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Release Notes NetVanta 7000 Series Products

AOS Release A4.03.00.SB
January 7, 2011

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Purpose

AOS Release A4.03.00.SB provides new features, enhances existing features and addresses field issues.

AOS A4.03.00.SB is considered a Generally Available (GA) release. This AOS is recommended only for new installations and existing installations that were waiting for solutions included in this release. Prior to the GA release this AOS has been subjected to both Design Verification and Product Qualification testing as well as completed field beta testing in both supervised and unsupervised capacity. Results obtained during this testing have been evaluated and the code has been determined to be ready for general availability. Issues discovered during testing but not addressed in this build are listed as Errata in [Appendix A](#).

Stage	Status	Date
Internal and Field Evaluation	Completed	12/16/2010
Controlled Release	Completed	12/27/201
Generally Available Release	Completed	1/7/2011

A listing of available documents for this release appears in [Appendix B](#). Further configuration guides, white papers, data sheets, and other documentation may be found in ADTRAN's Knowledgebase, <http://kb.adtran.com>.

Important Notices

NetVanta 7100 Hardware Considerations

New features coming with any AOS release warrant some attention before use by the customers, specifically the choice of the hardware platform on which the new AOS will be installed

There have been 2 revisions of NetVanta 7100 hardware in the field. These are denoted by different part numbers: 1200796L1 (old) and 1200796E1 (new). Starting with AOS release A2.04, we don't recommend AOS use on the 1200796L1 (old) units. These units continue to be field worthy, and would continue to perform as expected for their useful life on previous software revisions. However due to differences in hardware, some or all of the new features may not be supported on the older hardware (1200796L1).

The 1200796L1 is explicitly not recommended for use for the following features

- Support for more than 50 users (DSP resources were increased on 1200796E1 units, allowing additional TDM to IP conversions). The user limit on the 1200796L1 remains unchanged.
- SIP trunks that require the NV7100 to do transcoding. (This conversion is required, if the SIP trunk provider does not support G.729)
- Use of the Echo Return Loss (ERL) tool

While there aren't any other known constraints for the other features at this time, please keep yourself updated on any future advisory by ADTRAN. The recommended hardware for the AOS A2.05 and later features is 1200796E1. Feel free to discuss with your ADTRAN representative on what options you have in case you have an L1 unit, and you still want to use AOS A2.05 features

Recommended AOS Image location(s)

AOS images can be stored on FLASH/NONVOL as well as on CFLASH. However, it is recommended that the Primary AOS image be stored on FLASH/NONVOL and the backup image be stored on CFLASH. One reason for this is that starting with AOS A2.01.00, there will no longer be enough space on FLASH/NONVOL to store 2 versions of AOS firmware. To copy the current image from FLASH/NONVOL to CFLASH, use the command "copy flash <filename> cflash <filename>".

Required Bootcode Version

When upgrading to AOS A4.03.00.SB, an upgrade to Bootcode version A2.06.B1.01 is required. Contact ADTRAN Technical Support for this Bootcode version and instructions for loading it.

Notice of Defined Voicemail File Limit

Starting with AOS A2.04.00.SC, the NetVanta 7000 Series products can maintain a maximum of 3000 voicemails per system. The implementation of voicemail message expiration will allow the system to remain within the defined limit. Upgrading the CFLASH card to a larger card will not result in more voicemail storage and is therefore not supported. Should you have a need to replace a failed CFLASH card, please contact ADTRAN Technical Support for assistance.

Required Phone Firmware

For this AOS Version, the following versions of phone firmware are required to address issues found in the field and to support new features added with this AOS release. For instructions on upgrading to these phone firmware versions, see the [Upgrading Instructions](#) section of this document.

- ADTRAN IP 700 Series Phones – Version 1.3.14
- Polycom IP 321/331 Phones – Version 3.1.3C and version 4.1.2b bootrom
 - This version is available as a supplemental download and not necessary if IP 321/331 phones are not installed.
- Other ADTRAN approved Polycom Phones – Version 3.1.3b application and version 4.1.2b bootrom

These files can be downloaded by going to <http://www.adtran.com/support>, selecting Software Downloads, and choosing the appropriate Phone Model under IP 700 Series. Contact ADTRAN Post Sales Technical Support if you are unable to download these files.

AOS Versions for SIP Trunking and Networking Applications

For NetVanta 6355 and Total Access 900 Series products used in SIP Trunking applications involving the NetVanta 7000 Series, AOS A2.06.00 and later for these products is required for proper operation. For SIP Networking between NetVanta 7000 Series products, the same AOS version should be used in each product with A2.06.00 being the minimum recommended version.

Default Configuration Changes

- Applicable to all versions since AOS A4.01.00:
 - Forward Disconnect delay set to 750ms for Analog User Accounts
 - Voicemail authentication for new User Accounts is "password only". Existing User Accounts will not be affected by this change.
 - SIP Keepalive enabled by default for all new User Accounts. A SIP INFO message will be sent to the phone every 60 seconds to verify the call is still active. If no response is received from the phone, the call will tear down.

Default Firewall Configuration Changes

NOTE: This change only impacts remote phones and SIP trunking applications. Changes to default behavior do not impact local phones on the NV7000 series. ADTRAN has made this change to increase security of voice platforms when connected to the Internet.

