ADTRAN The Network Access Company

Release Notes
NetVanta 7000 Series
Products

AOS Release A2.04.00.SC
October 9, 2009

Contents

Purpose

AOS Release A2.04.00.SC provides several new IP-PBX features, enhances existing features and addresses several field issues as well

AOS A2.04.00.SC 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	8/3/2009
Controlled Release	Completed	10/7/2009
Generally Available Release	Available	10/9/2009

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 for AOS A2.04 and beyond

ADTRAN takes great pride in providing new and innovative features for its customers. Continuing in the same tradition we are now releasing AOS A2.04 in a controlled release manner. The new features coming with this release warrants 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 should on the 1200796L1 remains unchanged.
- Use of SIP trunks that require the NV7100 to do the 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 A2.04 feature 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 A2.04 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>".

Notice of Defined Voicemail File Limit

Starting with AOS A2.04.00.SA, 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 drive to a larger drive will not result in more voicemail storage and is therefore not recommended.

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 on page 14 of this document.

- ADTRAN IP 700 Series Phones – Version 1.3.11
- 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 A2.04.00 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 A2.04.00.SC being the currently recommended version.

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.SA and above, this traffic is no longer allowed by default. Instead the installer is required to selectively customize the Public policy class to allow SIP traffic from remote sites and SIP Trunking providers.

Updated list of Supported IP Phones

The web interface has been updated to reflect support for the following phones phone config generation with the NetVanta 7000 Series products:

- ADTRAN IP 706
- ADTRAN IP 712
- Polycom Soundpoint IP 320/330
- Polycom Soundpoint IP 430
- Polycom Soundpoint IP 501
- Polycom Soundpoint IP 550
- Polycom Soundpoint IP 601/650
- Polycom Soundstation IP 4000/6000

Summary of New Features

This section highlights the major features, commands and behavioral changes for AOS A2.04.00.SC.

Increased Voice User Support (from 50 to 100 users)

- Starting with AOS A2.04, the same NetVanta Chassis of 7100 and 7060 will be able to support up to 100 SIP Station accounts. This is double the number of previously supported maximum IP Stations.
- The above capacity enhancement is accompanied by:
  - A doubling of the voicemail storage capacity from a previously stated maximum of 1,500 voicemails to 3,000 voicemails.
  - The number of voicemail boxes increase from 80 to the new maximum value of 130.
  - The number of Auto Attendant and Voicemail ports continues to be 8 each, thus allowing for 8 simultaneous sessions of AA and VM each.

NOTE: Configuring and using more than 50 users is not recommended for first hardware version NetVanta 7100s. See the statement under *Important Notices* earlier in this document for how to identify the hardware version of your NetVanta 7100. The NetVanta 7060 and second hardware version NetVanta 7100 are unaffected.

Handset Paging Groups

- This feature allows the user to page other phones directly through the speakerphone. This is helpful for environments where Overhead Paging is not available or practical. For environments that need both methods of paging, the Overhead Paging port on the NV7000 Series can be included in the configured Paging Group.
- The page is delivered live rather than being stored and played back through the phones.
- To define which phones will be paged, you can configure a Paging Group and add those phones as members of the group.
- For more details on this feature, refer to the Technical Support Knowledgebase guide.

NOTE: Configuring and using Handset Paging Groups is not recommended for first hardware version NetVanta 7100s. See the statement under *Important Notices* earlier in this document for how to identify the hardware version of your NetVanta 7100. The NetVanta 7060 and second hardware version NetVanta 7100 are unaffected.

Capabilities
- Paging Groups can be members of other Paging Groups. For example, the Sales Paging Group and the Service Paging Group can both be part of a Paging Group. However the combined membership of such a group should not exceed the recommended maximum members in a paging group.
- Paging Group members that are on a call will not be interrupted, but the page will be presented as another call if they have an available appearance.
- Phones that are forwarded or have Do Not Disturb (DND) enabled will not receive the page and this will not interfere with the presentation of the page to other phones in the group.
- If the G.729 codec (default) is used, one "beep" is played for the caller to note that all phones are ready to receive the page and to proceed with the announcement. Two "beeps" are played if one or more phones cannot receive the page but the caller can still proceed with the announcement.
- Page recipients may be local or remote users.

Restrictions
- A maximum of 16 phones can be paged at one time across all Paging Groups. Hence 16 is the maximum number of recommended members to be added to any one paging group.
- A maximum of 10 Paging Groups can be defined in the system.

