The Network Access Company

Corporate Office
Adtran, Inc.
901 Explorer Boulevard
Huntsville, AL 35806

U.S. Mail
P.O. Box 140000
Huntsville, AL 35814-4000

General Information
800 9ADTRAN
info@adtran.com
www.adtran.com

Pre-Sales Technical Support
800 615-1176
application.engineer@adtran.com
www.adtran.com/support

Post-Sales Technical Support
888 423-8726
support@adtran.com
www.adtran.com/support

ACES Help Desk
888 874-ACES
aces@adtran.com
www.adtran.com/support

Release Notes NetVanta 7000 Series Products

AOS Release A2.03.00.SC
April 22, 2009

Contents

Purpose

AOS Release A2.03.00 provides enhancements and addresses several field issues.

Release A2.03.00 is a Controlled Release. This AOS is recommended only for new installations and existing installations with issues resolved in this release. Once it completes the Controlled Release period, it will be considered Generally Available AOS. It 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 Release. Issues discovered during testing but not addressed in this build are listed as Errata in Appendix A.

Stage	Status	Start Date
Internal and Field Evaluation	Completed	2/13/2009
Controlled Release	Begun	3/14/2009

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

Notice of Defined Voicemail File Limit

The NetVanta 7000 Series products can maintain a maximum of 1500 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.

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 as of AOS A2.01.00, there will no longer be enough space on FLASH/NONVOL to store 2 versions of A2.xx.xx. To copy the current image from FLASH/NONVOL to CFLASH, use the command “copy flash <filename> cflash <filename>”.

Required ADTRAN IP 700 Series Phone Firmware

For this AOS Version, IP 700 Series Phone firmware version 1.3.3 or above is required. Version 1.3.7 is now released to address issues found in the field and is available for download at http://www.adtran.com/support by selecting IP Phones and Stations and then IP 700 Series Phones.

Required Polycom Phone Firmware

For this AOS version, it is necessary that your phones are running Polycom SIP version 2.1.2 for proper operation with the NetVanta 7000 Series Products.

Use the following links to access the latest Phone Firmware.

- **Polycom application version 2.1.2 and bootrom version 3.2.3 -** http://kb.adtran.com/PolycomFirmware212/Version2.1.2.zip
    - **This zip file contains: sip.ld, sip.cfg, and bootrom.ld**
    - **All files are necessary to upgrade the phones**

These files can also be downloaded by going to http://www.adtran.com/support , selecting IP Phones and Stations and then ADTRAN Branded Polycom phones. Contact ADTRAN Post Sales Technical Support if you are unable to download these files.

Default Configuration/Operation Changes

The following configuration items have had their default value changed

AOS Changes

- “blind-dial” (on analog Trunk Accounts) : now disabled by default as of A2.03.00
- “collector primary x.x.x.x”: changed default reporting protocol from TCP to UDP

Updated design of onboard analog FXO ports (trunks 0/1 and 0/2) for the NetVanta 7060 and NetVanta 7100

New FXO design enhances ring trip performance with certain ATA’s (Analog Telephone Adapters) and improves high frequency noise immunity (discovered when ADSL was not adequately filtered). A minority of installations exhibited problems and have been resolved with new hardware. Existing installations not experiencing such an issue will not require or benefit from these changes.

These enhancements are currently available on the 4-Port Trunk/FXO (part number 1202691G1) and 2-Trunk(FXO)/2-Station(FXS) (part number 1202691G1) Voice Interface Modules.

Updated list of Supported IP Phones

The following phones are supported for operation and 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 4000/6000/7000

The web interface will be updated to reflect this list in AOS A2.04.00.

Summary of New Features

This section highlights the major features, commands and behavioral changes for AOS A2.03.00.

Automated ERL Tool

- In AOS A1.02.00 a new debug was added to analyze analog lines connected to FXO ports. This capability has been expanded into an automated tool. This automated tool has completed testing and is now released. The details of the tool and debug command are outlined in the ***NetVanta 7000 Series Echo Return Loss Measurement Guide*** - http://kb.adtran.com/article.asp?article=2345&p=2

Note that this command is only available on NetVanta 7060 and NetVanta 7100 units with the most recent DSP hardware and part number 1200796E1. The process to determine which DSP hardware you have is included in the* NetVanta 7000 Series Echo Return Loss *Measurement Guide.