- In AOS versions A2.01.00 through A2.03.00.SC, the default Public policy class allowed SIP traffic (destined for UDP port 5060) inbound.
- For AOS A2.04.00.SC and above, this traffic is no longer allowed by the factory default configuration. Instead the installer is required to selectively customize the Public policy class to allow SIP traffic from remote sites and SIP Trunking providers.
- Units that were shipped with AOS Versions up through A2.03.00.SC will contain a default configuration that allows this SIP traffic and these configurations should be modified before deployment. Guidelines for this configuration are given in the ***NetVanta 7000 Series Security Guide*** located in our Tech Support Knowledgebase here:
<http://kb.adtran.com/article.asp?article=3399&p=2>

Summary of New Features

AOS A4.03.00.SB does not include any additional features. This section highlights the major features, commands and behavioral changes for AOS A4.03.00 and previous A4 versions.

Features Included in AOS A4.03.00

Support for FindMe-FollowMe in Personal Phone Manager

- Adds the ability to configure Call Coverage to FindMe-FollowMe per System Mode.
- Adds the ability to define Contact Groups and Call Actions from the Personal Phone Manager.
- Note: the FindMe-FollowMe feature must be enabled for the User Account by the phone system admin.
- See link for configuration guide in [Appendix B](#).

Ability to backup voicemail greetings and mailbox names with n-Command MSP

- Adds the ability through Auto-Link to notify MSP to backup the mailbox name, default and standard greetings.
- Backup of the alternate greeting will be supported in a future release.

E911 Enhancements

- Added the ability to configure User Local Emergency Response Location (ERL).
- Added the ability to translate emergency Email Alerts into the configured system language.
- See link for configuration guide in [Appendix B](#).

Support for E1/PRA VIM

- Support for the new E1/PRA Voice Interface Module was added.
- Calls using the G.729 codec are limited to 25 for E1 PRI calls.

Support for ISDN Redirecting Number

- Added ability to enable the transmission of redirecting number on a PRI interface.
- Calls that are received on an ISDN trunk and forwarded or referred out an ISDN trunk will include the Redirecting Number in the call setup messaging and original Caller ID will be preserved.

[UC]: T.38 Support over FXO interfaces

- Added ability to initiate T.38 negotiation to UC Server for calls received on analog FXO trunks.
- This feature is also supported in the NetVanta 6355.

Features Included in AOS A4.02.00

Multi-lingual Support for Voicemail and Dial-By-Name Directories

- Supports a caller-selected language via Digit Action in the system Auto Attendant, which includes the Dial-By-Name directory.
- Supports Voicemail system prompts that will change based on the configuration of Voicemail for that User Account or Ring Group.
- Customers still maintain the ability to be record Auto Attendant greetings in any language desired. As before they can be recorded as multiple languages within the same greeting. The difference is in having a way to nest your Auto Attendants to have the caller pick up their language of choice once and ensure they get messages played back to them in that same language for the entire duration of the call.

Features Included in AOS A4.01.00

FindMe-FollowMe

- FindMe-FollowMe is a productivity feature for phone users who may not want to miss a call. It allows a user of the system to be reachable by phone all the time, primarily based on a combination of who is calling and the System Scheduler mode.
- FindMe-FollowMe allows calls to be directed based on who is calling. In a typical scenario, an incoming call is routed to FindMe-FollowMe via Call Coverage on a User account. The call is first sent to the user's phone and if the user does not answer, the call is sent to the user's cell phone. If configured, the caller is required to record their name. If the called party answers, they have the option to accept or reject the call. If the called party does not answer, or rejects the call, the call is then sent to the user's voicemail. In AOS, up to four simultaneous calls can be generated for the caller to find the called party.
- Configuring FindMe-FollowMe revolves around two main actions: configuring the user's Contact Groups, and configuring the actions, which determine the numbers to dial in the FindMe-FollowMe call sequence. Contact Groups are the groups of callers from whom the user anticipates receiving calls and would like to give a preconfigured treatment. For example, a user might have a separate Contact Group for family members and one for business partners. By default, AOS creates a **Priority** group, a **Blacklist** group, and an **Everyone** group. New groups can be added or groups can be removed at any time, and each user can have up to 5 Contact Groups. In addition, Contact Groups can be configured to play a courtesy greeting, prompt the caller for their name, and provide ringback to the inbound caller.
- The second part of configuring FindMe-FollowMe is to specify the actions taken when an incoming call is received. The configurable actions include:
 - Calling an internal extension
 - Calling an external number
 - Referring the call to another number (new call placed and ends actions)
 - Forwarding the call to voicemail
 - Sending a notification to the called party's primary or secondary email address.
- FindMe-FollowMe actions do not, however, include calling ring groups. Each action that places a call can be configured for a specific timeout in which to stop the call as well as prompt the called party for DTMF digits to manage the call. Each contact group can have up to **10** actions, and each action can place up to **4** simultaneous calls.
- Since FindMe-FollowMe can be resource intensive (processor utilization and consuming multiple trunks), enabling the feature is an administrative task, and should be done after careful resource planning and user training. There are two modes for FindMe-FollowMe in the NetVanta 7000 Series: Basic and Enhanced. Enhanced FindMe-FollowMe allows the AOS device to remain in the talk path for external calls and to accept Dual-Tone Multi-Frequency (DTMF) tones for more user call control. Enhanced mode of FindMe-FollowMe requires the most resources because of the NV7000 must stay in the talk path to listen for DTMF digits. The system limits the number of users that can enable FindMe-FollowMe (Basic or Enhanced) to **10**.
- Enhanced FindMe-FollowMe can collect digits from the connected call which initiates a transfer. Calls connected via Enhanced FindMe-FollowMe may be transferred by two methods - *7 to quickly transfer the call to your desk extension, or *88<NUMBER> to transfer the call to any number. If the user dials *7, their desk extension will ring and the call will only be transferred if the desk extension is answered. The application for quickly transferring to your desk extension (*7) would be answering a FindMe-FollowMe call while returning from

lunch and then sending the call to your desk phone upon returning to the office. When transferring a call to any number (*88<NUMBER>), the number dialed must be valid or the call is disconnected. The number may be an internal extension or external number and routes based on existing Dial Plan entries. The application for transferring to any number would include receiving a call but the caller needs to speak to someone else within the company.

