

Asterisk Appliance 50

Feature List

Software v1.0.3.3
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Introduction

The features contained within this document are available with the release of v1.0.3.3 in December 2007.

Acronyms

ACD.....	Automatic Call Distribution
AGI.....	Asterisk Gateway Interface
CDR.....	Call Detail Records
CTI.....	Computer Telephony Integration
DHCP.....	Dynamic Host Configuration Protocol
DNS.....	Dynamic Name Service
DTMF.....	Dual Tone Multiple Frequency (Touchtone)
HTTP.....	Hyper Text Transport Protocol
FTP.....	File Transfer Protocol
FXO.....	Foreign Exchange Office (connects to PSTN trunks)
FXS.....	Foreign Exchange Station (connects to phones)
GSM.....	Global System for Mobile communications
GUI.....	Graphical User Interface
IAX.....	Inter-Asterisk eXchange protocol
IE.....	Internet Explorer
IP.....	Internet Protocol
IVR.....	Interactive Voice Response (a.k.a. automated attendant)
LAN.....	Local Area Network
MAC.....	Media Access Control
MGCP.....	Media Gateway Control Protocol
MOH.....	Music On Hold
MTU.....	Maximum Transmission Unit
NAT.....	Network Address Translation
NTP.....	Network Time Protocol
OS.....	Operating System
PIN.....	Personal Identification Number
POTS.....	Plain Old Telephone Service
PSTN.....	Public Switched Telephone Network
SCCP.....	Skinnny Client Control Protocol (Cisco proprietary)
SIP.....	Session Initiation Protocol
SMTP.....	Simple Mail Transport Protocol
SSH.....	Secure SHell
UNIStim.....	Unified Network Internet protocol Stimulus protocol (Nortel proprietary)
URL.....	Uniform Resource Locator
VoIP.....	Voice over Internet Protocol
WAN.....	Wide Area Network

Configurations Available

- S800i – All VoIP (no analog ports)
- S844B – 4 FXO and 4 FXS ports
- S808B – 8 FXO ports

Compact Flash Storage

- Sound packs in several formats: G.711 a-law & μ -law, G.722, G.729, and GSM
- Polycom firmware (v2.2.0.0047) and configurations

Graphical User Interface (GUI) fully functional in the following browsers:

- Firefox
- Safari
- Internet Explorer (prior to IE7)

Users

A list of extensions is provided that includes all user extensions as well as conference bridges, voice menus for the IVR, call queues, ring groups, and the extension that's used to check voicemail. The system can support a maximum of 50 users that can be added, deleted, or edited at any time.

For each user extension, the following items can be specified during initial setup of the user or can be modified once the user has been created:

- Extension
- Name
- Password (for SIP and IAX)
- Voicemail password
- Email address (for voicemail delivery by email)
- Caller ID number (for presentation to other users)
- Analog phone port (if available and if this user has a POTS phone)
- Dial plan (set of calling rules that applies to this user)
- Polycom serial number used for auto provisioning

Additionally, several features can be enabled on a per user basis, including the following:

- Voicemail
- SIP
- IAX
- CTI
- Call waiting
- SIP Reinvite
- 3-way calling

- Put them in the system phone directory
- Make the user an agent/member of a call queue
- Enable the user's SIP account to work behind a NAT router
- DTMF Mode (advanced setting)
- Codecs allowed/disallowed (advanced setting)

It should also be noted that the “Users” GUI page provides a BuyNow button that provides a direct link to Digium web page.

Conference Bridges

The system can support a maximum of 20 conference bridges (G.711 non-transcoded SIP to SIP calls) that can be added, deleted, or edited at any time. Conference bridges allow multiple users or outside lines to participate in the same conversation. Each conference bridge is assigned an extension so that it can be directly dialed or be used in the voice menus (IVR).

Each conference bridge can be configured with an optional PIN code for user level access as well as an optional PIN code for administrator access. Several options can be set for each conference bridge including the following items:

- Playing music on hold for the first caller into the conference
- Providing a menu to the user that presses the '*' key while the conference
- Announcing new callers into the conference
- Quiet mode operation (not playing sounds when participants enter and leave the conference)

Voicemail

Voicemail is available for all users who have the “Voicemail” configuration box checked during the user setup. In the Voicemail setup, the extension that will be dialed for checking voicemail is configured. Configurable options for the voicemail system include the following:

