the Asterisk CLI, issue the “core show application VQA” command. Default value is **7** and the internal name is DITECH\_VQA\_OUTBOUND.
- **Enable Custom Device States**- If this is set, it assumes that you are running Asterisk 1.4 or higher and want to take advantage of the fun\_devstate.c backport from Asterisk 1.6. This allows custom hints to be created to support BLF for server side feature codes such as daynight, Follow Me, etc. Default value is **True** and the internal name is USEDEVSTATE.
- **Extension Concurrency Limit**- Default maximum number of outbound simultaneous calls that an extension can make. This is also very useful as “Security Protection” against a system that has been compromised. It will limit the number of simultaneous calls that can be made on the compromised extension. This default is used when an extension is created. A default of **0** means no limit. The internal name is CONCURRENCYLIMITDEFAULT.
- **Feature Codes Beep Only**- When set to “True,” a beep is played instead of a confirmation message when activating/de-activating call forward, call waiting, daynight, DND and Find-MeFollow-Me. Default value is **False** and the internal name is FCBEEPONLY.
- **Force All Internal Auto Answer**- Force all extensions to operate in the “Internal Auto Answer”

mode regardless of their individual settings. (See “Internal Auto Answer Default” for more info). Default value is **False** and the internal name is `FORCE_INTERNAL_AUTO_ANSWER_ALL`.

- **Generate Diversion Headers**- If set to “True,” then calls going out of your outbound routes that originate from outside your PBX and were subsequently forwarded through a call forward, ring group, Follow Me or other means will have a SIP diversion header added to the call with the original incoming DID assuming there is a DID available. This is useful with some carriers as they may, under certain circumstances, require this. Default value is **False** and the internal name is `DIVERSIONHEADER`.
- **Internal Auto Answer Default**- This is a default setting for new extensions. When set to “Intercom,” calls to extensions/users from other internal users act as if they were intercom calls. This means they will be auto-answered if the endpoint supports this feature and the system is configured to operate in this mode. All the normal white and black list settings will be honored if they are set. External calls will still ring as normal as will certain other circumstances such as blind transfers and when a Follow Me is configured and enabled. If disabled, the phone rings normally. Default value is **disabled** and the internal name is `DEFAULT_INTERNAL_AUTO_ANSWER`.
- **NoOp Traces in Dialplan**- Some modules will generate lots of `NoOp()` commands proceeded by a `[TRACE]` (trace\_level) that can be used during development or while trying to trace call flows. These `NoOp()`s commands serve no other purpose, so if you don’t want to see excessive `NoOp()`s in your dialplan, set this to “0.” The higher the number, the more detailed level of `NoOp()`s will be generated. Default value is **0** and the internal name is `NOOPTRACE`.
- **Occupied Lines CW Busy**- For extensions that have CW enabled, report unanswered calls as busy (resulting in busy voicemail greeting). If set to “no,” unanswered calls report as no-answer. Default value is **True** and the internal name is `CWINUSEBUSY`.
- **Only Use Last CID Prepend**- Some modules allow the CNAM to be prepended. If a previous prepend was done, the default behavior is to remove the previous prepend and only use the most recent one. Setting this to “False” will turn that off allowing all prepends to be stacked in front of one another. Default value is **True** and the internal name is `CID_PREPEND_REPLACE`.
- **Polling Interval for Stopping Asterisk**- When Asterisk is stopped or restarted with the “ampportal stop/restart” commands, it does a graceful stop waiting for active channels to hang up. This sets the polling interval to check if Asterisk is shutdown and updates the countdown timer. Default value is **2** and the internal name is `ASTSTOPPOLLINT`.
- **Use bad-number Context**- Generate the “bad-number” context, which traps any bogus number or feature code and plays a message to that effect. If you use the “Early Dial” feature code on some Grandstream phones, you will want to set this to “False.” Default value is **True** and the internal name is `AMPBASNUMBER`.
- **Use Google DNS for Enum**- Setting this flag will generate the required global variable so the `enumlookup.agi` will use Google DNS 8.8.8.8 when performing an ENUM lookup. Not all DNS can deal with a NAPTR record, but Google does. The drawback is that Google tracks every lookup. If you are not comfortable with this, then do not enable this setting. Please read the Google FAW about this at <http://code.google.com/speed/public-dns/faw.html#privacy>. Default value is **False** and the internal name is `USEGOOGLEDNSFORENUM`.
- **Waiting Period to Stop Asterisk**- When Asterisk is stopped or restarted with the “ampportal stop/restart” commands, it does a graceful stop waiting for active channels to hang up. This sets the maximum time, in seconds, to wait prior to force Asterisk to stop. Default is **120** and

# Advanced Settings

## User Guide

the internal name is ASTSTOPTIMEOUT.

- **Display CallerID on Calling Phone**- When set to “True” and CONNECTEDLINE() capabilities are configured and supported by your handset, the CID value being transmitted on this call will be updated on your handset in the CNAM field prepended with CID. This is so you know what is presented to the caller (if the trunk supports and honors the setting of the transmitted CID). Default value is **True** and the internal name is OUTBOUND\_CID\_UPDATE.
- **Display Dialed Number on Calling Phone**- When set to “True” and CONNECTEDLINE() capabilities are configured and supported by your handset, the number actually dialed will be updated on your handset in the CNUM field. This allows you to see the final manipulation of your number after outbound route and trunk dial manipulation rules have been applied. For example, if you have configured 7 digit dialing on a North American dialplan, the ultimate 10 or 11 digit transmission will be displayed back. Any “Outbound Dial Prefixes” configured at the trunk level will NOT be shown as these are often analog line pauses (w) or other characters that distort the CNUM field on updates. Default value is **True** and the internal name is OUTBOUND\_DIAL\_UPDATE.
- **Use Automixmon for One-Touch Recording**- Starting with Asterisk 1.6, one-touch-recording can be toggled on and off during a call if the dial options had “x” and/or “X” options set. When this is set to “True,” the “In-Call Asterisk Toggle Call Recording” will use the asterisk “automixmon” option instead of “automon” to set this. Only one or the other can be set from the GUI. You need to set the proper options of “x” and/or “X” when using this or “w” and/or “W” if using the older “automon” version. Setting this to “True” will have no effect on systems running Asterisk 1.4 or earlier. Default value is **False** and the internal name is AUTOMIXMON.
- **Conference Room App**- The Asterisk application to use for conferencing. If only one is compiled into Asterisk, FreePBX will auto detect and change this value if set incorrectly. The app\_confbridge application is considered “experimental” with known issues and does not work on Asterisk 10 where it was completely rewritten and changed from the version in 1.6 and 1.8. Default value is **app\_meetme** and the internal name is ASTCONFAPP.

### Dialplan and Operational

Block CNAM on External Trunks ?	True False	
Call Forward Ringtimer Default ?	0	
CW Enabled by Default ?	True False	
Disable -custom Context Includes ?	True False	
Ditech VQA Inbound Settings ?	7	
Ditech VQA Outbound Setting ?	7	
Enable Custom Device States ?	True False	
Extension Concurrency Limit ?	5	
Feature Codes Beep Only ?	True False	
Force All Internal Auto Answer ?	True False	
Generate Diversion Headers ?	True False	
Internal Auto Answer Default ?	disabled	

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NoOp Traces in Dialplan ?	0
Occupied Lines CW Busy ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False
Only Use Last CID Prepend ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False
Polling Interval for Stopping Asterisk ?	2
Use bad-number Context ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False
Use Google DNS for Enum ?	<input type="checkbox"/> True <input checked="" type="checkbox"/> False
Waiting Period to Stop Asterisk ?	120
Display CallerID on Calling Phone ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False
Display Dialed Number on Calling Phone ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False
Use Automixmon for One-Touch Recording ?	<input type="checkbox"/> True <input checked="" type="checkbox"/> False
Conference Room App ?	app_meetme

## Flash Operator Panel

- These settings allow you to change the FOP password and sort mode.
  - **FOP Password**- Password for performing transfers and hangups in the FOP (Flash Operator Panel). Default value is **passw0rd** (can be left blank) and the internal name is FOPPASSWORD.
  - **FOP Sort Mode**- The FOP sorts' extensions by last name [lastname] or by extension [extension]. Default Value is **extension** and the internal name is FOPSORT.