- See the updated **User Accounts Configuration Guide** (referenced in [Appendix B](#)) for details on configuring FindMe-FollowMe.

Call Queuing

- Call Queuing pools multiple incoming calls into a single call queue that can be answered by members of the queue. The system supports up to 5 separate queues and a maximum of 16 callers queued in the system.
- Call Queuing can be configured to play Music on Hold and unique user-customized messages that inform callers of the status of their call. Each queue can have its own MoH playlist.
- Calls can be delivered automatically using a Linear or a Ring All Call Distribution. Local SIP users (such as a supervisor) can also call the Dequeue Extension to retrieve the next waiting call.
- A configurable Maximum Queued Calls (from 1 to 16) determines whether calls will roll to the configurable Overflow Extension. Maximum Hold Time (from 30 to 6400 seconds) can also determine when callers will be directed to the Overflow Extension.
- The following Greeting types can be configured for callers in the queue and default recorded greetings are included for each Greeting type.
 - Welcome
 - Periodic
 - Pickup
- These greetings can be recorded under Voice->Applications->Audio Prompts or custom prompts can be uploaded.
- Built in reporting including 24 hour Call Queue statistics and a real time dashboard are supported.
- Callers can opt out of holding in the queue and be transferred to an operator or voicemail as configured.
- Rest Time can be configured for per-queue for Call Queue members to allow for proper time between handling of calls.
- Call Queue members can also log-in and log-off via BLF and SPRE codes.
- Call Queues can be monitored via Status Groups/BLF. Calls may also be de-queued by pressing the same BLF key used to monitor the queue (SIP Phones only).
- See the **Call Queuing Configuration Guide** (referenced in [Appendix B](#)) for details on configuring Call Queuing.

Internal Music on Hold

- In addition to the external Music on Hold (MOH) jack on the back of the NetVanta 7000 Series products, MoH can now be sourced internally from audio files stored on the filesystem.
- While external or internal MoH can be used for music while normal calls are on-hold, certain features such as Call Queuing and FindMe-FollowMe only make use of internal MoH.
- Six different genres of music files are available for download from Software Downloads at <http://www.adtran.com> for units upgrading from previous AOS versions.
- Player Lists can be defined to play different songs for different features. At least one music file should always be defined in the default System playlist.
- Music can be sourced from the default files or can be from custom sources. Conversion of customer-owned audio can be facilitated using a conversion tool that is available under Software Downloads at <http://www.adtran.com>.
- See the **Music on Hold Configuration Guide** (referenced in [Appendix B](#)) for details on configuring FindMe-FollowMe.

NOTE: Please research the laws governing MoH usage, as some music is not royalty free and may not be legal source for MoH. ADTRAN cannot recommend or be held liable for improper usage of the software MoH feature.

Ring Group Call Waiting

- Previously, Ring Group members could only receive one Ring Group call at a time. Subsequent Ring Group calls would only be delivered to other members. Now User Accounts can be configured to allow multiple Ring Group calls to the same member. This is only currently only configurable in the CLI with the command "group-ring-call-waiting" on the User Account of the Ring Group member.

Ring Group DSP resource assignment

- In previous AOS versions, a DSP resource was required for each Ring Group member in order to deliver a call to that member on calls from a TDM trunk (FXO/PRI/CAS T1). Now, only 1 resource is assigned when delivering a Ring Group call. This allows for Ring Group calls to be delivered to more members than before without dependency on available DSP resources.

Per-user Caller ID Name override on external calls

- Adds the ability to configure a Caller ID Name to be used for external calls on a per-User Account basis. Note that Caller ID Name and Number substitution is subject to Trunk Account configuration and then to any overrides the Service Provider may apply on calls through the trunk.

Per-user Caller ID override on emergency calls

- Adds the ability to configure a Caller ID number to be used for emergency calls on a per-User Account basis. Note that Caller ID Name and Number substitution is subject first to Trunk Account configuration and then to any overrides the Service Provider may apply on calls from the trunk. This feature can be used to meet the unique Caller-ID requirements for E911 support.

Added ability to configure Ring Group Prefix from web interface

- Ring Group Prefix is a 40-character string that can be appended to the Calling Party Number when receiving a Ring Group call. This feature can now be configured in the web interface under Voice->Ring Groups->Edit Ring Group.

Added remote and graphical method to delete Voicemail messages from the system using the Personal Phone Manager Web interface

- On the Voicemail tab of the Personal Phone Manager, you could view the Voicemail messages for your mailbox. Now you can also delete those messages from that page.

Added support for additional Polycom phone models

- Added support for the Polycom IP 560, and IP 670.
- These phone models should be available for selection in IP Phone Configs.

Summary of Enhancements

There are no new enhancements included in AOS A4.03.00.SB. Enhancements Included in AOS A4.02.00 are listed below.

Ability to source Auto-Link from different interfaces

- Auto-Link traffic is sourced from a single interface and that specific interface is not directly configurable. However, configuring “ip http source-interface” will change the source interface for Auto-Link packets as well since it uses HTTP by default.

Summary of Bug Fixes

This section highlights major bug fixes in AOS version A4.03.00.SB and previous A4 versions.