Direct Transfer to Voicemail on Polycom phones

- With AOS A2.04.00.SC and Polycom 3.1.3b firmware, a call can now be transferred directly to Voicemail from a Softkey on the Polycom phone. The "SendVM" key is available during a call if you press the "More" key. Press "SendVM", enter the desired mailbox and press the "Enter" key.

Enhanced SIP Authentication Password (Security Enhancement)

- A 16 digit alphanumeric password can now be assigned and used for SIP User Account registrations. The default passwords remain the same as before.

Shared Call Appearances

- Allows a Shared Line Account (SLA) to be configured, but not associated with a Trunk Account. This will allow the Shared Call Appearance (SCA) to appear on multiple phones for a small office environment.
- The advantage of this feature is that the same extension can now appear on multiple phones, thus allowing a user to have simultaneous ring and public hold across those phones with the shared call appearance.
- Another advantage of this feature is that you can allow DIDs from SIP or PRI trunks to show up as "appearances" on a phone and emulate the same Shared Line Account/Key System functionality previously available only with analog lines.

Capabilities
- SCAs can be configured and appear on multiple phones.
- SCAs can now be created as dial-able extensions and be called from any other endpoint.
- Calls received on a SCA can be placed on Public Hold and be available from any other phone that has that SCA configured.

Restrictions
- Only 1 call can be received per SCA. Call Waiting is not supported.
- Aliases and DIDs cannot be configured on an SCA. Instead, you should create the SCA as a dial-able extension.
- SPRE codes are not supported from SCAs (or SLAs).
- Voicemail is not available for a SCA. You can, however, configure Call Coverage to a specific mailbox.
- A SCA cannot be placed on Forward or DND.
- Call permissions or restrictions are based on the private extension of the phone. If no private extension is used on that phone, then the permissions or restrictions of the SCA are used.

DNIS over FXS

- FXS interfaces will now be able to send dialed digits out to devices on the FXS interface via DTMF.
- This feature is helpful when sending calls to an attached fax server with analog interfaces.
- It can be enabled on the User Account associated with the FXS interface from the web interface or from the CLI with the command "dnis-digits x" where "x" is 1-16 digits.
- The DNIS digit outpulse can be configured with delays before sending digits (after the call is answered) and before connecting the call but after the digits have been sent.

Summary of Enhancements

This section highlights the enhancements for AOS A2.04.00.SC.

Enhanced IP Phone & User Account Configuration Web Interface

- With AOS A2.04.00.SC, the User Account and IP Phone Config web pages of the NetVanta 7000 have changed considerably to accommodate configuring larger numbers of users and phones.
- These changes allow for an easier way of creating phone and user config..
- Note the following changes:
  - o The User Account page has been made easier to use and retains a similar look and feel of the previous AOS versions
  - o When new User Accounts are created, the default global Call Coverage List of "go_to_voicemail" is applied and the default Voicemail Class of Service "normal_voicemail" is assigned.
  - o The Phone Configs tab of the IP Phone Configs page has been fully redesigned to allow for individual, batch and CSV template import phone config generation methods.
  - o When a Phone Model is selected, the OUI (first 3 octets of the MAC address) are automatically populated.
  - o The remaining tabs on the IP Phone Configs page have been moved to a new page called IP Phone Globals since all of these settings are global to all or a group of phones.

NOTE: The updated pages could require an additional plug-in for your web browser to view the Flash-based content (Adobe Flash Player Version 10 or later). Most browsers will prompt you for this and allow you to download it automatically.

[Web]: Status Groups now auto detect members

- Starting with A2.04.00.SC, status groups may be configured to include members of an entire team (i.e. Sales). When that Status Group is added to a phone, the individual user is omitted. This enhancement keeps users from monitoring themselves and eliminates the need for creating multiple status groups with each one removing a single member.

NOTE: This feature is considered at beta status because it has not completed our full testing suite.

Summary of Bug Fixes

This section highlights major bug fixes in AOS version A2.04.00.SC

User Account resource assigned but not released
Issue Detail
- User Account resource does not free. If a BYE is received at the same time an INFO message is sent as a keep alive, the call will not terminated properly. If all the User Account resources remain in this state, the user will not be able to place phone calls. This only occurred with SIP keep-alive messages enabled on the User Accounts.