Flash Operator Panel

FOP Password ?	passw0rd
FOP Sort Mode ?	extension

## Follow Me Module

- The advanced Follow Me module settings are useful if you plan to change defaults when creating new extensions.
  - **Create Follow Me at Extension Creation Time**- When creating a new user or extension, setting this to “True” will automatically create a new Follow Me for that user using the default settings listed below. Default value is **False** and the internal name is FOLLOWME\_AUTO\_CREATE.
  - **Disable Follow Me Upon Creation**- This is the default value for the Follow Me “Disable” setting. When first creating a Follow Me or if auto-created with a new extension, setting this to “True” will disable the Follow Me setting, which can be changed by the user or admin in multiple locations. Default value is **True** and the internal name is FOLLOWME\_DISABLED.

# Advanced Settings

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- **Default Follow Me Ring Time**- This is the default ring time for a Follow Me set upon creation and used if auto-created with a new extension. Default value is **20** and the internal name is FOLLOWME\_TIME.
- **Default Follow Me Initial Ring Time**- This is the default initial ring time for a Follow Me set upon creation and used if auto-created with a new extension. Default value is **7** and the internal name is FOLLOWME\_PRERING.
- **Default Follow Me Ring Strategy**- This is the default ring strategy selected for a Follow Me set upon creation and used if auto-created with a new extension. Default value is **ringallv2-prim** and the internal name is FOLLOWME\_RG\_STRATEGY.

### Follow Me Module

Create Follow Me at Extension Creation Time ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False
Disable Follow Me Upon Creation ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False
Default Follow Me Ring Time ?	<input type="text" value="20"/>
Default Follow Me Initial Ring Time ?	<input type="text" value="7"/>
Default Follow Me Ring Strategy ?	<input type="text" value="ringallv2-prim"/>

## GUI Behavior

- Using custom extensions is a great way to inform the PBX of the extension numbers you are using in scripts to keep the PBX from using the same number somewhere else resulting in a conflict.
- **Abort Config Gen on Bad Dest**- Setting this to “True” will result in retrieve\_conf aborting during a reload if an extension conflict is detected or if a destination is invalid. It is usually better to allow the reload to go through and then correct the problem. However, this can be set if a more strict behavior is desired. Default value is **False** and the internal name is BADDESTABORT.
- **Abort Config Gen on Exten Conflict**- Setting this to “True” will result in retrieve\_conf aborting during a reload if an extension conflict is detected or if a destination is invalid. It is usually better to allow the reload to go through and then correct the problem. However, this can be set if a more strict behavior is desired. Default value is **False** and the internal name is XTNCONFLICTABORT.
- **Check Server Referrer**- When set to the default value of “True,” all requests into the PBX that might possibly add/edit/delete settings will be validated to assure the request is coming from the server. This will protect the system from CSRF (Cross Site Request Forgery) attacks. It will have the effect of preventing legitimately entering URLs that could modify settings. Changing this to “False” essentially allows or ceases to prevent. Default value is **True** and the internal name is CHECKREFERER.
- **Include Server Name in Browser**- Precede browser title with the server name. Default value is **False** and the internal name is SERVERINTITLE.
- **Report Unknown Dest as Error**- If “False,” then the destination registry will not report unknown destinations as errors. This should be left in the default setting of **True** and custom

# Advanced Settings

## User Guide

destinations should be moved into the new custom apps registry. Internal name is CUSTOMASERROR.

- **Require Confirm with Apply Changes**- When set to “False,” the extra step of “confirm” is bypassed when you apply changes. Default value is **True** and the internal name is RELOADCONFIM.
- **Show Categories in Nav Menu**- This controls how menu items in the admin interface are sorted. Choose “True” to sort by category or “False” to sort alphabetically with no categories shown. Default value is **True** and the internal name is USECATEGORIES.
- **Use freepbx\_menu.conf Configuration**- When set to “True,” the system will check for a freepbx\_menu.conf amongst the normal configuration files and if found, it will be used to define and remap the menu tabs and contents. See the template supplied with FreePBX for details on how to do this. Default value is **False** and the internal name is USE\_FREEPBX\_MENU\_CONF.
- **Use wget For Module Admin**- Module Admin normally tries to get its online information through direct file open type calls to URLs that go back to the module server. If it fails, it's typically because of content filters in firewalls that do not like the way PHP formats the requests. The code will fall back and try a wget to pull the information. This will often solve the problem. However, in such an environment there can be a significant timeout before the failed file open calls to the URLs return and there are often 2-3 of these that occur. Setting this to “True” will force the PBX to avoid the attempt(s) to open the URL. Default value is **False** and the internal name is MODULEADMINWGET.
- **Dashboard Info Update Frequency**- Update rate, in seconds, of the Info section of the System Status panel. Default value is **30** and the internal name is DASHBOARD\_INFO\_UPDATE\_TIME.
- **Dashboard Max Calls Initial Scale**- Use this to pre-set the scale for maximum calls on the dashboard display. If not set, the scale is dynamically sized based on the active calls on the system. **No** default provided, field can be left blank and the acceptable values are **0-3000**. Internal name is MAXCALLS.
- **Dashboard Stats Update Frequency**- Update rate, in seconds, of all sections of the System Status panel except the Info box. Default value is **6** and the internal name is DASHBOARD\_STATS\_UPDATE\_TIME.

### GUI Behavior

Abort Config Gen on Bad Dest ?	True	False
Abort Config Gen on Exten Conflict ?	True	False
Check Server Referrer ?	True	False
Include Server Name in Browser ?	True	False
Report Unknown Dest as Error ?	True	False
Require Confirm with Apply Changes ?	True	False
Show Categories in Nav Menu ?	True	False
Use freepbx_menu.conf Configuration ?	True	False
Use wget For Module Admin ?	True	False
Dashboard Info Update Frequency ?	30	
Dashboard Max Calls Initial Scale ?		
Dashboard Stats Update Frequency ?	6	

### Paging

- Limiting the maximum paging participants is an important feature on systems with large numbers of users as paging uses quite a bit of system resources.
  - **Max Paging Participants**- Maximum number of participants allowed in a page. Default is **40** and the internal name is PAGINGMAXPARTICIPANTS.

Paging

Max Paging Participants ?	40
---------------------------	----

### Queues Module

- Most of the advanced Queues module settings are concerning systems running versions prior to Asterisk 1.8.
  - **Set Agent Name in CDR dstchannel**- Only valid with Asterisk 1.6 and higher. This option is implemented to mimic chan\_agents behavior of populating CDR dstchannel field of a call with an agent name, which is set if available at the login time with the “AddQueueMember membername” parameter or with static members. Default value is **False** and the internal name is QUEUES\_UPDATECDR.
  - **Use MixMonitor for Recordings**- Setting this to “True” will use the MixMonitor application instead of Monitor. The concept of “joining/mixing” the in/out files now goes away when this is enabled. Default value is **True** and the internal name is QUEUES\_MIX\_MONITOR.
  - **Hide Queue No Answer Option**- It is possible for a queue to NOT answer a call and still enter callers to the queue. The normal behavior is that all callers are answered before entering the queue. If the call is not answered, it is possible that some early media delivery would still allow callers to hear recordings, MOH, etc. However, this can be inconsistent and potentially unreliable. For that reason, this option is not displayed by default. If a queue is set to not answer, then the setting will be displayed for that queue regardless of this setting. Default value is **True** and the internal name is QUEUES\_HIDE\_NOANSWER.
  - **Asterisk Queues Patch 15168 Installed**- Setting this flag will generate the required dialplan to integrate with the following Asterisk patch:  
<https://issues.asterisk.org/view.php?id=15168>. This setting is obsolete on Asterisk 1.8+ systems where the hint state is now standard and always used. This Asterisk patch is only available on Asterisk 1.4 and trying to use it on Asterisk 1.6 will break some queue behavior and should be avoided. Default value is **False** and the internal name is USEQUEUESTATE.
  - **Generate queuenum\*/\*\* Login/off Codes**- Queue login and logout codes were historically queuenum\* and queuenum\*\*. For the most part, these have been replaced by the \*45 queue toggle codes. The legacy codes are required to login/logout a third party user that is not the extension dialing. These can be removed from the system by setting this to “False.” Default value is **True** and the internal name is GENERATE\_LEGACY\_QUEUE\_CODES.