Issues Resolved in AOS A4.03.00.SB

Reboot due to memory corruption

Issue Detail

- Unit may reboot when SIP re-invites are frequently being transmitted on a SIP trunk.
- This scenario is most likely to present itself where multiple NV7000 products are networked together with SIP Trunks.

Reboot during call clearing

Issue Detail

- If a BYE is received prior to a ringback re-INVITE response, a reboot may occur.

Issues Resolved in AOS A4.03.00.SA

No ring-back on calls hair-pinned through PRI trunk

Issue Detail

- Calls that were received on a PRI interface and redirected back out the same PRI interface (such as when an extension is forwarded), resulted in no ring-back to the caller.

Possible reboot with cancelled call

Issue Detail

- In rare circumstances, an inbound call to a ring-group could have caused a reboot if the receiving phone answered and the inbound caller disconnected at exactly the right time.

Reboot when receiving ISDN call with null Caller ID

Issue Detail

- When a call was received on an ISDN PRI trunk, if the Calling Party Number was null, the unit would reboot.

Incorrect formatting on some SIP NOTIFY messages

Issue Detail

- Some NOTIFYs for Status Group updates were being sent with additional text after the contents of the update.

Parked call drops on Park Return

Issue Detail

- When a parked call returns to the phone that parked it (after 60 seconds by default), the call will drop when the phone attempts to answer it.

Issues Resolved in AOS A4.03.00

Calls transferred from an Auto Attendant to a Ring Group may fail

Issue Detail

- If a call was received from a SIP trunk into an Auto Attendant and was transferred to a Ring Group, the call would drop when the Ring Group member answers.

Can't forward calls from Auto Attendant or Ring Group to Call Queue

Issue Detail

- With Ringback Only enabled, calls forwarded from AA or Ring Groups to a Call Queue would fail.

Can't modify ISDN Number Templates directly

Issue Detail

- ISDN Number Templates could not be modified directly and must be removed and replaced instead.
- ISDN Number Template 0 could not be modified or removed.

Call cannot be retrieved from hold after more than 2 minutes (Metaswitch)

Issue Detail

- If a call is received from a SIP trunk from a Metaswitch and placed on-hold, the call cannot be retrieved if the call has been on-hold for more than 2 minutes.
- This occurs only with a Polycom phone.

Call delivery time longer than configured for analog users

Issue Detail

- Call Coverage rules for analog FXS voice users would take approximately twice the configured number of rings to trigger.

Internal Music on Hold stops playing after receiving re-INVITE

Issue Detail

- Music on Hold will stop playing after a re-INVITE is received from a SIP trunk call placed on hold by a local SIP user on a NV7100. The NV7100 responds to the re-INVITE with a different connection port and/or IP address in the SDP from the original music on hold source. The mismatch of the ports and/or IP addresses in the SDP will cause the softswitch to generate a re-INVITE to the far-end directing its media source away from the music on hold server. This is more readily seen on a Metaswitch since it sends re-INVITES every 30 seconds by default.

Call placed on hold by an analog phone disconnects upon receiving re-INVITE

Issue Detail

- An inbound SIP trunk call to an analog user on a NV7100 will disconnect upon receiving a re-INVITE from the softswitch. When the NV7100 receives the re-INVITE, the call is dropped. This is more readily seen on a Metaswitch since it sends re-INVITES every 30 seconds by default.

Calls retrieved via Call Pickup/Groups and Call Queues sent to speakerphone

Issue Detail

- On IP 700 Series phones, when a user retrieves a call via Call Pickup, Pickup Groups, or Call Queues the audio for the retrieved call would be presented on the Speakerphone of the IP 700 Series phone.
- Pressing the Speakerphone button would put the call back on the user's handset and pressing the Headset button would put the call back on the user's headset.

Attended transfer to FMFM or ring group user is unsuccessful

Issue Detail

- An attended transfer would fail if attempting to transfer a call to a FindMe-FollowMe user or a Ring Group. A call is placed to a standard SIP user on a NV7100. The user attempts a transfer and dials a user configured for FindMe-FollowMe (or a Ring Group). The user answers the call and the transferor and the transfer-target (FindMe-FollowMe or Ring Group) talk while the transferee remains on hold. The transferor then presses the transfer button again to complete the attended transfer, but the transfer fails. The transferee will either receive a fast busy or no talk path will be established.

Can't remove Status Group entry from the CLI

Issue Detail

- System mode entries in a status-group could not be removed on the CLI.

[Web]: Call Quality Stats missing from web interface

Issue Detail

- The web interface equivalent of the CLI command "show voice quality-stats" was missing from the web interface.
- It is now located under Voice->Reports->Call Quality Stats.

[Web]: Ethernet 0/0 interface missing from Physical Interfaces page

Issue Detail

- Under System->Physical Interfaces, Ethernet 0/0 is not accessible. If an IP address is configured on the interface, it is accessible from Data->IP Interfaces. Otherwise, it can still be configured from the Command Line Interface (CLI).

[Web]: CSV import incorrect for names with spaces

Issue Detail

- If a First Name contained a space such as "Station 1" and no Last Name, the CSV import would create a User Account and Phone Config with a first name of "Station" and last name of "1".

[Web]: Override Number missing from PRI interface config with Internet Explorer

Issue Detail

- When using Internet Explorer, the configuration options for Override Number are not visible under the PRI Interface. They are visible if Firefox is used and they are accessible via the CLI.

[Web]: Creating new user from existing user doesn't copy VM schedule

Issue Detail

- When creating a new User Account based on an existing one, the existing Voicemail notification schedule does not get copied.