- Enable the sending of emails with voicemail attachments
- Maximum voicemail greeting length in seconds (maximum of 3600 seconds)
- Ability for callers into the voicemail application to dial '0' to reach the operator extension (any single extension can be configured as the operator extension)
- Maximum number of messages per mailbox (choices are 10, 25, 100, 250, 500, and 1000)
- Maximum message time (choices are 1, 2, 5, 15, and 30 minutes as well as Unlimited)
- Minimum message time (choices are 1, 2, 3, 4, and 5 seconds as well as No minimum)
- Voicemail retrieval by email only
- Enable the announcement of the sender's caller ID
- Enable the announcement of the message duration

- Disable the announcement of information about the messages before playback
- Allow callers to review their message after leaving it

The compact flash storage system allows for an estimated 20 hours of voicemail to be stored in wav format. The amount of storage available depends on what is also stored on the compact flash (other than the default sound packs and Polycom firmware/configuration).

In order to facilitate the emailing of voicemail messages, options to configure the SMTP server address and port number are also provided.

Call Queues

The system can support a maximum of 10 call queues that can be added, deleted, or edited at any time. Call Queues provide ACD functionality inside of the Appliance.

In the Call Queues setup, the following options are available:

- An extension must be defined for the queue
- A name must be given to the queue
- A ringing strategy may be defined. Available options include:
 - RingAll – ring all available agents until one answers
 - RoundRobin – take turns ringing each available agent, always starting with the first agent
 - LeastRecent – ring the agent that was least recently called
 - FewestCalls – ring the agent with the fewest answered calls
 - Random – ring a random agent
 - RRmemory – take turns ringing each available agent, remembering which agent last answered the queue and starting with the next agent in the queue
- Assign any user extension as an agent/member of the queue
- Timeout – the amount of time a call is allowed to ring an individual agent before moving on to the next available agent (minimum of 0 second, maximum of 600 seconds)
- Wrap-up Time – the amount of time given an agent after completing a call before another call is sent to them (minimum of no delay, maximum of 600 seconds)
- Max Len – the maximum number of calls that may wait in the queue (valid range is from 0 [unlimited] to 20)
- Music On Hold – provided while callers are in the queue
- Auto Fill – an option to operate in parallel mode where calls are sent to as many as possible available agents simultaneously, rather than one call at a time
- Auto Pause – agents that do not answer calls will be marked as paused, and they will not receive additional calls until they again log into the queue.

- Join Empty – an option to allow callers to enter queues even though there are no agents logged in
- Leave When Empty – an option to force all callers out of the Queue if all agents log out
- Report Hold Time – an option to inform the agent of the duration of the caller’s hold time before the call is activated.

Service Providers / Trunks

The system can support a maximum of 10 service provider or trunk interfaces that can be added, deleted, or edited at any time. The Service Providers configuration provides analog and VoIP trunk setup capability for the Appliance. Three choices are available: Analog, VoIP, and Custom VoIP.

Analog allows the creation of trunks that are comprised of individual analog lines (not phones/stations) or groups of lines. Before configuring analog trunks, the physical phone lines should first be plugged into the Asterisk Appliance. At the end of the analog configuration process, the administrator will be prompted to run the Easy Calibration tool. This tool should be run during each installation to ensure that the Appliance is calibrated to the line impedance and gains necessary for proper operation. Further gain adjustment can be accomplished via the Audio Levels submenu, although this should not be necessary for most installations.

VoIP allows the choice of predefined certified Asterisk Appliance service provider partners. Several of the providers require entry of username and password for authentication. If an account is not already in hand, simply click on the provider logo to register for one. Other providers rely on IP-based authentication and do not require entry of username and password – but still require an active account with the provider.

Custom VoIP allows the creation of custom VoIP service providers. The acceptable options are:

- Comment – an alphanumeric name for the provider
- Protocol – the protocol (IAX or SIP) to be used for the provider
- Register – an option to force registration with the provider (useful for latent links and for IAX links behind NAT)
- Host – the hostname or IP address of the provider
- Username – the username used for authentication
- Password – the password used for authentication

Once the Service Provider trunks have been configured, they can be edited or deleted by selecting the appropriate option from the drop down “Options” menu. For Analog lines, they can be recalibrated or have their gain settings adjusted via the Audio Levels submenu. VoIP and Custom VoIP configurations can choose which codecs will be allowed/disallowed. For VoIP and Custom VoIP service providers, the following Advanced Settings can also be modified:

- trunkname
- insecure

- port
- Caller ID
- fromdomain
- fromuser
- contact

Calling Rules

The system can support a maximum of 20 Dial Plans that can be added, deleted, or edited at any time. Each Dial Plan can have a maximum of 50 Calling Rules that provide for the creation of pattern matches mapped to Service Providers.