# Advanced Settings

## User Guide

### Queues Module

Set Agent Name in CDR dstchannel <sup>?</sup>	True	False
Use MixMonitor for Recordings <sup>?</sup>	True	False
Hide Queue No Answer Option <sup>?</sup>	True	False
Asterisk Queues Patch 15168 Installed <sup>?</sup>	True	False
Generate queuenum*/** Login/off Codes <sup>?</sup>	True	False

### Ring Groups Module

- This setting changes whether or not to display ring group info on an extension page (if the extension is a ring group member). See the “Ring Group Membership” graphic for an example.

#### ▪ Ring Group Membership

#### 600 Local

- **Display Extension Ring Group Members**- When set to “True,” extensions that belong to one or more ring groups will have a ring group section and link back to each group they belong to. Default value is **False** and the internal name is EXTENSION\_LIST\_RINGGROUPS.

### Ring Group Module

Display Extension Ring Group Members <sup>?</sup>	True	False
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### System Setup

- The System Setup advanced settings allow for some fine-tuning in the overall operation of the system. In general, the only setting of interest is the User & Devices mode.
- **PBX Log Routing**- Determine where to send log information if the log is enabled (‘Disable PBX’ AMPDISABLELOG). There are two places you can route the logs. “FILE” will send all log messages to the defined “PBX Log File” (FPBX\_LOG\_FILE). All the other settings will route the log messages to your system, logging subsystem (syslog) using the specified log level. This can be configured to route different levels to different locations. See syslog documentation (man syslog) on your system for more details. Default value is **FILE** and the internal name is AMPSYSLOGLEVEL.
- **Disable PBX Log**- Whether or not to invoke the PBX log facility. Default value is **False** and

the internal name is AMPDISABLELOG.

- **Log Verbose Messages**- The PBX has many verbose and useful messages displayed to users during module installation, system installations, loading configurations and other places. In order to accumulate these messages in the log files as well as the on screen display, set this to “True.” Default value is **True** and the internal name is LOG\_OUT\_MESSAGES.
- **Send Dashboard Notifications to Log**- When enabled, all notification updates to the dashboard notification panel will also be logged into the specified log file (when enabled). Default value is **True** and the internal name is LOG\_NOTIFICATIONS.
- **PBX Log File**- Full path and name of the PBX Log File used in conjunction with the syslog level (AMPSYSLOGLEVEL) being set to FILE, not used otherwise. Initial installs may have some early logging sent to /tmp/freepbx\_pre\_install.log when it is first bootstrapping the installer. Default value is **/var/log/asterisk/freepbx.log** and the internal name is FPBX\_LOG\_FILE.
- **PHP Error Log Output**- Where to send PHP errors, warnings and notices by the PBX PHP error handler. Set to “debug” and they will go to the debug file regardless of whether debug logging is disabled or not. Set to “freepbxlog” and these will be sent to the PBX Log. Set to “off” and they will be ignored. Default value is **debug** and the internal name is PHP\_ERROR\_HANDLER\_OUTPUT.
- **User & Devices Mode**- This sets the extension behavior in the PBX. If set to “extensions,” devices and users are administered together as a unified extension and appear on a single page. If set to “deviceanduser,” devices and users will be administered separately. Devices (for example, each individual line on a SIP phone) and users (for example, 101 or 102) will be configured independent of each other. “deviceanduser” allows association of one user to many devices and/or allows users to log in and out of devices. Default value is **extensions** and the internal name is AMPEXTENSIONS.
- **Allow Login With DB Credentials**- When set to “True,” admin access to the PBX GUI will be allowed using the PBX configured AMPDBUSER and AMPDBPASS credentials. This only applies when the authorization type is “database” mode. Default value is **False** and the internal name is AMP\_ACCESS\_DB\_CREDS.
- **User Portal Admin Username**- This is the default admin name used to allow an administrator to login to ARI, bypassing all security. Change this to whatever you want and do not forget to change the “User Portal Admin Password” as well. **No** default provided, field may be left blank. The internal name is ARI\_ADMIN\_USERNAME.
- **User Portal Admin Password**- This is the default admin password used to allow an administrator to login to ARI, bypassing all security. Change this to a secure password. Default value is **ari\_password** and the internal name is ARI\_ADMIN\_PASSWORD.
- **Asterisk VMU Mask**- Asterisk defaults this to “077,” allowing only the Asterisk user to have any permission on VM files. If set to something like “007,” it will allow the group to have permissions. This can be used if setting apache to a different user than Asterisk, so that the apache user (and thus ARI) can have read/write/delete access to the VM files. If changed, some of the voicemail directory structures may have to be manually changed. Default value is **077** and the internal name is AMPVMUMASK.
- **Browser Stats**- Setting this to “True” will allow the development team to use Google analytics to anonymously analyze browser information to help make better development decisions. Default value is **True** and the internal name is BROWSER\_STATS.
- **PBX Web Address**- This is the address of your web server. It is mostly obsolete and derived

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## User Guide

when not supplied and will be phased out. There are still some areas expecting a variable to be set and if you are using it, this will migrate your value. **No** default provided, field can be left blank. Internal name is AMPWEBADDRESS.

- **Use Google Distribution Network for js Downloads**- Setting this to “True” will fetch system javascript libraries such as jQuery and jQuery-ui from ajax.googleapis.com. This can be advantageous if accessing remote or multiple different FreePBX systems since the libraries are only cached once in your browser. If external Internet connections are problematic, setting this to “True” could result in slow systems. FreePBX will always fallback to the locally available libraries if the CDN is not available. Default value is **False** and the internal name is USE\_GOOGLE\_CDN\_JS.
- **Convert Music Files to WAV**- When set to “False,” the MP3 files can be loaded and WAV files converted to MP3 in the MOH module. The default behavior of “True” assumes that mpg123/sox loaded and this will convert MP3 files to WAV. This is highly recommended as MP3 files heavily tax the system and can cause instability on a busy phone system. Default value is **True** and the internal name is AMPMPG123.
- **Dashboard Non-Std SSH Port**- SSH port number configured on your system when not using the default SSH port of 22. This allows dashboard monitoring to watch the proper port. **No** default provided, field can be left blank and has acceptable values of **1-65535**. The internal name is SSHPORT.
- **Recordings Crypt Key**- Crypt key used by this recording module when accessing the recording files. Change from the default of “moufduu3nma0” if desired. **No** default provided, field can be left blank. The internal name is AMPPLAYKEY.
- **Use Old Parking Patch**- Set to generate some additional dialplan if using a particular patch that was available in Asterisk 1.2 and 1.4 to add behavior to parking, such as adding alert-info and CID prepends to timed out parked calls. This patch is mostly obsolete and this setting will probably go away at some point when Asterisk 1.6+ parking enhancements are added to the Parking module. Default value is **False** and the internal name is PARKINGPATCH.