RTP resource assigned but not released
Issue Detail
- A resource used to produce ring-back in the auto-attendant was not being released after transcoding from G.729 to G.711. A symptom of this issue was the lack of ring back when a user was dialing an extension via the Auto Attendant.

Caller-ID-Override fails in unattended transfer
Issue Detail
- The command "caller-id-override number-inbound" on a Trunk Account would modify the Caller ID when a call was transferred from the station that originally received it. The override is only supposed to occur initially.

Only one call or user can access Voicemail
Issue Detail
- Attended transfers to Voicemail or Auto Attendants could result in resources for those accounts getting stuck in a busy state.

User can be locked out of Voicemail access
Issue Detail
- When accessing voicemail, it was possible for a user to be locked out of the voicemail system if the user entered multiple incorrect passwords.

Voicemail check account busy
Issue Detail
- In rare cases, a voice mail account could get stuck in a busy state.

Calling Party Number Presentation Restricted not honored by Voicemail
Issue Detail
- When a call is received on a PRI trunk with the Presentation field set to Restricted, the Calling Party Number should not be displayed or reported. When this type of call follows Call Coverage to Voicemail, however, the restricted number is recorded in the Voicemail message envelope and played when the message is retrieved.

Possible reboot when accessing CFLASH
Issue Detail
- A reboot could occur due to filesystem access priority error.

Outgoing SMTP rejected
Issue Detail
- Some SMTP servers or proxies would reject outbound email logging or Voicemail notifications due to incorrect formatting.

Possible reboot during heavy voicemail load

Issue Detail

- Under very stressful voicemail loads, the unit could reset. Conditions require a full Voicemail system (1500 messages), and access to VM by multiple callers (8 max) simultaneously for 10's to 1000's of hours.

SPRE *34 fails to forward remote user account

Issue Detail

- When entering a SPRE code to forward a specific account, the user account that originated the SPRE code is forwarded instead of the specified account number.
- For example:
    1. Extension 8867 calls *343029*1234*3430*
    2. The NV7000 accepts the forward but forwards extension 8867 to 3430 instead of extension 3029.
    3. Since the phone has no way of knowing about User Account forwards, the phone will no longer receive any calls.
- This would affect older configurations using *34 to forward a virtual extension for Day/Night operation. However, the recommended method to achieve Day/Night mode operation is to implement System Modes (***Configuring System Modes in the NetVanta 7000 Series*** - http://kb.adtran.com/article.asp?article=2313&p=2).

Paging port/account busy

Issue Detail

- If a user calls the SPRE code to page overhead and the call terminates abnormally, the paging account/port would remain busy.
- A timeout value can now be configured using the command "voice overhead-paging connected-timeout x" where x is a value in seconds and the default is 120 seconds.

Phone display freezes with a call that is no longer actively ringing

Issue Detail

- In the case where a call is delivered to a phone via a User Account, Ring Group or Shared Line Account, a SIP INVITE message is sent to the phone. If the caller terminates the call before the phone responds to the INVITE with a 100 TRYING message, the NetVanta 7000 Series product may not send a CANCEL message to clear the call properly. This resulted in the phone display showing a call ringing when it had been cleared from the trunk side.

Congestion may cause loss of encrypted packets for VPN

Issue Detail

- Under heavy load, the NetVanta 7000 Series product could not service packets at the same rate required to encrypt for VPN tunnels. This caused the unit to drop packets. Also, input decryption errors were reported to the terminal due to encrypted packets missing in the sequence. Throughput performance was slightly affected.

No ringback on multi-level transfer (Polycom phone only)

Issue Detail

- In the following scenario, the caller does not hear ringback.
    1. Call is received from PSTN via a SIP Trunk.
    2. Call is routed to an Auto Attendant and the caller selects "0" for the Operator Group.
    3. An Operator Group member answers the phone and then transfers to an internal user extension. During the transfer, no ringback is heard by the external caller.

On transfer, Caller ID Override "if-no-cpn" option fails

Issue Detail
- When "caller-id-override number-inbound X if-no-cpn" is configured on a Trunk Account, the Caller ID should only be changed if there is no Caller ID number as the call is received on the trunk. In this case, if a call with Caller ID was transferred, the Caller ID number was changed based on what is configured for the Caller ID override.

User Account Speed Dials limited to 13 digits

Issue Detail
- User Account Speed Dials can be configured with more than 13 digits, but when they were stored they are truncated to 13 digits. This affected speed dial entries for international numbers.