[Web]: Problems updating analog users

Issue Detail

- Existing analog users could not be updated from the GUI when all FXS ports have been assigned. The existing FXS port assigned to the user would appear to be blank, and the page would not allow you apply any changes.
- This would not occur when there are unassigned FXS ports.
- This could be avoided by configuring the user from the CLI instead of the web interface.

Issues Resolved in AOS A4.02.00

Reboot when PRI Called Party Number is null

Issue Detail

- When a call is received on a PRI and the Called Party information element is present but empty, a reboot could occur.

Performing Update Directories on an IP 7XX phone give an error

Issue Detail

- If "Update Directories" was performed on an IP 7xx phone, an error would be given. This was because the IP 7xx phones do not have personal directories, and the GUI was trying to update a non-existent file.

***67 for restricting CID not preserved on calls forwarded externally**

Issue Detail

- When performing *67 to block caller ID on a call to another extension, if the called party had their phone forwarded to an external number, the caller ID block would be lost.

Possible reboot related to Music on Hold

Issue Detail

- If Internal Music on Hold doesn't have a SDP answer when it tries to start playing, a reboot could occur.

Reboot due to files missing from MoH playlist

Issue Detail

- If internal MOH is configured and there are no files in the playlist, or files in the playlist were missing from the filesystem, a reboot could occur.

Analog phone Call Queue members do not hear greeting on pick-up

Issue Detail

- When an analog phone that is part of a Call Queue answers a call, the initial pick-up greeting is not heard but audio is established between the caller and the analog queue member.

Analog phone Call Queue members Rest Time incorrectly calculated

Issue Detail

- Rest Time for analog Call Queue members is calculated from the point where the analog port is ringing rather than from the point when it is off-hook. This results in unexpected Rest Time values and possible delays.
- This is only an issue with analog CQ members and will be addressed in a future AOS version.

Pickup Extension not enabled by default in "executive_users" Class of Service

Issue Detail

- Generally, all features are enabled in the "executive_users" CoS. However, Pickup Extension is not enabled by default for this CoS. It can be enabled by editing the CoS.

Filesystem timestamps incorrect

Issue Detail

- Some FTP clients such as Filezilla and Windows Explorer may report the time and date on files on CFLASH as incorrect values.
- To correct this, a command has been added to allow these timestamps to be corrected just after boot-up. The command is “run checkdisk”. This command can only be run from the enabled prompt during the first 5 minutes after a NV7000 boots so as to reduce the risk of affecting the Voicemail system integrity.

[Web]: Can't edit multiple Pickup Groups

Issue Detail

- Multiple Pickup Groups could be selected and Edit could be selected but on the first group would open for editing.

[Web]: Find Me/Follow Me Help text incorrect

Issue Detail

- The Help text for the number field (used for Ring External action) states that a “9” must be added to the number in the field. This is incorrect. The “9” is not required and should not be entered.

[Web]: Error when using Update Directories for IP 700 Series phones

Issue Detail

- When selecting phone configs for IP 700 Series phones and then selecting Other Actions->Update Directories, an error is given stating that it cannot retrieve the directory for those phones. This is just a cosmetic error, as IP 700 series phones do not have per-phone directory files.

[Web]: Upload of Music on Hold files from web interface fails

Issue Detail

- Uploading a music file of G.729 or PCMU type from the Music on Hold web page fails.
- Custom music files can be uploaded via FTP to the Music folder on CFLASH. There are two folders there for G.729 and PCMU.
- This will be addressed in a future AOS release.

Issues Resolved in AOS A4.01.00

INVITE not canceled when sent to forwarded User Account

Issue Detail

- When a User Account is forwarded at the system level instead of on the phone, an INVITE is only sent to the user's phone if Forward Courtesy Ring is enabled (which is enabled by default). This is so the phone will have a notification of a missed call that was forwarded. However, in some cases this INVITE was not being cancelled due to not receiving the proper SIP 1xx provisional response quickly enough. The unit now waits longer for the proper 1xx provisional response so the call can be properly cancelled.

Cannot disable Override System Mode from Auto Attendant

Issue Detail

- When the Override System Mode was enabled, there was no option to disable Override from the Auto Attendant System Mode Digit Action. Now, the only mode offered when accessing the System Mode Digit Action is Override when the Override mode is enabled. Selecting this will disable Override.

“Num-rings 0” fails on Ring Groups

Issue Detail

- When configuring the Number of Rings to "0" on a Ring Group, calls are to be delivered without going to Call Coverage until one of the members answers. However, with previous AOS versions, calls would follow the largest "num-rings" of the Ring Group members instead and go to Call Coverage.

*61 SPRE Does Not Work for International Numbers

Issue Detail

- When adding a speed dial for international numbers with the *61 SPRE code, the system would return an error (484 Address Incomplete).

Remote Phone boot settings undefined

Issue Detail

- A file containing phone config defaults for remote phones was missing. This file has been added to the AOS image so that it will be extracted when the unit boots on this AOS version if the file was not already present from previous configuration.

Reboot upon failed Park/Retrieve

Issue Detail

- If an attempt to Park or Retrieve a call fails, it was possible that the unit could reboot.

Status Groups could be assigned to phones that don't support them

Issue Detail

- Only the ADTRAN IP 706/712 and Polycom IP 450, IP 550, and IP 601/650 phones support Status Groups. Previously status groups could be assigned to phones that did not support them, such as the IP 32x/33x, IP 430, and IP 501.

Cannot remove Call Coverage on SLA to Auto Attendant

Issue Detail

- If Call Coverage configured on a Shared Line Account and contained a reference to an Auto Attendant that was no longer configured, the Call Coverage statement could not be removed.