### System Setup

PBX Log Routing ?	FILE	
Disable PBX Log ?	True False	
Log Verbose Messages ?	True False	
Send Dashboard Notifications to Log ?	True False	
PBX Log File ?	/var/log/asterisk/freepbx.log	
PHP Error Log Output ?	debug	
User & Devices Mode ?	extensions	
Allow Login With DB Credentials ?	True False	
User Portal Admin Username ?		
User Portal Admin Password ?	ari_password	
Asterisk VMU Mask ?	007	
Browser Stats ?	True False	
PBX Web Address ?		
Use Google Distribution Network for js Downloads ?	True False	
Convert Music Files to WAV ?	True False	
Dashboard Non-Std SSH Port ?		
Recordings Crypt Key ?		
Use Old Parking Patch ?	True False	

### Time Condition Module

- The Time Condition module advanced settings can assist in reducing output when debugging and/or help certain BLF states stay accurate.
  - **Enable Maintenance Polling**- If set to “False,” this will override the execution of the time conditions maintenance task launched by call files. If all the feature codes for time conditions are disabled, the maintenance task will not be launched anyhow. Setting this to “False” would be fairly un-common. You may want to set this temporarily if debugging a system to avoid the periodic dialplan (the maintenance task launches) running through the CLI as it can be distracting. Default value is **True** and the internal name is TCMAINT.
  - **Maintenance Polling Interval**- The polling interval in seconds used by the time conditions maintenance task, launched by an Asterisk call files used to update time condition override states as well as keep custom device state hint values up-to-date when being used with BLF. A shorter interval will assure that BLF key states are accurate. The interval should be less than the shortest configured span between two time condition states, so that a manual override during such a period is properly reset when the new period starts. Default value is **60** and the internal name is TCINTERVAL.

Time Condition Module

Enable Maintenance Polling ?

True

False

Maintenance Polling Interval ?

60

### Voicemail Module

- Control the display of IMAP credential fields in the extension page.
  - **Provide IMAP Voicemail Fields**- Installations that have configured voicemail with IMAP should set this to “True” so that the IMAP username and password fields are provided in the voicemail setup section in extensions. If an extension already has these fields populated, they will be displayed even if this is set to “False.” Default value is **False** and the internal name is WM\_SHOW\_IMAP.

Voicemail Module

Provide IMAP Voicemail Fields ?

True

False

### Recap

- Advanced settings can be quite useful when first deploying a system and when doing development. One must use extreme caution when making any changes in this module and it is advised that your first step is to always make a backup.

### Appendix A

- Appendix A is a listing of additional advanced settings exposed when “Display Hidden Settings” is set to “True.”

### Asterisk Manager (hidden settings)

- The Asterisk Manager hidden settings allow for more granular control of the Asterisk Manager. Most users will want to leave the defaults as is.
  - **Asterisk Manager Host**- Hostname for the Asterisk Manager. Default value is **localhost** and the internal name is ASTMANAGERHOST.
  - **Asterisk Manager Port**- Port for the Asterisk Manager. Default value is **5038** and acceptable values are **1024-65535**. The internal name is ASTMANAGERPORT.
  - **Asterisk Manager Proxy Port**- Optional port for an Asterisk Manager Proxy. **No** default provided, field can be left blank and has acceptable values of **1024-65535**. Internal name is ASTMANAGERPROXYPORT.
  - **Asterisk Manager Write Timeout**- Timeout in ms, for write timeouts for cases where Asterisk disconnects frequently. Default value is **5000** and can be left blank. The internal name is ASTMGRWRITETIMEOUT.

#### Asterisk Manager

Asterisk Manager Host ?	<input type="text" value="localhost"/>
Asterisk Manager Password ?	<input type="text" value="amp111"/>
Asterisk Manager Port ?	<input type="text" value="5038"/>
Asterisk Manager Proxy Port ?	<input type="text"/>
Asterisk Manager User ?	<input type="text" value="admin"/>
Asterisk Manager Write Timeout ?	<input type="text" value="5000"/>

### Developer and Customization (hidden settings)

- The Developer and Customization hidden settings are useful for development and in certain call center applications.
  - **AMPLOCALBIN Dir for retrieve\_conf**- If this directory is defined, “retrieve\_conf” will check for a file called “retrieve\_conf\_post\_custom.” If that file exists, it will be included after other processing, thus having access to the full environment for additional customization. **No** default provided, field can be left blank. The internal name is AMPLOCALBIN.
  - **Disable Module Admin Caching**- Module Admin caches a copy of the online XML document that describes what is available on the server. Subsequent online update checks will use that cached information if it is less the 5 minutes old. To bypass the cache and force it to go to the server each time, set this to “True.” This should normally be set to “False,” but can be useful

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during testing. Default value is **False** and the internal name is MODULE\_ADMIN\_SKIP\_CACHE.

- **POST\_RELOAD Script**- Automatically execute a script after applying changes in the AMP Admin. Set POST\_RELOAD to the script you wish to execute after applying changes. If POST\_RELOAD\_DEBUG=true, you will see the output of the script in the web page. **No** default provided, field can be left blank. The internal name is POST\_RELOAD.
- **PRE\_RELOAD Script**- Optional script to run just prior to doing an extension reload to Asterisk through the manager after pressing “Apply Configuration Changes” in the GUI. **No** default provided, field can be left blank. The internal name is PRE\_RELOAD.
- **Post Call Recording Script**- An optional script to be run after the call has been disconnected. You can include the “MixMon” variables such as \${CALLFILENAME}, \${MIXMON\_FORMAT} and \${MIXMON\_DIR}. To ensure that your variables are properly escaped, use the following notation: ^{MY\_VAR}. **No** default provided, field can be left blank. The internal name is MIXMON\_POST.

### Device Settings

Show all Device Setting on Add?	<input type="radio"/> True <input checked="" type="radio"/> False
Require Strong Secrets?	<input checked="" type="radio"/> True <input type="radio"/> False
Remove mailbox Setting when no Voicemail?	<input type="radio"/> True <input checked="" type="radio"/> False
SIP canrenivate (directmedia)?	<input type="text" value="no"/>
SIP trustpid?	<input type="text" value="yes"/>
SIP sendrpid?	<input type="text" value="no"/>
SIP nat?	<input type="text" value="yes"/>
SIP encryption?	<input type="text" value="no"/>
SIP qualifyfreq?	<input type="text" value="60"/>
SIP and IAX qualify?	<input type="text" value="yes"/>
SIP and IAX allow?	<input type="text"/>
SIP and IAX disallow?	<input type="text"/>
SIP and DAHDi callgroup?	<input type="text"/>
SIP and DAHDi pickupgroup?	<input type="text"/>

### Dialplan and Operational (hidden settings)

- The settings exposed (not hidden) offer a few additional settings that are generally best to leave as default.
- **Convert ZAP Settings to DAHDi**- If set to “True,” the PBX will check if you have chan\_dahdi installed and if so, it will automatically use all your ZAP configuration settings (devices and trunks) and silently convert them, under the covers, to DAHDi so no changes are needed. The GUI will continue to refer to these as ZAP, but it will use the proper DAHDi channels. This will

# Advanced Settings

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also keep ZAP Channel DID's working. Default value is **False** and the internal name is ZAP2DAHDICOMPAT.

- **Dynamically Generate Hints**- If set to "True," core will not statically generate hints, but instead make a call to the AMPBIN PHP script and generate\_hints.php through an Asterisk #exec call. This requires asterisk.conf to be configured with the "execincludes=yes" in the [options] section. Default value is **False** and the internal name is DYNAMICHINTS.
- **Compressed Copy of Main CSS**- For internal use only. **No** default provided, field can be left blank. The internal name is MAINSTYLE\_CSS\_GENERATED.