Calls on Shared Line Accounts should not be allowed to be parked

Issue Detail
- It is invalid to allow calls into Shared Line Accounts to be parked. However, options on some phones allowed this function. The result was that the SPRE code to park the call was processed out the Trunk Account for the SLA.

[Web]: IP 700 Display Name truncated to 15 characters

Issue Detail
- On the detailed Phone Config page, you can configure a Display Name that is longer than 15 characters, but the config file generation truncated the Display Name to 15 characters.

[Web]: Can't add more than 5 secondary lines to phones configs

Issue Detail
- On the detailed Phone Config page, a single phone config was limited to 5 secondary lines. It is valid in some cases to configure more but this is limited by the web interface.

[Web]: Grammar settings reset when ANI or DNIS substitution is added

Issue Detail
- When adding a DNIS or ANI:DNIS substitution on a Trunk Account, the From Header grammar setting reverted to the default value.

[Web]: Shared Call Appearance created incorrectly with Internet Explorer

Issue Detail
- When using Internet Explorer to create a new SLA with the SCA box checked, the new account was created but the SCA box was unchecked and all fields were blank (Description, Line Seize Timeout, and SIP Auth Password).
- If the SCA box was checked again, it would not gray out the Trunk Accounts selection.
- If the rest of the values were completed and Apply was clicked, the values were not saved and the page failed to close and return to Shared Line Accounts.
- This only occurs with Internet Explorer. Firefox and other browsers work without issue.

[Web]: Global Call Coverage list reported incorrectly

Issue Detail
- When viewing a User Account, the Call Coverage List displayed the last configured value instead of the currently configured value. If you clicked Apply it would change the existing value to the one that was displayed.

[Web]: Polycom phone config changes from Remote to Local when adding a speed dial entry

Issue Detail
- When a speed dial entry is added on the Button Map tab for a phone config, the Boot Profile on the Phone Settings tab was changed from Remote Phone to Local Phone.

[Web]: Assigned extensions are not removed from the list of available extensions

Issue Detail
- If an extension is added to a phone config, that extension was still listed as available to add until the phone config was saved.

[Web]: The same extension can be added to two different IP Phone Configs

Issue Detail
- When adding phone config files in batch (manual or CSV), a User Account could be added to two different phone configs. This was because the User Account was not assigned to the phone config until it was saved.

[Web]: Tab after adding Speed Dial opens Advanced Settings

Issue Detail
- After adding a Speed Dial entry and using Tab to move to the next available field after Label, the Advanced Settings box would open.

[Web]: Incorrect PIN or SIP Auth Password saved

Issue Detail
- When changing from the default PIN/SIP Auth Password of "1234", the value would be saved as "encrypted" when Password Encryption was not enabled.

[Web]: Sync dialog box cannot be closed

Issue Detail
- When importing a large CSV file for batch phone configs, once you click Continue you are prompted with a dialog box asking if you wish to sync the large list of phones. This box could not be acknowledged.

[Web]: More than one Status Group can be selected in a phone config

Issue Detail
- On the Button Map tab, a Status Group could have been selected even if another Status Group was already assigned to begin with another key. The result in saving this config though was that the existing Status Group was replaced with the newly selected Status Group. Only one Status Group is supported per phone.

[Web]: Adding new extension to existing phone config results in wrong CoS

Issue Detail
- When adding a new User Account to a phone config and creating the User Account in the process, the new user default Class of Service and Voicemail Class of Service were not assigned.

[Web]: Status Group extension members are added with a leading "**"

Issue Detail
- Status Group members for ADTRAN Phone configs were generated with a "**" in front of the extension which results in an invalid Status for that member (icon with a question mark over it).

[Web]: New user defaults not assigned when creating virtual users
Issue Detail
- When a User Account was created as Virtual (instead of SIP or Analog), the default values for Class of Service, Call Coverage, and Voicemail were not assigned.

[Web]: Default Settings are not applied to new and existing users
Issue Detail
- On the IP Phone Globals page, when applying values on the Default Settings tab you can choose whether to apply to New Configurations or New and Existing Configurations. Selecting New and Existing Configurations would only update the settings for New Configurations.

[Web]: Adding a new extension defaults to one line key
Issue Detail
- When adding a secondary extension to an existing phone config, only one line key was added by default. For ADTRAN phones, at least 2 line keys are needed for that extension to allow transfers.

[Web]: Error when clicking an extension on the Voicemail Status page
Issue Detail
- A 404 Not Found error was displayed when navigating to the Voicemail Status page and clicking on an extension.

