

AA50 1.3 Software F.A.Q. and What's New

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Revision History

Revision	Date	By	Notes
0.1	March 12, 2009	Malcolm Davenport	Initial Draft
0.2	March 18, 2009	Malcolm Davenport	Updates to features descriptions
0.3	March 19, 2008	Malcolm Davenport	Address reviewers' comments



Digium has recently announced the release of its 1.3 software for the AA50 Asterisk Appliance product. This document provides a Frequently Asked Questions and a “What’s New” guide to this release.

Frequently Asked Questions / F.A.Q

- Where Can I download the 1.3 software for my AA50?
 - Users who have registered their AA50 hardware and their subscription, as included with their AA50 purchase, at <http://www.digium.com/register> may visit the Asterisk Business Edition downloads portal at <https://be.digium.com>. Once logged in, you will see a "Downloads" link. Click it. Then, you'll see a tab called "AA50 Files." Click it. You will now see a 1.3 Folder. Go there. Inside of this folder, you will see a Changelog file that contains information about what's new in the release, several bitmap image files for Polycom phone logos, and the new firmware itself - a uImage file. You may download this uImage file to your local PC.
- Once I've downloaded the uImage file, how do I perform an update?
 - First, log into the web UI for your AA50. Once there, go to the "Updates" menu. There, you will see a form that allows you to upload to the AA50 the uImage file that you've downloaded to your local PC. Once you've uploaded the file, your AA50 will need to reboot. The reboot process when upgrading a firmware can take about 8 minutes, so go grab a cup of coffee. When you come back from your coffee, log into the web interface again and the upgrade process will be complete.
- Did you fix issue xyz or add new feature abc that I reported to your Support team?
 - Possibly, keep reading below for more information. We get a ton of feature requests from customers every day, and we get reports of bugs, too. We've done our best to address some of those requests and issues that we think will have the largest impact on the customer base. Your feature request and your issue report are important to us and we're always interested to hear your suggestions.

What's New?

G.729a Support

VoIP calls sounding poor when your bandwidth utilization is high? Don't have enough bandwidth to support all of the VoIP calls you need to your service provider? Perhaps you should look into compressing your VoIP traffic using a highly efficient codec like G.729a. Regular uncompressed VoIP calls require around 88kbit/s for a single call. When compressing your VoIP audio using G.729a, you can drop that requirement down to around 32kbit/s, a 60% savings.

Beginning with the 1.3 release of the AA50, Digium is pleased to be able to offer G.729a support. Customers who have enjoyed G.729a support in Asterisk and Digium's other products, like Asterisk Business Edition and Switchvox, now have access to this same codec on our embedded platform, the AA50.

Digium licenses G.729a on a per-channel basis. A channel is an in-progress bi-directional call. Two calls to your VoIP service provider using G.729a, that's two channels. Three callers using a conference bridge, that's three channels. Two people making one call to each other, that's only one channel.

Customers may purchase Digium's standard G.729a for Asterisk channel licenses from your Digium reseller or from Digium's online store. Users who enjoy the web-based interface of the AA50 over the command line interface of regular Asterisk will be pleased to see the new web-based registration system for G.729a.

The AA50 can handle roughly 8 concurrent G.729a station to station calls – somewhat less if you plan on putting those calls in a MeetMe or send them all to VoiceMail or an IVR. Customers may register multiple G.729a keys to a single AA50. Note that it is possible to register a key for more than 8 channels of G.729a to a single AA50, but that its use for higher numbers of concurrent calls is not supported and Digium will not split your keys to allow you to reuse any unused channel licenses.

More information about Digium's G.729a for Asterisk software is available here:

<http://www.digium.com/en/products/g729codec.php>

So, if you need more calls than you have bandwidth, try G.729a, now for the AA50.



Paging and Intercom

Ever found yourself at the front desk playing a bell boy in a movie about your life and needing to Page someone? Or, perhaps your significant other is moving out due to your misplaced affection for hockey, when you should have been focusing on golf?