The system provides the ability to choose an existing dial plan or to create new dial plans. Once a dial plan has been created, calling rules may be built, while existing rules can always be edited or deleted. By default, the Appliance comes with the following calling rules:

- Longdistance – Begins with 91 and followed by 10 or more digits
- IAXTEL – Begins with 91700 and followed by 7 or more digits
- Local – Begins with 9256 and followed by 7 or more digits
- International – Begins with 9011 and followed by 7 or more digits
- Local – Begins with 9 and followed by 7 or more digits
- 911 – Begins with 911 and followed by 0 or more digits

When editing a Calling Rule or creating a new Calling Rule, the following options are found:

- Rule Name (an alphanumeric name for the rule)
- Place this Call Through (a drop-down box from which a service provider for the calls falling under this rule may be chosen)
- Dialing Rules (in basic and advanced formats)
- Stripping (removing a number of digits from the end of a dial string)
- Prepending (adding a number of digits to the beginning of a dial string)

Inside of Basic Pattern, a user inputs the digits that begin the pattern to be matched, the number of digits that follow, and whether or not the following digits are a fixed length or variable. Clicking on the “define a custom pattern” text link provides further controls. A text-field is provided that allows for the exact dial pattern to be matched. “N” defines a digit from 2-9, “X” defines a digit from 0-9, and “.” defines any number of additional digits.

Incoming Calls

The system can support a maximum of 100 Incoming Call Rules that can be added, deleted, or edited at any time. Incoming Call Rules allow the routing of calls that are inbound to this system from other systems, such as a VoIP provider or the PSTN. The following options are available when adding or editing an

incoming rule:

- Route (choose either unmatched calls or calls that match a specific pattern)
- From Provider (choose the Service Provider from which the calls are coming)
- To Extension (choose the extension to which the calls should be routed)

Network Setup

Several different configurable Networking parameters exist for the AA50. These options fall into four different categories: general, WAN, LAN, and Timezones.

The general options include:

- Hostname – an alphanumeric hostname for the system)
- NTP server – the address of the nearest NTP server, e.g. pool.ntp.org)
- Enable SSH – checking this box enables the SSH daemon on the Appliance. Note: it will be enabled on both WAN and LAN interfaces, so this may pose a security risk
- URL used for Polycom auto provisioning – by default this points to <http://<ip of appliance>/phoneprov>

WAN options include:

- DHCP – checking this box will cause the Appliance to request its WAN IP address from your externally accessible DHCP server
- IP – if DHCP is not to be used, this box is to be filled with the static WAN IP address of the Appliance
- Subnet – if DHCP is not to be used, this box is to be filled with the WAN Subnet mask of the Appliance
- Gateway – if DHCP is not to be used, this box is to be filled with the WAN Default Gateway address for the Appliance
- DNS – if DHCP is not to be used, this box is to be filled with the IP address of the DNS server that the Appliance should contact for host name resolution
- Enable GUI on WAN Interface – checking this box turns on GUI access via the WAN port. Note that this is disabled by default due to the possible security risk.

LAN options include:

- Domain – this box allows the entry of an alphanumeric name for the domain if the LAN DHCP server is enabled.
- Enable DHCP Server – checking this box enables the Appliance's LAN-facing DHCP server
- IP – this box allows the entry of the IP address for the LAN-facing network interface of the Appliance
- Subnet – this box allows the entry of the Subnet mask to be used on the LAN-facing network interface of the Appliance
- DNS – this box allows the entry of the DNS address to be used on the

LAN side of the Appliance; setting this to the LAN IP address of the Appliance will allow pass-through resolution of DNS from the WAN side of the Appliance

- IP Start Range – the first IP address in the range of IP addresses to be served by the Appliance if the LAN side DHCP server is enabled
- IP End Range – the last IP address in the range of IP addresses to be served by the Appliance if the LAN side DHCP server is enabled
- Lease Time – the time, in seconds, for which IP address leases are to be valid if the LAN side DHCP server is enabled (valid range is 0 [no expiration] to 31536000)
- Max Leases – the maximum number of leases that the Appliance will hand out if its DHCP server is enabled (valid range is 0 to 100)

The Time zone option provides the ability for the user to upload the timezone file appropriate for their Appliance. The timezone files are included on the Appliance CD-ROM. Timezone files taken from Linux machines, i.e. /usr/share/zoneinfo, may also be used. This sets the local time zone of the Appliance.