SIP Keep-alive for IP Endpoints

- Support of sending INFO or OPTIONS based on configuration. This will terminate connected calls if the SIP endpoint no longer responds after a configurable timeout.

Automatic Gain Level Control for Voicemail Attachments

- A new command, “voicemail attachment-level”, has been developed to adjust the audio level of WAV attachments to Email Notification of Voicemail. The default setting is enabled.

Configurable Ring Group Prefix

- Previously, a prefix displayed as “OPR_” or “GRP_” on an inbound call to the Operator Group or a Ring Group could be enabled or disabled. Now an alphanumeric prefix can be configured that will still be followed by the trailing ‘_’ and then the Caller ID number if available. There is a 40 character limit for this prefix. Configuration of this option is currently available from the CLI only. It will be configurable from the web interface in a future AOS release.

SIP Diversion Header Support

- This feature allows for the retention of the originating number when a call is processed through an Auto Attendant, transferred from a user extension, or is forwarded by a user phone to an external number. For these calls, the number included in the From: field of the SIP messages will be the original Caller ID and is subject to the ANI Substitution and a SIP Diversion Header is added with the original Caller ID number. The number included in the diversion header is added as the SIP Identity (if configured) of the User Account or Ring Group that is forwarding the call. If not configured, it is the ring group/user account extension.
- Calls from internal extensions are subject to the ANI Substitution configured on the Trunk Accounts without the addition of a SIP Diversion Header.
- This feature only applies to calls received on a Trunk Account and that are transferred or forwarded out another Trunk Account.
- CLI command added: [no] diversion-supported (on SIP Trunk Account)
- The default is for this option to be disabled (“no diversion-supported”).

[Web]: Added ability to select ADTRAN IP 7XX Phone Firmware

Issue Detail

- Under IP Phone Configs and Boot Settings, on Default Firmware tab you can now select the phone firmware images (AppName and BootName) from a drop-down list for both the IP 706 and IP 712 phones. Previously, this was changed by editing the adtran_firmware_706.txt and adtran_firmware_712.txt files in the ADTRAN folder on CFLASH.

Summary of Enhancements

This section highlights the enhancements for AOS A2.03.00.

[Web]: Dial String Source can be configured in the web interface

Issue Detail

- The Dial String Source can now be configured in the web interface on a per-Trunk Account basis.
- The choices are:
  - Request URI
  - To Header

[Web]: All Special Event Actions are now available in AA Dial By Extension

Issue Detail

- Within Dial by Extension Details in the GUI, you can define Special Event Actions for when the Caller presses *, #, or a timeout occurs. Now, all 3 of the following actions are available for each of these special events.
  - Return to Attendant Menu
  - Transfer to Operator
  - Dial Collected Digits
- Adding all 3 actions to all 3 Special Events allows for a caller to press # at the end of a dialed extension in the Auto Attendant Dial-By-Extension action and act on the collected digits.

Summary of Bug Fixes

This section highlights major bug fixes in AOS version A2.03.00.SC

Possible T.38 fax negotiation issue

Issue Detail

- If the time between a CED indicator packet and a V21 preamble packet was greater then 3 seconds, the NV7000 product would spoof a V.21 CRP. This could have caused a fax call to fail. A change has been made to prevent the V.21 CRP. This issue has been addressed.

This section highlights major bug fixes in AOS version A2.03.00.

Reboot with the AOS Fast Forwarding Engine

Issue Detail

- Internal caching issue exposed by VRF and packet fragmentation.

Reboot when receiving ISDN call with Caller ID name but no number

Issue Detail

- If a call is received from an ISDN PRI Trunk Account with Calling Party Name given, but no Calling Party Number the system may reset.

Reboot when CFLASH is corrupt

Issue Detail

- If a CFLASH card becomes corrupt beyond repair by the system, the unit might not boot from the backup image on NONVOL storage.

Reboot possible when performing VM to email conversion

Issue Detail

- If the VM message to be converted to an email attachment failed to open, a reboot could occur. Verified file could be opened before decoding contents.

Reboot when removing a “cross-connect” command

Issue Detail

- When the command “no cross-connect x” was issued to remove a cross-connect from an ATM interface to a PPPoE interface, the unit would reset.