The 1.3 software for the AA50 is in a position to help.

Now, using a friendly Paging/Intercom configuration page, you can dial individual extensions using custom defined paging or intercom prefixes. Define the paging prefix as ## and then simply dial ##6000 to page extension 6000, or ##6001 to page extension 6001. Define the intercom prefix as #* and then dial #*6001 to intercom with extension 6001. Need to page or intercom more than one phone at a time? You can also define, using a simple menu, groups of phones that can be added to paging or intercom groups that can be dialed with a dedicated extension number.

Find me and Follow me

The AA50 1.3 software release adds Find me/Follow me capability with an easy to use configuration page. Through this interface, you can enable or disable Find me/Follow me on a per User basis and define, when enabled, numbers that the AA50 will try to find a user. These numbers can be called sequentially or together in steps, and each step supports a timeout. Failing the AA50's ability to reach a user through one of the defined Follow me numbers, it will direct the caller to that user's voicemail.

Note that Asterisk does not support answer supervision across analog ports. Thus, any Follow me numbers that ring to an analog port, instead of a SIP trunk, will be considered answered.

Custom Feature Codes

With software release 1.3, the AA50 now provides support for a standard set of call feature codes, including blind transfer, attended transfer, call disconnect, and call parking. These call features allow you to control calls by entering DTMF sequences while the call is in progress. Enabling, for example, the attended transfer call feature, and setting it to *2, will allow you, while already on a call, to dial *2 to access a menu from which you can transfer to another

extension the person with which you are speaking. Call Features may be enabled on a per feature basis and may be set to be actionable by the calling party, the called party, or both.

In addition to these standard feature codes, an interface is provided for the creation of custom feature codes. The custom feature codes interface allows advanced users to define keypress combinations that result in the execution of a selected Asterisk application – from a list of available Asterisk CLI applications.

Polycom Phone Configuration

A number of improvements related to Polycom Phone configuration and provisioning have been added including:

- 1) The ability to upgrade the Polycom firmware via the web interface. Using the Polycom upgrade utility, the AA50 will connect to a download site provided by Digium and retrieve the current Polycom firmware (3.1.2 as of this writing). After retrieving the updated firmware, the AA50 will reboot to load it.
- 2) The ability to assign multiple line appearance keys to a single extension. Previously, extensions were assigned to one line appearance key. With this update, extensions can now be assigned a user configurable number of line appearances. Note that your Polycom phone supports only a limited number of line appearances, dependent upon your phone model.
- 3) Support for customizing your phone's digit mapping and digit timeout. Via this interface, you can now configure the digit map pushed to Polycom phones that are provisioned by the AA50. This will solve all of those pesky timeout problems waiting for phones to dial.

Bulk Adds

A new option available under the “Advanced Options” settings is the Bulk Adds utility. The Bulk Adds utility allows for either the input of a CSV list of extensions or for the creation of a specific number of users beginning with a starting extension number.

Other Stuff

In addition to the changes listed above, we've done a number of other things:

Provided an "Update Timezones" button so that our users upgrading from older versions have an easy way to get a complete timezones list without having to upload anything to the AA50.

Corrected an issue with displaying IAX2 trunks names status in the Status page, so that trunk names with lots of characters do not return as unregistered on the webpage when they are, in fact, registered.

Corrected an issue with the uploading of Backups using Internet Explorer IE6 and IE7 browsers.

Corrected an issue with inconsistent ring-back on extension dialing.

Added the ability to select WAV as a file type option for recording Voice Menu prompts.

Added Parked calls and Conference calls display to the System Status page.

Added, for Support purposes, the tcpdump utility to the command line interface.

Added the ability to see assigned DHCP leases.

Added SSMTP authentication options to allow sending mail via relays that require authentication.

Modified the Backups page to also allow backing up of users' stored voicemail and any custom-recorded prompts.

We thank you for your purchase and continued support for Digium!