Voice Menus

The system can support a maximum of 20 Voice Menus that can be added, deleted, or edited at any time. The Voice Menus option allows the user to build IVR menu trees. By default, the Appliance ships with a sample menu called “Welcome.” In building a menu, several options are used to configure each menu:

- Name – the alphanumeric name of the Menu
- Extension – the extension number dialed to enter the menu
- Steps – an ordered list of the actions to take place inside of the menu
- Up and Down – buttons that are used to reorder the actions
- Add a New Step – a drop-down box for selecting new steps to be added that include the following options:
 - Answer
 - Authenticate
 - Background
 - Busy Tone
 - Congestion
 - Digit Timeout
 - DISA
 - Response Timeout
 - Ringback
 - Wait
 - WaitExten
 - Goto Menu
 - Goto Directory
 - Goto Extension
 - Dial Ring Group
 - Hangup

- Dial Other Extensions – a checkbox that when checked allows the user to dial extensions other than those inside of the menu
- KeyPress Events – a window for creating actions based upon key presses received from the caller. Options for key presses include:
 - Disabled
 - Goto Menu
 - Goto Extension
 - Custom
 - Hangup
 - Play Invalid

Record a Menu

The Record a Menu option provides the capability of recording new prompts directly into the Appliance. When recording a new menu, the user is prompted to enter a filename and to input the extension used for recording. The extension that is input will then be dialed at which point the caller will be prompted to speak the prompt and to save it. When completed, the prompt will then be available for playback as a part of the selection of prompts in the Voice Menus option.

Call Parking

The system can support a maximum of 10 Call Park zones. The Call Parking option provides configuration of the call parking features of the Appliance. By default, Call Parking is mapped to extension 700. When a user is on a call and wishes to park that call by placing it in a holding lot that is not directly tied to the users phone, the user executes an assisted transfer into the call parking extension. The Appliance will read back the lot into which the caller was parked, as configured in the “What extensions to park calls on” option, and will leave the call there for the number of seconds specified in the option box in the GUI. To retrieve the call, simply use another phone attached to the Appliance to dial the extension that Asterisk parked the call onto. At the end of the timeout period, if the call is not otherwise handled, the call will drop out of the parking lot and ring back to the phone that originally parked it.

Ring Groups

The system can support a maximum of 20 Ring Groups. Ring Groups are logical groups of devices that can be rung collectively. Within the concept of ring groups, there are many options that are available:

- Name – the alphanumeric name for the ring group
- Strategy – the ringing strategy, either Ring All or Ring in Order
- Ring Group Members and Available Channels – boxes with moving arrows that allow adding devices into the Ring Group from the available listing of devices
- Extension for this Ring Group – the alphanumeric extension used to ring the devices in the group
- Number of seconds to ring the members

- An option in the event that none of the phones answers – send the call to the voicemail box of the selected user, send the call to an IVR extension, or hang up the call.

Active Channels

An active channels tab is available to give visibility into which channels are currently in use. Any active channel can be disconnected or transferred to another extension using this interface.

System Information

Users can gain access to system information that falls into one of the following categories: general, ifconfig, and resources.

General system information that is available to the system administrator includes the following:

- OS version
- System uptime
- Asterisk software version
- Date and time of the unit
- Hostname

Interface information that is available for each Ethernet (WAN, LAN, and loopback) includes:

- MAC address
- IP address
- Subnet broadcast address
- Subnet mask
- MTU
- Receive packets
- Receive bytes
- Receive errors
- Receive dropped packets
- Receive overruns
- Receive frame errors
- Transmit packets
- Transmit bytes
- Transmit errors
- Transmit dropped packets
- Transmit overruns
- Transmit carrier errors
- Collisions
- Transmit queue length
- Loopback

Information about the following system resources is also available:

- Disk usage (Internal and Compact Flash)
 - File system name
 - Total number of 1KB blocks
 - Total number of 1KB blocks in use
 - Total number of 1KB blocks available
 - Percentage in use
 - Mount point
- Memory usage
 - Total bytes
 - Used bytes
 - Free bytes
 - Shared bytes
 - Buffers

Configuration Backup

The Configuration Backup allows a user to save and restore backups that are stored on the Compact Flash. Clicking the “Take a Backup” option allows the user to enter the filename to be used when saving the backup. Once a backup has been taken/saved, a user may elect to restore from the available backups or delete any of the backups.

System Update

Updated firmware can be downloaded from the Digium portal after registration, provided that the software warranty period for the user’s Appliance is still in effect. It can then be uploaded to the unit from the GUI or placed on an HTTP or FTP site where the unit can pull it from. Further instructions on this are provided in a specialized PowerPoint, Asterisk Appliance 50: Software Updates and System Recovery.

