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Release Notes NetVanta 7000 Series Products

AOS Release A2.07.00.SB
August 3, 2010

Contents

Purpose

AOS Release A2.07.00.SB provides a few new features, enhances existing features and addresses several field issues.

AOS A2.07.00.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	6/24/2010
Controlled Release	Completed	7/30/10
Generally Available Release	In Progress	8/3/10

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

Supported Hardware Platforms

This AOS image supports the following hardware:

- **NetVanta 7100**
    - Part Numbers: 1200796E1 and 1200796L1 (see Hardware Considerations below)
- **NetVanta 7100 Small Business Bundle**
    - Part Number: 4200796G1#VSMB
    - More details on the VSMB Bundle available here: http://www.adtran.com/bigdeal .
- **NetVanta 7060**
    - Part Number: 1700706G1

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 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 A2.05 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 A2.05 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>".

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 card to a larger card will not result in more voicemail storage and is therefore not recommended. Should you have a need to replace a failed CFLASH card, please contact ADTRAN Technical Support for assistance.

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 A2.07.00.SB 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 A2.07.00.SB being the currently recommended version.

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 1.3.13
- 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.

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.

Summary of New Features

This section highlights the major features, commands and behavioral changes for AOS A2.07.00.SB and A2.06.00.

Features added in AOS A2.07.00.SB

Added configurable "notify-ringing" feature to Status Groups

- This feature, which is enabled by default, will allow the Status Group to be notified when the Status Group member phone is ringing in addition to being on an active call. This is compatible with existing functionality.
- To only monitor whether the member phone is idle or busy with a call, you can disable this feature with the command "no notify-ringing" in the CLI. This command is not supported in the web interface at this time.
- Disabling this feature may reduce the number of Status Group updates sent to phones and could help with phone performance issues on phones with large status groups (such as the Polycom IP 601).

Added ability to define a default Voicemail Class of Service

- A Voicemail Class of Service can now be defined as the New User Default.
- When new User Accounts are created from the web interface, the default Voicemail Class of Service will be applied as long as a Voicemail Class of Service is selected as the default. This also applies to new User Accounts created from the CSV Import feature in IP Phone Configs.

Features added in AOS A2.06.00

Added support for additional Polycom phone models

- Added support for the Polycom IP 450, IP 321, and IP 331.
- These phone models should be available for selection in IP Phone Configs.

Enhanced IP Phone Config Generation with CSV Import

- AOS A2.04.00.SC added the ability to import a CSV file containing user extension information to generate phone configs in bulk. Now this capability has been expanded to allow all of the following to be defined within the CSV Template:
    - MAC address of the phone to be configured
    - Extension number for the new phone
    - Phone Model
    - First and Last Name of the user associated with this phone
    - SIP Authentication Password
- Since the Extension, First and Last Name are defined in the CSV template, a new User Account is created for each extension that is populated.

Summary of Bug Fixes

This section highlights major bug fixes in AOS version A2.07.00.SB and A2.06.00.

Resolved in AOS A2.07.00.SB

One-way audio after attended transfer

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 was possible for the original calling party and the party receiving the transfer to have one way audio.

Possible reboot when receiving a REFER with missing Caller ID from a SIP trunk

Issue Detail

- When a call is sent to the NetVanta 7000 Series product via a SIP REFER that did not contain the Caller ID information, a condition could be created where the unit would reboot.

Unit may reboot with Email Forwarding enabled

Issue Detail

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

Lockup condition upon unit restart

Issue Detail

- Configuring the "voice compand-type" different than the System Country default could cause the unit to stop responding after rebooting to apply the configuration change.

System may reboot after a call clears

Issue Detail

- If a call is blind transferred to an endpoint that redirects immediately to another endpoint that answers immediately, a reboot could have occurred. This could also have occurred on a transfer to a phone that has Do Not Disturb (DND) enabled and Call Coverage goes to Voicemail.

Unit reboot when packet replication occurs

Issue Detail

- When invoking packet replication to copy audio to multiple endpoints (such as Paging Groups), the unit could possibly reboot.

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 would hear “Mailbox?” instead of the user’s Voicemail greeting.

Removed “voice forward-mode” command

Issue Detail

- Removed command "voice forward-mode" from NetVanta 7000 Series because an incorrect setting of this command will result in compatibility problems.

Inbound DTMF from PSTN not always detected

Issue Detail

- DTMF detection was previously performed after the point where the RTP gains are applied. If the amplification results in clipping, DTMF will not be detected reliably. To resolve this issue, DTMF detection has been moved to before the application of RTP gains.

Audio from recorded prompts not marked with proper DSCP value

Issue Detail

- The configured DSCP value was not set on RTP packets from Voicemail, Auto Attendants, or Loopback Accounts.

Error with "traceroute" command

Issue Detail

- When entering only "traceroute" on the CLI and pressing Enter, the CLI would prompt for a "Target IP address" but would not accept the address. Instead, the CLI would continue to prompt for "Target IP address".

SendVM function on Polycom phones fails in some cases

Issue Detail

- The default phone configuration for Polycom phones would not allow for the transfer to Voicemail if a call was on-hold or if the system extensions were more than 4 digits.

[Web]: Voicemail notification schedule incorrectly formatted

Issue Detail

- The web interface would format the Voicemail notification schedule such that it may incorrectly show an end time being before a start time.

Resolved in AOS A2.06.00

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.

ISDN Reboot condition

Issue Detail

- Unique network call conditions and usage levels uncovered a reboot condition in the ISDN code.

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.

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 coverages 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.

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.

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 was not enabled by default for this CoS.

"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.

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.

[Web]: Default Settings not applied to new and existing Phone Configs

Issue Detail

- When making changes on the Default Settings tab under IP Phone Globals (such as Dial Strings, etc), the changes were not being applied to new or existing phone configs.

[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 did 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]: 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.

[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.

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 A2.07.00.SB

AOS A2.07.00.SB is a Controlled Release. Contact ADTRAN Technical Support or your local Sales Representative for a copy of this AOS

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-A2-07-00-SB-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 A2.07.00.SB, 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 A2.07.00.SB

The following is a list of errata that exists in A2.07.00.SB.

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.

Attended transfer to Ring Group may fail

Issue Detail

- When performing an attended transfer to a Ring Group, the call will drop when the Ring Group member answers if the call to announce the transfer is shorter than 10 seconds.
- If blind or unattended transfer methods are used or if the call to the transfer target to announce the attended transfer is longer than 10 seconds this will not occur.

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.
- This will be addressed in a future AOS release.

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 will be presented on the Speakerphone of the IP 700 Series phone.
- Pressing the Speakerphone button will put the call back on the user's handset and pressing the Headset button will put the call back on the user's headset.

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.

*67 Caller ID restriction not preserved on calls forwarded externally

Issue Detail

- If a user invokes Caller ID restriction using the *67 SPRE code and the party he calls has forwarded their phone, the Caller ID restriction will not be honored if the call is forwarded out a trunk.

[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.

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]: 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.
- This will 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.
- 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.