### Dialplan and Operational

Block CNAM on External Trunks <sup>?</sup>	True False	
Call Forward Ringtimer Default <sup>?</sup>	0 <sup>?</sup>	
Convert ZAP Settings to DAHDi <sup>?</sup>	True False	
CW Enabled by Default <sup>?</sup>	True False	
Disable -custom Context Includes <sup>?</sup>	True False	
Ditech VQA Inbound Settings <sup>?</sup>	7 <sup>?</sup>	
Ditech VQA Outbound Setting <sup>?</sup>	7 <sup>?</sup>	
Dynamically Generate Hints <sup>?</sup>	True False	
Enable Custom Device States <sup>?</sup>	True False	
Extension Concurrency Limit <sup>?</sup>	5 <sup>?</sup>	↻
Feature Codes Beep Only <sup>?</sup>	True False	
Force All Internal Auto Answer <sup>?</sup>	True False	
Generate Diversion Headers <sup>?</sup>	True False	
Internal Auto Answer Default <sup>?</sup>	disabled <sup>?</sup>	
NoOp Traces in Dialplan <sup>?</sup>	0 <sup>?</sup>	
Occupied Lines CW Busy <sup>?</sup>	True False	
Only Use Last CID Prepend <sup>?</sup>	True False	
Polling Interval for Stopping Asterisk <sup>?</sup>	2 <sup>?</sup>	
Use bad-number Context <sup>?</sup>	True False	
Use Google DNS for Enum <sup>?</sup>	True False	
Waiting Period to Stop Asterisk <sup>?</sup>	120 <sup>?</sup>	
Display CallerID on Calling Phone <sup>?</sup>	True False	
Display Dialed Number on Calling Phone <sup>?</sup>	True False	
Use Automixmon for One-Touch Recording <sup>?</sup>	True False	
Compressed Copy of Main CSS <sup>?</sup>	assets/css/mstyle_autogen_1340226255.css	
Conference Room App <sup>?</sup>	app_meetme <sup>?</sup>	

### Directory Layout (hidden settings)

- The Directory Layout advanced settings allow changes for adaption to certain Asterisk installations.
  - **Asterisk AGI Dir**- This is the default directory for the Asterisk agi files. Default is `/var/lib/asterisk/agi-bin` and the internal name is ASTAGIDIR.
  - **Asterisk bin Dir**- This is the default directory for the Asterisk lib files. Default is `/var/lib/asterisk/` and the internal name is ASTVARLIBDIR.
  - **Asterisk etc Dir**- This is the default directory for the Asterisk configuration files. Default is `/etc/asterisk/` and the internal name is ASTETCDIR.
  - **Asterisk Log Dir**- This is the default directory for the Asterisk log files. Default is `/var/log/asterisk/` and the internal name is ASTLOGDIR.
  - **Asterisk Modules Dir**- This is the default directory for the Asterisk modules. Default is `/usr/lib/asterisk/modules` and the internal name is ASTMODDIR.
  - **Asterisk Run Dir**- This is the default directory for the Asterisk run files. Default is `/var/run/asterisk/` and the internal name is ASTRUNDIR.
  - **Asterisk Spool Dir**- This is the default directory for the Asterisk spool directory. Default is `/var/spool/asterisk/` and the internal name is ASTSPOOLDIR.
  - **CGI Dir**- The path to the Apache cgi-bin directory (leave off the trailing slash). Default value is `/var/www/cgi-bin` and the internal name is AMPCGIBIN.
  - **PBX bin Dir**- Location of the PBX command line scripts. Default value is `/var/lib/asterisk/bin` and the internal name is AMPBIN.
  - **PBX sbin Dir**- Where (root) command line scripts are located. Default value is `/usr/sbin` and the internal name is AMPSBIN.
  - **PBX Web Root Dir**- The path to Apache webroot (leave off the trailing slash). Default value is `/var/www/html` and the internal name is AMPWEBROOT.
  - **MoH Subdirectory**- This is the subdirectory for the MoH files/directories located in the ASTVARLIBDIR. Older installations may be using mohmp3, which was the old Asterisk default and should be set to that value if the music files are located there relative to the ASTVARLIBDIR. Default value is `moh` and the internal name is MOHDIR.
  - **Override Call Recording Location**- Override the default location where Asterisk will store call recordings. Be sure to set proper permissions on the directory for the Asterisk user. **No** default provided, field can be left blank. The internal name is MIXMON\_DIR.

#### Directory Layout

Asterisk AGI Dir ?	<input type="text" value="/var/lib/asterisk/agi-bin"/>
Asterisk bin Dir ?	<input type="text" value="/var/lib/asterisk"/>
Asterisk etc Dir ?	<input type="text" value="/etc/asterisk"/>
Asterisk Log Dir ?	<input type="text" value="/var/log/asterisk"/>
Asterisk Modules Dir ?	<input type="text" value="/usr/lib/asterisk/modules"/>
Asterisk Run Dir ?	<input type="text" value="/var/run/asterisk"/>
Asterisk Spool Dir ?	<input type="text" value="/var/spool/asterisk"/>

CGI Dir ?	<input type="text" value="/var/www/cgi-bin"/>
PBX bin Dir ?	<input type="text" value="/var/lib/asterisk/bin"/>
PBX sbin Dir ?	<input type="text" value="/usr/sbin"/>
PBX Web Root Dir ?	<input type="text" value="/var/www/html"/>
MoH Subdirectory ?	<input type="text" value="moh"/>
Override Call Recording Location ?	<input type="text"/>

### Flash Operator Panel (hidden settings)

- Useful as other operator panel's are being integrated to the PBX.
  - **FOP Web Root Dir**- Path to the Flash Operator Panel webroot or other modules providing such functionality (leave off the trailing slash). **No** default provided, field can be left blank. The internal name is FOPWEBROOT.

Flash Operator Panel

FOP Password ?	<input type="text" value="passw0rd"/>
FOP Sort Mode ?	<input type="text" value="extension"/>
FOP Web Root Dir ?	<input type="text" value="xactview"/>

### Internal Use (hidden settings)

- In general, these are for internal use only.
  - **Asterisk Application VQA**- Set to the application name if the application is present in this Asterisk installation. **No** default provided, field can be left blank. The internal name is AST\_APP\_VQA.
  - **Asterisk Function CONNECTEDLINE**- Set to the function name if the function is present in this Asterisk installation. **No** default provided, field can be left blank. The internal name is AST\_FUNC\_CONNECTEDLINE.
  - **Asterisk Function DEVICE\_STATE**- Set to the function name if the function is present in this Asterisk installation. **No** default provided, field can be left blank. The internal name is AST\_FUNC\_DEVICE\_STATE.
  - **Asterisk Function EXTENSION\_STATE**- Set to the function name if the function is present in this Asterisk installation. **No** default provided, field can be left blank. The internal name is AST\_FUNC\_EXTENSION\_STATE.
  - **Asterisk Function MASTER\_CHANNEL**- Set to the function name if the function is present in this Asterisk installation. **No** default provided, field can be left blank. The internal name is AST\_FUNC\_MASTER\_CHANNEL.
  - **Asterisk Function SHARED**- Set to the function name if the function is present in this Asterisk installation. **No** default provided, field can be left blank. The internal name is AST\_FUNC\_SHARED.
  - **Asterisk Version**- Last Asterisk version detected (or forced). **No** default provided, field can

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be left blank. The internal name is ASTVERSION.

- **Last Mod Time of Main CSS**- For internal use only. **No** default provided, field can be left blank. The internal name is MAINSTYLE\_CSS\_MTIME.
- **Browser Stats Notice**- Internal use to track if “Notice” has been given that anonymous browser stats are being collected. Default value is **False** and the internal name is NOTICE\_BROWSER\_STATS.
- **Repo Server**- This is the URL(s) for the online PBX module file repository. Default value is set on installation and is recommended that it not be changed. The internal name is MODULE\_REPO.