[Web]: Applying changes to a Shared Call Appearance changes seize-timeout
Issue Detail
- When applying any changes to a Shared Line Account configured as SCA (Shared Call Appearance), the seize-timeout value would be changed from the default of 15 seconds to 5 seconds. This could result in a busy signal when a user takes longer than 5 seconds to dial from a SCA.

[Web]: Communication error during CSV import
Issue Detail
- When importing a CSV file containing phone MAC and model information, a communication error was presented when the staged phone configs were saved.

[Web]: Forward Disconnect Delay disabled when creating a new analog user
Issue Detail
- When creating a new analog User Account, the value shown for Forward Disconnect Delay on the User Config tab was "Disabled" but the actual value configured in the system was "500ms". If you clicked Apply, the web interface would disable Forward Disconnect Delay. If you clicked Cancel first and edit the User Account again, the web interface would reflect the correct value of "500ms".

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 A2.04.00.SC

AOS A2.04.00.SC 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 cflash NV7100A-A2-03-00-E.biz X Y verify" where "X" is the location of the backup firmware image and "Y" 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, then click Reboot unit or enter "reload" and select "y" to save and to reload.

AOS Bootcode Details
When upgrading to AOS A2.04.00.SC, an upgrade to the Bootcode is not required.

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 your boot server 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 A2.04.00.SC

The following is a list of errata that exists in A2.04.00.SC.

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.

Errata Justification
- 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.

*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 will return an error (484 Address Incomplete).

Errata Justification
- This will be addressed in a future release of AOS.

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.

Errata Justification
- Attended transfer is not supported when initiated by SNOM M3 phones.

"Num-rings 0" fails on User Accounts and Ring Groups

Issue Detail
- For User Accounts, a call may be cancelled after 17 rings and Call Coverage will be processed.
- For Ring Groups, a call may be cancelled after the highest "num-rings" value for members in the Ring Group. Call Coverage will then be processed.

Errata Justification
- This will be addressed in a future release of AOS.

[Web]: Error when viewing Trunk Account config page

Issue Detail
- If a SIP Trunk Account has ANI or DNIS Substitution configured, that page will fail to load with a 503 Server error.

Errata Justification
- The configuration for the Trunk Account can be modified from the Command Line Interface (CLI).

[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.

Errata Justification
- If you select the phone config and view/edit it, the order of the Button Map is correct.
- This will be addressed in a future release of AOS.

[Web]: Personal Phone Manager Caller-ID List disappears intermittently

Issue Detail
- The Caller-ID List content disappears from the Personal Phone Manager webpage. However, this only lasts for a few hours and the list contents will be visible again.

Errata Justification
- This will be addressed in a future release of AOS.

[Web]: Error on configuring Call Coverage to external number

Issue Detail
- When configuring Call Coverage to an external number, the web interface does not insert a "9" in front of the number.

Errata Justification
- This will be addressed in a future release of AOS.

[Web]: Class of Service Member List may not load

Issue Detail
- Intermittently, the Member List may not load when clicking on the number of Users Assigned to a specific Class of Service.

Errata Justification
- This will be addressed in a future release of AOS.

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

Issue Detail
- For Shared Line Accounts (SLAs), 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.
- For Shared Line Accounts configured as 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 and instead reorders the list.

Errata Justification
- These values can be configured in the Command Line Interface (CLI) without issue.
- This may be addressed in a future release of AOS.

[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.

Errata Justification
- If the number of MAC addresses is limited, the issue will likely not occur.
- This may be addressed in a future release of AOS.

[Web]: Delay after clicking Apply on Shared Line Account detail page

Issue Detail
- After clicking Apply on a specific Shared Line Account (SLA) page, there could be a 10-30 second delay before returning to the main SLA page.

Errata Justification
- This may be addressed in a future release of AOS.

Appendix B – Related Documents

The following are documents related to the new features included in this AOS Release as well as other new documents that have been recently posted to the ADTRAN Technical Support Knowledgebase.

NetVanta 7000 Series Feature Related Documents

Handset Paging Groups Configuration Guide -
http://kb.adtran.com/article.asp?article=3102&p=2
- Configuration guide for new feature that allows hands-free paging through the phones.

Shared Call Appearances Configuration Guide -
http://kb.adtran.com/article.asp?article=3101&p=2
- Configuration guide for new feature that allows hands-free paging through the phones.

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