[UC]: Message count on ADTRAN phones only shows 1 new message

Issue Detail

- The Message Count at the top of the phone would only show 1 new message instead of reporting the actual message count. The reporting of the actual message count is not supported by NetVanta UC Server so the NetVanta 7100 has been changed to only light the Message Waiting Indication (MWI) LED and not show a Message Count on the ADTRAN IP 700 Series phones.

[Web]: User Accounts not listed in order in IP Phone Configs

Issue Detail

- When adding a secondary extension to a phone config, the list of User Accounts was not ordered by extension number.

[Web]: IP Phone Configs CSV import doesn't allow certain characters

Issue Detail

- Characters such as periods or apostrophes were not supported in the CSV import tool.

[Web]: Error when trying to change an SLA to an extension in Button Map

Issue Detail

- When trying to change an SLA button to one for an extension or other type, a scroll bar would appear and the box would be incorrectly formatted.

Issues Resolved in AOS A2.07.00.SB

DNS queries to secondary server with no valid route causes a lockup

Issue Detail

- If the configured primary DNS server stops responding to queries, the NetVanta 7000 will failover to the secondary DNS server. If there is no route to the secondary DNS server present in the route table, the unit will lockup when trying to query the secondary server.
- If both DNS servers are routable by the default route in the unit, this issue will never occur.

Can't login to change System Mode in Auto Attendant

Issue Detail

- When changing the System Mode via an Auto Attendant, the Auto Attendant would prompt the calling party for a password. The user would have to press # at the end of the password string in order to change the mode. This has been fixed by software making the "#" unnecessary.

Buzzing noise heard instead of ringback

Issue Detail

- When calls were transferred out of the Auto Attendant to a phone, instead of ringback the caller would hear a buzzing noise. The call would complete as normal but the normal ringback tone would not be heard.

Dial-By-Name Directory members lost on reboot

Issue Detail

- Members could be added to a Dial-By-Name directory and the directory would operate correctly until the unit was rebooted. The members in the directory would not restore in the configuration as the unit was booted.

"Callers" and "Dir" softkeys missing from Polycom IP3xx phones

Issue Detail

- The shortcut keys to the Callers and Directories menus of the Polycom IP3xx phones were disabled.

Calls to Ring Group members on a TA900 fail to complete

Issue Detail

- This occurs when calling a Ring Group or a User Account that coversages to a Ring Group with multiple TA900 analog users. When one of the TA900 analog users answers the call, the TA900 sends a re-INVITE to the NV7100, the NV7100 and then re-INVITES the original SIP user. The phone responds with a 200 OK, but the NV7100 never sends a 200 OK to the TA900, instead it sends an ACK back to the SIP phone with SDP containing all zeros.

Calls transferred to users on DND or Forward not routed properly

Issue Detail

- When calls are routed to phones or User Accounts where the Do Not Disturb (DND) or Forward features are enabled, the caller may hear "Mailbox?" instead of the user's Voicemail greeting.

Inbound DTMF from PSTN not always detected

Issue Detail

- DTMF detection is performed after the point where the RTP gains are applied. If the high RTP gains are applied and the amplification results in audio clipping, DTMF will not be detected reliably.
- To resolve this issue, DTMF detection will be moved to before the application of RTP gains in a future AOS release.

Unit may reboot with Email Forwarding enabled

Issue Detail

- With Email Forwarding enabled and very unique system configurations, the unit may reboot.
- Due to the uniqueness of the conditions, this reboot is not likely to occur in most configurations.

[Web]: Update Directories does not update Polycom phone directories

Issue Detail

- When using the procedure to update the System Directory for phones outlined here: <http://kb.adtran.com/article.asp?article=2605&p=2>, Polycom phone directories do not get updated.

[Web]: Caller-ID and Dialed-Number lists in the Personal Phone Manager disappear intermittently

Issue Detail

- When the caller-id string included an apostrophe the associated Caller-ID and Dialed-Number list in the personal phone manager was unable to parse correctly ultimately preventing the entire list to be populated. This has been fixed by allowing the apostrophe to be parsed.

[Web]: Forward Disconnect incorrectly disabled

Issue Detail

- When you create a new analog User Account, the web interface takes you to the detailed config page for the new User Account. On the User Config tab, Forward Disconnect Delay is shown as Disabled but the CLI shows it is 500ms. If you click Apply or change any values and click Apply, the web interface disables the Forward Disconnect Delay since the value selected is Disabled in the web interface.

[Web]: Cannot Add or Delete Reject Template for SLAs and SCAs

Issue Detail

- For Shared Line Accounts (SLAs) or Shared Call Appearances (SCAs), clicking Add Reject Template adds the value to the Accept Template. Also, selecting a template and clicking Delete Template does not remove the template or provide an error.
- These values could be configured in the Command Line Interface (CLI) without issue.

Upgrade Instructions

Several steps need to be taken to assure a valid upgrade. First, save your existing configuration via the Configuration page in the web interface under Utilities (remember to include voice settings).

Accessing AOS A4.03.00.SB

AOS A4.03.00.SB is available by going to <http://www.adtran.com/support> and selecting Software Downloads. Then choose the NetVanta 7060 or 7100.

AOS Upgrade Instructions

1. Upload the AOS Image to FLASH via the Firmware page in the web interface or via FTP.
2. From the web interface, choose the new image as the Primary Firmware and click Apply.
3. (Optional) Copy previous Primary AOS image to CFLASH.
4. If using the web interface, select the Primary and Backup images from the drop-down lists and click Apply. If using the Command Line Interface in Global Configuration Mode, enter "boot system flash NV7100A-A4-03-00-SB-E.biz <location> <filename> verify" where <location> is the location of the backup firmware image and <filename> is the name of that firmware image..