Internal Use

Asterisk Application VQA ?	<input type="text"/>
Asterisk Function CONNECTEDLINE ?	CONNECTEDLINE
Asterisk Function DEVICE_STATE ?	DEVICE_STATE
Asterisk Function EXTENSION_STATE ?	EXTENSION_STATE
Asterisk Function MASTER_STATE ?	MASTER_CHANNEL
Asterisk Function SHARED ?	SHARED
Asterisk Version ?	1.8.17.0
Last Mod Time of Main CSS ?	1315795767
Browser Stats Notice ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False
Repo Server ?	<input type="text" value="http://mirror.pbxact.com/"/>

## Queues Module (hidden settings)

- Queues module advanced settings allow a slightly more granular control when using multiple queues and agents.
  - **Persistent Members**- Store each dynamic member in each queue in the astdb so that when Asterisk is restarted, members will be automatically read into their recorded queues. Default value is **True** and the internal name is QUEUES\_PERSISTANTMEMBERS.
  - **Honor Wrapup Time Across Queues**- Only valid with Asterisk 1.6+. This will make the last call and calls received be the same as the members logged in more than one queue. This is useful to make the queue respect the wrapuptime of another queue for a shared member. Default value is **True** and the internal name is QUEUES\_SHARED\_LASTCALL.








Queues Module

Persistent Members ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False	
Honor Wrapup Time Across Queues ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False	
Set Agent Name in CDR dstchannel ?	<input type="checkbox"/> True <input checked="" type="checkbox"/> False	
Use MixMonitor for Recordings ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False	
Hide Queue No Answer Option ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False	
Asterisk Queues Patch 15168 Installed ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False	
Generate queuenum*/** Login/off Codes ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False	

### Remote CDR Database (hidden settings)

- In some Asterisk installations, it is desirable to house the CDR database on a remote host. These settings help facilitate that.
  - **Remote CDR DB Host**- DO NOT set this unless you know what you are doing. Only used if you do not use the default values provided by the PBX. Hostname of db server is not the same as AMPDBHOST. **No** default provided, field can be left blank. The internal name is CDRDBHOST.
  - **Remote CDR DB Name**- DO NOT set this unless you know what you are doing. Only used if you do not use the default values provided by the PBX. It is the name of the database used for cdr records. **No** default provided, field can be left blank. The internal name is CDRDBNAME.
  - **Remote CDR DB Password**- DO NOT set this unless you know what you are doing. Only used if you do not use the default values provided by the PBX. Password for connecting to db is not the same as AMPDBPASS. **No** default provided, field can be left blank. The internal name is CDRDBPASS.
  - **Remote CDR DB Port**- DO NOT set this unless you know what you are doing. Only used if you do not use the default values provided by the PBX. It is the port number for db host. **No** default provided, field can be left blank and acceptable values are **1024-65535**. The internal name is CDRDBPORT.
  - **Remote CDR DB Table**- DO NOT set this unless you know what you are doing. Only used if you do not use the default values provided by the PBX. It is the db table where the cdr is stored, typically called cdr. **No** default provided, field can be left blank. The internal name is CDRDBTABLENAME.
  - **Remote CDR DB Type**- DO NOT set this unless you know what you are doing. Only used if you do not use the default values provided by the PBX. Defaults to your configured AMDBENGINE. **No** default provided, field can be left blank. The internal name is CDRDBTYPE.
  - **Remote CDR DB User**- DO NOT set this unless you know what you are doing. Only used if you do not use the default values provided by the PBX. Username for connecting to the db is not the same as AMPDBUSER. **No** default provided, field can be left blank. The internal name is CDRDBUSER.

#### Remote CDR Database

Remote CDR DB Host 	<input type="text"/>
Remote CDR DB Name 	<input type="text"/>
Remote CDR DB Password 	<input type="password"/>
Remote CDR DB Port 	<input type="text"/>
Remote CDR DB Table 	<input type="text"/>
Remote CDR DB Type 	<input type="text"/>
Remote CDR DB User 	<input type="text"/>

### Styling and Logos

- Styling and Logos advanced settings offer control of many graphical elements.
  - **tiff2pdf Author**- Author to pass to tiff2pdf's -a option. Default value is set on install. The internal name is PDFAUTHOR.
  - **Image: Left Upper**- Left upper logo. Path is relative to admin. Default is **images/tango.png** and the internal name is BRAND\_IMAGE\_TANGO\_LEFT.
  - **Image: Footer**- Logo in footer. Path is relative to admin. Default is **images/freepbx\_small.png** and the internal name is BRAND\_IMAGE\_FREEPBX\_FOOT.
  - **Image: Footer**- Logo in footer. Path is relative to admin. Default is **images/schmooze-logo.png** and the internal name is BRAND\_IMAGE\_SPONSOR\_FOOT.
  - **Alt for Left Logo**- Alternate to use in place of image and title hover value. Defaults to FreePBX. **No** default provided, field can be left blank. The internal name is BRAND\_FREEPBX\_ALT\_LEFT.
  - **Alt for Footer Logo**- Alternate to use in place of image and title hover value. Defaults to FreePBX. Default value is **FreePBX®**, field can be left blank. The internal name is BRAND\_FREEPBX\_ALT\_FOOT.
  - **Alt for Footer Logo**- Alternate to use in place of image and title hover value. Defaults to FreePBX. Default value **www.schmoozecom.com**, field can be left blank. The internal name is BRAND\_SPONSOR\_ALT\_FOOT.
  - **Link for Left Logo**- Link to follow when clicking on the logo. Defaults to **http://www.freepbx.org**. The internal name is BRAND\_IMAGE\_FREEPBX\_LINK\_LEFT.
  - **Link for Footer Logo**- Link to follow when clicking on the logo. Defaults to **http://www.freepbx.org**. The internal name is BRAND\_IMAGE\_FREEPBX\_LINK\_FOOT.
  - **Link for Sponsor Footer Logo**- Link to follow when clicking on the sponsor logo. Default value is **www.schmoozecom.com**, field can be left blank. The internal name is BRAND\_IMAGE\_SPONSOR\_LINK\_FOOT.
  - **Primary CSS Stylesheet**- Set this to replace the default mainstyle.css style sheet with your own, relative to admin. **No** default provided, field can be left blank. The internal name is BRAND\_CSS\_ALT\_MAINSTYLE.
  - **Optional Additional CSS Stylesheet**- Optional custom CSS style sheet included after the primary one and module specific ones are loaded, relative to admin. **No** default provided, field can be left blank. The internal name is BRAND\_CSS\_CUSTOM.
  - **View: freepbx\_admin.php**- This should never be changed except for very advanced layout changes. Default value is **views/freepbx\_admin.php** and the internal name is VIEW\_FREEPBX\_ADMIN.
  - **View: freepbx.php**- This should never be changed except for very advanced layout changes. Default value is **views/freepbx.php** and the internal name is VIEW\_FREEPBX.
  - **View: freepbx\_reload.php**- This should never be changed except for very advanced layout changes. Default value is **views/freepbx\_reload.php** and the internal name is VIEW\_FREEPBX\_RELOAD.
  - **View: freepbx\_reloadbar.php**- This should never be changed except for very advanced layout changes. Default value is **views/freepbx\_reloadbar.php** and the internal name is VIEW\_FREEPBX\_RELOADBAR.
  - **View: welcome.php**- This should never be changed except for very advanced layout

# Advanced Settings

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changes. Default value is **views/welcome.php** and the internal name is VIEW\_WELCOME.