Reboot after multiple transfers with ANI substitution

Issue Detail

- If ANI substitution was defined to change the incoming number before sending a call to phones and the call went through a Call Coverage several times, the system could reset. ANI substitution was being applied on each transfer via Call Coverage. Now, the ANI substitution is only applied once per call.

Delay in outbound audio when calling a Ring Group/Operator Group

Issue Detail

- This mainly occurs with Ring Groups that are “Ring All” type.
- Outbound audio to the caller, such as the initial greeting (“Hello, this is…”), can be lost when the phone answering the call is processing Status Group updates. The Status Group notification method has been streamlined to reduce this delay. Other methods to reduce this delay further are to configure the Status Group on the Ring Group phone so it doesn’t monitor other Ring Group members or modify the Ring Group type to UCD or Linear.

Voicemail Setup Wizard fails to set voicemail password with encryption

Issue Detail

- With "service password-encryption" enabled, the Voicemail Setup Wizard would not set the Voicemail Password correctly and the new user could not login to Voicemail.

Invalid character sent to phone could cause the phone display to freeze

Issue Detail

- Accented characters, such as those in French, were being interpreted as extended ASCII characters and sent in the SIP To: and From: fields to IP Phones. This was not valid operation per the SIP RFC. The To: and From: field values will now be truncated at the invalid character.

One-way audio when a G.711 call follows coverage to Voicemail

Issue Detail

- If a call using the G.711 codec is sent to a User Account and the Call Coverage is to Voicemail or Auto Attendant the call will connect but outbound audio will not be present.

Lost audio on a call to Voicemail or Auto Attendant over a SIP Trunk

Issue Detail

- When audio from Voicemail or an Auto Attendant is played out a SIP trunk using G.729, periods of outbound audio are lost on the PSTN side of the call.

Configured outbound proxy port incorrectly affects SIP headers

Issue Detail

- When the configured outbound proxy uses a different port than the SIP server, the system incorrectly used the outbound proxy port in the URI, From, and To headers instead of just for the IP header.

System Speed Dial numbers limited to 13 digits

Issue Detail

- When configuring a Speed Dial number you can enter the full number using the command "voice speed-dial nn <number> <name>" but the "<number>" would be truncated to 13 digits.

Error in contents of Voicemail notification email with French-Canadian

Issue Detail

- The contents of the email notification would contain an error regarding the phrases used. This was due to the French-Canadian prompt definitions not being updated correctly. These have been updated for AOS A2.03.00 and are also available for download at http://www.adtran.com/support .

The "voice prompt-language" command may not be preserved on reboot

Issue Detail

- If Latin American Spanish was selected, the prompts would operate correctly until the system was reset. Then it would return to the English language.

[Web]: Reboot when selecting ETSI 300 ISDN Switch Types

Issue Detail

- Selections for ETSI 300 102 and ETSI 403 ISDN Switch Types are available in the web interface configuration for the PRI interface. These selections are not supported and choosing them could cause the system to reset.

[Web]: 503 Server Error on System Parameters page

Issue Detail

- When clicking Apply on the System Parameters page, a 503 Server Error would be reported to the browser.

[Web]: 503 Server Error on SIP Server, SIP Proxy, and VoIP Settings pages

Issue Detail

- When browsing to these pages the browser would report a 503 Server Error and the pages would not load.

[Web]: 503 Server Error on Trunk Accounts page

Issue Detail

- When browsing to these pages the browser would report a 503 Server Error and the pages would not load if ANI or DNIS Substitution entries were configured on this Trunk Account.

[Web]: Symbols in a Line Label create a parsing error

Issue Detail

- If a symbol, for example “&” was used in an IP Phone Config as a Line Label, a parsing error would occur when trying to load the config file from the web interface of the NetVanta 7000 Series Products.

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 A2.03.00

AOS A2.03.00 is a Controlled Release AOS. Contact your local ADTRAN Sales Representative or ADTRAN Technical Support to obtain 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-A2-03-00-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 A2.03.00, an upgrade to the Bootcode is not required.

Appendix A – Errata for A2.03.00

The following is a list of errata that still exists in A2.03.00.