The "verify" keyword tells the system to check the AOS image to make sure it was uploaded properly before applying it. Note that the filename may be different for other NetVanta 7000 Series products.

5. After the AOS image is applied, click Reboot unit or enter "reload" and type "y" to save and to reload.

AOS Bootcode Details

When upgrading to AOS A4.03.00.SB, an upgrade to Bootcode version A2.06.B1.01 required. Contact ADTRAN Technical Support for this Bootcode version and instructions for loading it.

IP 700 Series Phone Upgrade Instructions

To upgrade your IP 700 Series Phones, you will need to complete 3 basic steps:

1. Upload the new firmware and bootrom files to the ADTRAN directory on your unit via FTP.
2. Select the appropriate new version of firmware and bootrom for the IP 706 and IP 712 on the IP Phone Globals page, on the Boot Settings tab, under Default Firmware.
3. Reboot the phones and confirm they download the new firmware images.

Polycom Phone Upgrade Instructions

To upgrade your Polycom Phones, you will need to complete each of the following steps:

1. Copy the entire contents of the zip file found on ADTRAN's website to the Polycom folder on CFLASH using FTP.
2. Erase sip.ld from the Polycom folder on CFLASH.
3. Upload both sip.cfg and sip.ver from the zip file to FLASH/NONVOL using FTP. These files can also be copied from CFLASH using the following commands:
 - a. "copy cflash Polycom/sip.cfg flash sip.cfg"
 - b. "copy cflash Polycom/sip.ver flash sip.ver"
4. Reboot the phones and confirm they download the new firmware images.

Appendix A – Errata for A4.03.00.SB

The following is a list of errata that exists in A4.03.00.SB.

ERL test failing

Issue Detail

- ERL tests on the FXO interfaces will fail to test correctly in AOS A4.03.00.SA and above. The symptom seen is that the Operator Group will be called during the test.
- Downgrading to AOS A4.02 for this testing, completing the test and saving the results before upgrading back to A4.03.00.SB is recommended.

[Web]: Cannot change Call Coverage for Ring Groups or in Personal Phone Manager

Issue Detail

- Configuration of Call Coverage in multiple areas will appear to apply without an error but will not be saved.
- Areas affected are Call Coverage for Shared Line Accounts, Ring Groups, Global Call Coverage Lists, and to FindMe-FollowMe (in Personal Phone Manager).
- Command line configuration of Call Coverage is saved properly and can be used as a workaround.

Attended transfer with SIP trunk results in one-way audio

Issue Detail

- During an attended transfer involving a SIP trunk, if a SIP 183 message is received on the call leg used for the attended transfer, it is possible for the original calling party and the party receiving the transfer to have one way audio.

Reboot when receiving orbit number in SIP Proxy messaging

Issue Detail

- If there are SIP phones registered through the SIP proxy, the unit will reboot if the proxy phone sends a SIP message that contains an orbit number.

Memory leak due to HTTPS configuration

Issue Detail

- Using the HTTP secure server to access the GUI on a non-standard port number caused a memory leak.

FMFM extension used as ISDN Referring Number

Issue Detail

- If a call comes into a PRI with ISDN Redirecting Number enabled and is referred back out of the same PRI by FindMe-FollowMe, the ISDN Redirecting Number will contain "FMFM" and the extension of the voice user that was initially called. This may result in the call being rejected.

Ring Group call failures with TA900 user as a member

Issue Detail

- When a TA 900 user is a member of a Ring Group on a NV7000 and that user answers the call, it receives an INVITE w/ Replaces and which triggers a re-INVITE. The NV7000 (where the Ring Group resides) is still waiting on a re-INVITE response from the SIP server and responds with a SIP 491 message. If the SIP server does not respond in time, the TA 900 will give up and the call will fail.

Number of Rings for Ring Group not honored for all System Modes

Issue Detail

- Configured number of rings before Call Coverage is not honored in non-Default System Modes.

Echo not fully cancelled

Issue Detail

- In some installations, echo may not be fully canceled. The condition of the analog trunks is key in the cancellation of any echo in the system.

SIP access-class not working after reboot

Issue Detail

- SIP Access Classes are not restored to the configuration after a NV7000 is rebooted.

[UC]: T.38 fax to UC Server may fail to complete

Issue Detail

- Fax call does not process between NetVanta UC Server and NetVanta 7100 after the 7100 sends T.38 re-INVITE.

Unable to park placed call

Issue Detail

- Calls placed by a user cannot be parked.
- Calls received by a user can be parked.

Phone Firmware upload via GUI fails

Issue Detail

- Option to Upload Firmware from Default Firmware tab under IP Phone Globals->Boot Settings fails with an error.

Analog trunk receiving FSK and passing to multiple trunk ports

Issue Detail

- When a call is active on an analog trunk, the FSK passed on a new inbound call on another analog trunk can sometimes be heard by the local user using the first trunk.

Cannot transfer from FMFM via *88 to external number

Issue Detail

- With Enhanced FMFM enabled and while on a call delivered to a cellphone via FMFM, *88 cannot be used to transfer the call to an external number. This is because an incorrect SIP diversion header is sent on the external call.

Cannot perform attended transfer on call after Call Pickup is used

Issue Detail

- When using Call Pickup, you cannot perform an Attended Transfer on the picked up call.
- Unattended and Blind transfers will work..