- **View: welcome\_nomanager.php**- This should never be changed except for very advanced layout changes. Default value is **views/welcome\_nomanager.php** and the internal name is VIEW\_WELCOME\_NOMANAGER.
- **View: menuitem\_disabled.php**- This should never be changed except for very advanced layout changes. Default value is **views/menuitem\_disabled.php** and the internal name is VIEW\_MENUITEM\_DISABLED.
- **View: noaccess.php**- This should never be changed except for very advanced layout changes. Default value is **views/noaccess.php** and the internal name is VIEW\_NOACCESS.
- **View: unauthorized.php**- This should never be changed except for very advanced layout changes. Default value is **views/unauthorized.php** and the internal name is VIEW\_UNAUTHORIZED.
- **View: bad\_refferer.php**- This should never be changed except for very advanced layout changes. Default value is **views/bad\_refferer.php** and the internal name is VIEW\_BAD\_REFFERER.
- **View: loggedout.php**- This should never be changed except for very advanced layout changes. Default value is **views/loggedout.php** and the internal name is VIEW\_LOGGEDOUT.
- **View: panel.php**- This should never be changed except for very advanced layout changes. Default value is **views/panel.php** and the internal name is VIEW\_PANEL.
- **View: reports.php**- This should never be changed except for very advanced layout changes. Default value is **views/reports.php** and the internal name is VIEW\_REPORTS.
- **View: menu.php**- This should never be changed except for very advanced layout changes. Default value is **views/menu.php** and the internal name is VIEW\_MENU.
- **jQuery UI css**- CSS file for jQuery UI. Default value is **assets/css/jquery-ui.css** and the internal name is JQUERY\_CSS.
- **View: login.php**- This should never be changed except for very advanced layout changes. Default value is **views/login.php** and the internal name is VIEW\_LOGIN.
- **View: header.php**- This should never be changed except for very advanced layout changes. Default value is **views/header.php** and the internal name is VIEW\_HEADER.
- **View: footer.php**- This should never be changed except for very advanced layout changes. Default value is **views/footer.php** and the internal name is VIEW\_FOOTER.
- **Alternate JS**- Alternate JS file to supplement legacy.script.js. **No** default provided, field can be left blank. The internal name is BRAND\_ALT\_JS.
- **View: footer\_content.php**- This should never be changed except for very advanced layout changes. Default value is **views/footer\_content.php** and the internal name is VIEW\_FOOTER\_CONTENT.

### Styling and Logos

tiff2pdf Author

www.schmoozecom.com

Image: Left Upper?

modules/shmzskin/assets/images/pbxact-logo.png

Image: Footer?

Image: Footer?

images/schmooze-logo.png

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Alt for Left Logo <sup>?</sup>	<input type="text"/>
Alt for Footer Logo <sup>?</sup>	<input type="text"/>
Alt for Footer Logo <sup>?</sup>	<input type="text" value="www.schmoozecom.com"/>
Link for Left Logo <sup>?</sup>	<input type="text" value="http://www.pbxact.com"/>
Link for Footer Logo <sup>?</sup>	<input type="text"/>
Link for Sponsor Footer Logo <sup>?</sup>	<input type="text" value="http://www.schmoozecom.com"/>
Primary CSS Stylesheet <sup>?</sup>	<input type="text"/>
Optional Additional CSS Stylesheet <sup>?</sup>	<input type="text" value="modules/shmzskin/assets/css/custom.css"/>
View: freepbx_admin.php <sup>?</sup>	<input type="text" value="views/freepbx_admin.php"/>
View: freepbx.php <sup>?</sup>	<input type="text" value="views/freepbx.php"/>
View: freepbx_reload.php <sup>?</sup>	<input type="text" value="views/freepbx_reload.php"/>
View: freepbx_reloadbar.php <sup>?</sup>	<input type="text" value="views/freepbx_reloadbar.php"/>
View: welcome.php <sup>?</sup>	<input type="text" value="/var/www/html/admin/modules/shmzskin/views/welcome"/>
View: welcome_nomanager.php <sup>?</sup>	<input type="text" value="/var/www/html/admin/modules/shmzskin/views/welcome"/>
View: menuitem_disabled.php <sup>?</sup>	<input type="text" value="/var/www/html/admin/modules/shmzskin/views/menuitem"/>
View: noaccess.php <sup>?</sup>	<input type="text" value="/var/www/html/admin/modules/shmzskin/views/noaccess"/>
View: unauthorized.php <sup>?</sup>	<input type="text" value="views/unauthorized.php"/>
View: bad_refferer.php <sup>?</sup>	<input type="text" value="/var/www/html/admin/modules/shmzskin/views/bad_reffere"/>
View: loggedout.php <sup>?</sup>	<input type="text" value="views/loggedout.php"/>
View: panel.php <sup>?</sup>	<input type="text" value="views/panel.php"/>
View: reports.php <sup>?</sup>	<input type="text" value="views/reports.php"/>
View: menu.php <sup>?</sup>	<input type="text" value="modules/shmzskin/views/menu.php"/>
jQuery UI css <sup>?</sup>	<input type="text" value="modules/shmzskin/assets/css/jquery-ui.css"/>
View: login.php <sup>?</sup>	<input type="text" value="views/login.php"/>
View: header.php <sup>?</sup>	<input type="text" value="views/header.php"/>
View: freepbx.php <sup>?</sup>	<input type="text" value="views/footer.php"/>
Alternate JS <sup>?</sup>	<input type="text" value="modules/shmzskin/assets/js/custom.js"/>
View: footer_content.php <sup>?</sup>	<input type="text" value="modules/shmzskin/views/footer_content.php"/>

## System Apps (hidden settings)

- System Apps advanced settings allows for custom paths to certain apps.
  - **Path for cp**- The path to cp as auto-determined by the system. Overwrite as necessary. Default value is **/bin/cp**, field can be left blank. Internal name is WHICH\_cp.
  - **Path for grep**- The path to grep as auto-determined by the system. Overwrite as necessary. Default value is **/bin/grep**, field can be left blank. Internal name is WHICH\_grep.
  - **Path for gs**- The path to gs as auto-determined by the system. Overwrite as necessary.

# Advanced Settings

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Default value is **/usr/bin/gs**, field can be left blank. Internal name is WHICH\_gs.

- **Path for mkdir**- The path to mkdir as auto-determined by the system. Overwrite as necessary. Default value is **/bin/mkdir**, field can be left blank. Internal name is WHICH\_mkdir.
- **Path for mysqldump**- The path to mysqldump as auto-determined by the system. Overwrite as necessary. Default value is **/usr/bin/mysqldump**, field can be left blank. Internal name is WHICH\_mysqldump.
- **Path for pdfinfo**- The path to pdfinfo as auto-determined by the system. Overwrite as necessary. Default value is **/usr/bin/pdfinfo**, field can be left blank. Internal name is WHICH\_pdfinfo.
- **Path for prosodyctl**- The path to prosodyctl as auto-determined by the system. Overwrite as necessary. Default value is **/usr/bin/prosodyctl**, field can be left blank. Internal name is WHICH\_prosodyctl.
- **Path for ps**- The path to ps as auto-determined by the system. Overwrite as necessary. Default value is **/bin/ps**, field can be left blank. Internal name is WHICH\_ps.
- **Path for rm**- The path to rm as auto-determined by the system. Overwrite as necessary. Default value is **/bin/rm**, field can be left blank. Internal name is WHICH\_rm.
- **Path for tail**- The path to tail as auto-determined by the system. Overwrite as necessary. Default value is **/usr/bin/tail**, field can be left blank. Internal name is WHICH\_tail.
- **Path for tar**- The path to tar as auto-determined by the system. Overwrite as necessary. Default value is **/bin/tar**, field can be left blank. Internal name is WHICH\_tar.
- **Path for tiff2pdf**- The path to tiff2pdf as auto-determined by the system. Overwrite as necessary. Default value is **/usr/bin/tiff2pdf**, field can be left blank. Internal name is WHICH\_tiff2pdf.
- **Path for tiffcp**- The path to tiffcp as auto-determined by the system. Overwrite as necessary. Default value is **/usr/bin/tiffcp**, field can be left blank. Internal name is WHICH\_tiffcp.
- **Path for tiffinfo**- The path to tiffinfo as auto-determined by the system. Overwrite as necessary. Default value is **/usr/bin/tiffinfo**, field can be left blank. Internal name is WHICH\_tiffinfo.