Options

The Options tab provides other administrative options.

The Local Extension Settings provides some basic configuration items for the system:

- Local Extensions Are – provides a dropdown box to select the number of digits that are allowed for local extensions. Options include 2, 3, 4, and 5 digits as well as varied length extensions.
- First Extension Number – sets the extension number of the first numeric extension
- Operator Extension – defines which user extension is assigned to be the Operator Extension
- Allow Analog Phone to be assigned multiple extensions – provides the ability for an analog phone to ring with multiple extension numbers.
- Allow Extensions to be Alphanumeric – in the event that the phone used

for dialing might be IP hard phones, IP softphones, or other phones that allow the input of characters in addition to numeric digits, this functionality can allow the creation of extensions that are not purely digit based.

- Agent Login Extension – defines the extension Agents dial to log themselves into a queue.
- Agent Callback Login – defines the extension that Agents dial in order to do a callback login, rather than an on-hold login
- Agent Logout – Agents using Agent Login may simply hang up. Agents using Callback Login should input # as their callback number.
- Default Settings for a New User – the system provides a method for defining those settings that are enabled by default for new users that are created, including the following:
 - Is Agent
 - Voicemail
 - In Directory
 - CTI
 - SIP
 - IAX
 - Call Waiting
 - 3-way Calling
 - Default Voicemail Password

Change Password – allows the user to change the GUI login password.

Advanced – allows the Administrator to access other options not found in the basic GUI:

- File Editor – provides the ability to view and edit the suite of Asterisk configuration files
- Asterisk CLI – provides an Asterisk command line interface console
- VM Email Settings – provides the ability to modify the Template for Voicemail Emails
- Global SIP Settings – provides detailed configuration parameters associated with SIP and VoIP
- Global IAX Settings – provides detailed configuration parameters for IAX trunks.

Setup Wizard – allows the Setup Wizard to be run. Note that the Setup Wizard is currently disabled by default.

Basic Polycom Auto Provisioning

For each user, a Polycom phone MAC address can be specified in the serial field of the user configuration in the Users tab of the GUI. If a Polycom phone with that MAC address is seen by the Appliance, the AA50 will provide it with firmware (v2.2.0.0047) and do basic provisioning, including the extension assignment to the phone. Other features, including automatically assigning a user

selected extension, one other than 6050, to the “Messages” button are planned for future revisions.

Configuration Reset Switch

The configuration reset switch, labeled CFG RST and located on the back of the unit, can be pressed during bootup to factory default the unit. The switch should remain depressed for 10 seconds after power is applied in order to be effective.

Supported Codecs

The following codecs are supported for transcoding:

- G.711 u-law
- G.711 a-law
- G.726 (32kbit/s)
- GSM
- iLBC

Unsupported Asterisk Features

Although the Asterisk Appliance 50 supports many Asterisk features, it will inherently not support every feature that could otherwise be accomplished with Asterisk. A number of capabilities that are not supported or are otherwise restricted include:

- Call Detail Records – the Appliance does not presently generate CDR records.
- Call Recording – the Appliance does not support call recording
- AGI – AGI scripts are not supported.
- H.323 – the Appliance does not support H.323 VoIP connections
- SCCP – the Appliance does not support SCCP (“Skinny”) VoIP connections
- MGCP – the Appliance does not support MGCP VoIP connections
- UNISTim – the Appliance does not support UNISTim VoIP connections
- G.729a – the Appliance does not presently support G.729a.
- SLA – the Appliance does not presently support Shared Line Appearances
- Paging – the Appliance does not presently support Paging.

Appendix A – Table of min/max values

The following table provides a brief summary of the information contained with this feature document for the Asterisk Appliance 50.

Feature	Minimum Value	Maximum Value	Default Value
Number of users	1	50	
Conference bridges	0	20	
VM greeting length (seconds)	1	3600	
Max messages per mailbox	10	1000	
Max message time (minutes)	1	unlimited	
Min message time (seconds)	no minimum	5	
Total Voicemail (hours)		20	
Number of call queues	0	10	
Call queue timeout (seconds)	0	600	
Call queue wrapup time (seconds)	0	600	
Call queue max length (# callers)	0 (unlimited)	20	
Service Provider interfaces	0	10	
Dial Plans	1	20	
Calling Rules per Dial Plan	1	50	
Incoming Call Rules	0	100	
DHCP Lease Time (seconds)	0 (never expire)	31536000	
DHCP Max Leases	0	100	
Voice Menus	0	20	
Call Park Zones	0	10	
Ring Groups	0	20	