E1 interface goes into line loopback after LOS (Loss of Signal) occurs

Issue Detail

- After running a test pattern from the E1 VIM, a loss of signal condition causes the interface to go into loopback.

Unregistered User Account clears call during FindMe-FollowMe transfer

Issue Detail

- Actions including contacting an unregistered User Account will be skipped when FindMe-FollowMe is activated.

“9” stripped from calls that are transferred externally from Auto Attendant

Issue Detail

- Calls that are received into the Auto Attendant and by Digit Action are transferred to an external number will have a “9” stripped from the dialed number. The help text for this Digit Action does not reflect this properly and can result in mis-configuration.
- A “9” should be added to the external number string.

Noise heard when two parties are speaking at the same time

Issue Detail

- If both parties speak at the same time during a call, part of the audio from one party may be distorted or sound like “white noise”. This generally only occurs when the inbound audio from the trunk side is very low and echo cancellation is engaged to reduce what it perceives as echo. Adjusting audio gains can reduce or alleviate this issue.
- This issue can also occur when a caller speaks while external Music on Hold is being played.
- Modifications to echo cancellation are being made to address this issue in a future release of AOS.

SMTP Forwarding fails when SMTP Server returns a banner message

Issue Detail

- If a server sends a banner response to the initial SMTP EHLO message, the SMTP negotiation will fail and the email notifications will not be sent.
- Removing the banner from the SMTP server will alleviate this issue.

IP 700 Series Message LED doesn’t light for secondary extensions

Issue Detail

- If a secondary extension is added to an IP700 Series phone config, it does not include the configuration necessary to light the Messages LED.
- Call Coverage can be directed to the primary extension’s mailbox or manual editing of the adtran_customer.txt file can add the necessary config statement.
- Adding “MwiSecondary X” (where “X” is the registration number of the secondary extension) to the “adtran_customer.txt” file will alleviate this issue until a solution is provided in a future AOS release.

Calls from FindMe-FollowMe to a Metaswitch softswitch may fail

Issue Detail

- If a call is directed by FM-FM out a SIP trunk to a Metaswitch softswitch, the call may fail if the Metaswitch responds to the INVITE with a 183 Session Progress message instead of a 180 Ringing message.

Calls sent to FindMe-FollowMe from an Auto Attendant are marked as internal

Issue Detail

- In FindMe-FollowMe Groups configuration, internal calls can be implicitly included in a group if so desired. It is important to note that external calls that are routed through an Auto Attendant are considered internal after being transferred out of the Auto Attendant.

Mroute is not removed when mcast-stub fixed is removed from an interface

Issue Detail

- If you remove 'mcast-stub fixed' from an interface with an igmp static group, it will remove the static group from the interface but the mroute remains. If you manually remove the igmp static group first and then remove 'mcast-stub fixed' from the interface, the mroute will be deleted.

Calls into Voicemail or Auto Attendant incorrectly record Lost Packets

Issue Detail

- The output of "show voice quality-stats" reports many lost packets on calls to Voicemail or Auto Attendant.
- These statistics are inaccurate for Voicemail and Auto Attendant due to the fact that no RTP packets are sent during the silences between prompts. They do not reflect actual voice quality issues.

Attended transfer with SNOM M3 handset fails

Issue Detail

- When attempting attended transfers with SNOM M3 handsets, the call is redirected to the transferor incorrectly.
- Unattended/Blind transfers do not experience this issue.
- No AOS changes have been made due to the fact that attended transfer is not supported when initiated by SNOM M3 phones.

[Web]: Cannot create new VLAN

Issue Detail

- When creating a new VLAN from the web interface, a 503 server error is presented and the new VLAN cannot be created
- The new VLAN can be created from the CLI without issue.

[Web]: System Directory may retain old values

Issue Detail

- When a First or Last Name is removed from a User Account, the System Directory will retain the previous value and not accept an update with an empty value.

[Web]: Polycom speed dials lost with directory update

Issue Detail

- Speed Dial entries on Polycom phones are lost when the system directory is updated for these phone configs.

[Web]: New system users created with blank portal list

Issue Detail

- Creating a new user for administering the AOS results in configuration of a blank portal list for this user. This will prevent the new user from being able to log in.
- This can be avoided by configuring the user from the CLI instead of the web interface.

[Web]: Associated Accounts not listed in order configured on phone

Issue Detail

- On the IP Phone Configs page, the list of Associated Accounts for each phone is displayed in numeric order rather than the order configured on the phone keys.
- If you select the phone config and view/edit it, the order of the Button Map is correct.

[Web]: Error when adding more than 10 Polycom MAC addresses

Issue Detail

- On the Manual Input tab, when entering more than 10 Polycom MAC addresses at the same time and saving them an error is given and the phone config is not saved.
- If the number of MAC addresses is limited, the issue will likely not occur.
- This may be addressed in a future release of AOS.

Appendix B – Related Documents

For configuration guides, installation guides, white papers and more, visit ADTRAN's knowledge base at <http://kb.adtran.com>.

AOS A4.03.SA Documentation

Personal Phone Manager Configuration Guide (UPDATED, includes FindMe-FollowMe) – <http://kb.adtran.com/article.aspx?article=2196&p=2>

E911 Configuration Guide – <http://kb.adtran.com/article.aspx?article=3454&p=2>

AOS A4.01 Documentation

Call Queuing Configuration Guide - <http://kb.adtran.com/article.aspx?article=3417&p=2>

Music on Hold Configuration Guide - <http://kb.adtran.com/article.aspx?article=3418&p=2>

User Account Configuration Guide (UPDATED, includes FindMe-FollowMe) - <http://kb.adtran.com/article.aspx?article=1896&p=2>