### System Apps

Path for cp ?	/bin/cp
Path for grep ?	/bin/grep
Path for gs ?	/usr/bin/gs
Path for mkdir ?	/bin/mkdir
Path for mysqldump ?	/usr/bin/mysqldump
Path for pdfinfo ?	/usr/bin/pdfinfo
Path for prosodyctl ?	/usr/bin/prosodyctl
Path for ps ?	/bin/ps
Path for rm ?	/bin/rm
Path for tail ?	/usr/bin/tail
Path for tar ?	/bin/tar
Path for tiff2pdf ?	/usr/bin/tiff2pdf
Path for tiffcp ?	/usr/bin/tiffcp
Path for tiffinfo ?	/usr/bin/tiffinfo

### System Setup (hidden settings)

- System Setup advanced settings allow you to make customizations to match certain Asterisk installations.
  - **Authorization Type**- Authentication type to use for web admin. If the type is set to “database,” the primary AMP admin credentials will be the AMPDBUSER/AMPDBPASS as above. When using “database,” you can create users that are restricted to only certain module pages. When set to “none,” you should make sure you have provided security at the apache level. When set to “webserver,” the PBX will expect authentication to happen at the apache level, but will take the user credentials and apply any restrictions as if it were in “database” mode. Default value is **database** and the internal name is AUTHTYPE.
  - **Force Asterisk Version**- Normally the PBX gets the current Asterisk version directly from Asterisk. This is required to generate proper dialplan for a given version. When using some custom Asterisk builds, the version may not be properly parsed and improper dialplan may be generated. Setting this to an equivalent Asterisk version will override what is read from Asterisk. This SHOULD be left blank unless you know what you are doing. **No** default provided, field can be left blank. The internal name is FORCED\_AST\_VERSION.
  - **System Asterisk Group**- The user group Asterisk should be running as, used by freepbx\_engine. Most users should not change this. Default value is **asterisk** and the internal name is AMPASTERISKGROUP.
  - **System Asterisk User**- The user Asterisk should be running as, used by freepbx\_engine. Most users should not change this. Default value is **asterisk** and the internal name is AMPASTERISKUSER.
  - **System Device Group**- The user group that various device directories should be set to, used by the freepbx\_engine. Examples include: /dev/zap, /dev/dhadhi, /dev/misdn, /dev/mISDN and /dev/dsp. Most users should not change this. Default value is **asterisk** and the internal name is AMPDEVGROUPE.
  - **System Device User**- The user that various device directories should be set to, used by the freepbx\_engine. Examples include: /dev/zap, /dev/dhadhi, /dev/misdn, /dev/mISDN and /dev/dsp. Most users should not change this. Default value is **asterisk** and the internal name is AMPDEVUSER.
  - **System Web Group**- The user group httpd should be running as, used by freepbx\_engine. Most users should not change this. Default value is **asterisk** and the internal name is AMPASTERISKWEBGROUP.
  - **System Web User**- The user httpd should be running as, used by freepbx\_engine. Most users should not change this. Default value is **asterisk** and the internal name is AMPASTERISKWEBUSER.
  - **Telephony Engine**- The telephony backend engine being used. Asterisk is the only option currently. Default value is **asterisk** and the internal name is AMPENGINE.
  - **jQuery UI Version**- The version of jQuery UI that you wish to use. Default value is **1.8.9** and the internal name is JQUERYUI\_VER.
  - **jQuery Version**- The version of jQuery that you wish to use. Default value is **1.7.1** and the internal name is JQUERY\_VER.

# Advanced Settings

## User Guide

### System Setup

PBX Log Routing ?	FILE	
Disable PBX Log ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False	↻
Log Verbose Messages ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False	
Send Dashboard Notifications to Log ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False	
PBX Log File ?	/var/log/asterisk/freepbx.log	
PHP Error Log Output ?	debug	
User & Devices Mode ?	extensions	
Authorization Type ?	database	
Allow Login With DB Credentials ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False	
User Portal Admin Username ?		
User Portal Admin Password ?	ari_password	
Asterisk VMU Mask ?	007	
Browser Stats ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False	
Force Asterisk Version ?		
PBX Web Address ?		
System Asterisk Group ?	asterisk	
System Asterisk User ?	asterisk	
System Device Group ?	asterisk	
System Device User ?	asterisk	
System Web Group ?	asterisk	
System Web User ?	asterisk	
Telephony Engine ?	asterisk	
Use Google Distribution Network for js Downloads ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False	↻
Convert Music Files to WAV ?	<input checked="" type="checkbox"/> True <input type="checkbox"/> False	
Dashboard Non-Std SSH Port ?		
Recordings Crypt Key ?		
Use Old Parking Patch ?	<input type="checkbox"/> True <input checked="" type="checkbox"/> False	
jQuery UI Version ?	1.8.9	
jQuery Version ?	1.7.1	

## VmX Locator

- Used to do extremely advanced and customized changes to the macro-vm VmX locator.
  - **VMX Default Context**- Used to do extremely advanced and customized changes to the macro-vm VmX locator. Check the dialplan for a thorough understanding of how to use this. Default value is **from-internal** and the internal name is VMX\_CONTEXT.
  - **VMX Default Loop Exceed Context**- Used to do extremely advanced and customized changes to the macro-vm VmX locator. Check the dialplan for a thorough understanding of

how to use this. This is the default location that a caller will be sent if they press an invalid option too many times as defined by the maximum loop count. **No** default provided, field can be left blank. The internal name is VMX\_LOOPDEST\_CONTEXT.

- **VMX Default Loop Exceed Extension**- Used to do extremely advanced and customized changes to the macro-vm VmX locator. Check the dialplan for a thorough understanding of how to use this. This is the default location that a caller will be sent if they press an invalid option too many times as defined by the maximum loop count. Default value is **dovm** and the internal name is VMX\_LOOPDEST\_EXT.
- **VMX Default Loop Exceed Priority**- Used to do extremely advanced and customized changes to the macro-vm VmX locator. Check the dialplan for a thorough understanding of how to use this. This is the default location that a caller will be sent if they press an invalid option too many times as defined by the maximum loop count. Default value is **1** with acceptable values of **1-1000** and the internal name is VMX\_LOOPDEST\_PRI.
- **VMX Default Priority**- Used to do extremely advanced and customized changes to the macro-vm VmX locator. Check the dialplan for a thorough understanding of how to use this. Default value is **1** with acceptable values of **1-1000** and the internal name is VMX\_PRI.
- **VMX Default Timeout Context**- Used to do extremely advanced and customized changes to the macro-vm VmX locator. Check the dialplan for a thorough understanding of how to use this. This is the default location that a caller will be sent if they don't press any key (timeout) or press #, which is interpreted as a timeout. Default value is **dovm** (to voicemail) and the internal name is VMX\_TIMEDEST\_CONTEXT.
- **VMX Default Timeout Extension**- Used to do extremely advanced and customized changes to the macro-vm VmX locator. Check the dialplan for a thorough understanding of how to use this. This is the default location that a caller will be sent if they don't press any key (timeout) or press #, which is interpreted as a timeout. Default value is **dovm** (to voicemail) and the internal name is VMX\_TIMEDEST\_EXT.
- **VMX Default Timeout Priority**- Used to do extremely advanced and customized changes to the macro-vm VmX locator. Check the dialplan for a thorough understanding of how to use this. This is the default VmX timeout priority. Default value is **1** and has acceptable values of **1-1000**. The internal name is VMX\_TIMEDEST\_PRI.

### VmX Locator

VMX Default Context ?	from-internal
VMX Default Loop Exceed Context ?	
VMX Default Loop Exceed Extension ?	dovm
VMX Default Loop Exceed Priority ?	1
VMX Default Priority ?	1
VMX Default Timeout Context ?	
VMX Default Timeout Extension ?	dovm
VMX Default Timeout Priority ?	1



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