Possible reboot during heavy voicemail load

Issue Detail

- Under very stressful voicemail loads, the unit may 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.

Errata Justification

- These conditions are not very commonly found. This issue will be addressed in a future release of AOS.

SPRE *34 fails to forward remote user account

Issue Detail

- The user account entered in the SPRE code does not get forwarded, but instead the user account associated with the phone that dialed the SPRE code gets forwarded.
- 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.

Errata Justification

- 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 can remain busy.

Errata Justification

- A timeout value will be implemented for the paging port in a future release of AOS.

Phone display locks 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 would result in the phone display showing a call ringing when it has been cleared from the trunk side.

Errata Justification

- This will be addressed in future versions of phone firmware and AOS

Congestion may cause loss of encrypted packets for VPN

Issue Detail

- Under heavy load, the unit cannot service packets at the same rate at which they need to be encrypted for VPN tunnels. This causes the unit to drop packets. Also, input decryption errors are reported to the terminal due to encrypted packets missing in the sequence. Throughput performance is slightly affected.

Errata Justification

- This does not occur in all cases and is expected to be addressed in a future release of AOS.

Outgoing mail messages may be rejected due to formatting

Issue Detail

- When using Email Logging, outgoing messages may be sent with Line Feeds without a Carriage Return. Some SMTP servers will reject the messages with an error.

Errata Justification

- This will be resolved in a future AOS release.

Calling Party Number Presentation Restricted not honored by Voicemail

Issue Detail

- When a call is received on a PRI trunk with the Presentation variable 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.

Errata Justification

- This will be resolved in a future AOS release.

Call to check voicemail may not clear

Issue Detail

- This condition can occur when a user calls to check voicemail and the call terminates abnormally. If the user is configured to authenticate for Voicemail with password only, the call remains active even if it has cleared from the phone (by reboot, etc). The user can then no longer access voicemail to retrieve messages.

Errata Justification

- If the default method of authentication is configured (using mailbox number and password), the call will terminate after the menu repeats 10 times. This will be resolved in a future AOS release.

SIP phones will not ring forever if part of a Ring Group

Issue Detail

- Phones that are members of a Ring Group will ring for the maximum time defined for the User Account associated with the phones instead of the configured number of rings on the Ring Group account (num-rings 0 is for unlimited rings).

Errata Justification

- This is only a limitation of SIP phones, analog phones will ring continually. This issue will be addressed in a future release of AOS. In addition to changes in the AOS, another requirement to make this work for Polycom phones is to set “call.offeringTimeOut=0” in the “customer-sip.cfg” via the IP Phone Configs web page.

*61 SPRE Does Not Work for International Numbers

Issue Detail

- When adding a speed dial with *61 SPRE, adding an international number will cause the system to return an error (484 Address Incomplete).

Errata Justification

- The speed dial entry can be added from the CLI.

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.

Errata Justification

- This only occurs with the following scenario and a Polycom phone. ADTRAN phones do not have the issue. This will be addressed in a future release of AOS or in an updated version of certified Polycom firmware.

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.

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 has Caller ID and is transferred, the Caller ID number is changed based on what is configured for “X” in the command.

Errata Justification

- This will be resolved in a future AOS release.

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 are stored they are truncated to 13 digits. This affects speed dial entries for international numbers.

Errata Justification

- Speed Dial entries can be configured on the user’s phone instead. This will be resolved in a future AOS release.

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 will allow this function and the result is that the SPRE code to park the call is processed out the Trunk Account for the SLA.

Errata Justification

- In a future AOS release, the attempt to park the call will be captured and rejected without being sent out the trunk.

[Web]: Error when viewing Trunk Account config page

Issue Detail

- If a SIP Trunk Account has ANI or DNIS Substitution configured, those entries cannot be re-ordered or deleted from that page.

Errata Justification

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

[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 truncates the Display Name to 15 characters.

Errata Justification

- The config file can be modified manually to correct this issue. This will be resolved in a future AOS release.

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

Issue Detail

- On the detailed Phone Config page, you can configure no more than 5 secondary lines in one phone config. It is valid in some cases to configure more but this is limited by the web interface.

Errata Justification

- The config file can be modified manually to correct this issue. This will be resolved in a future AOS release.

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

Custom Trunk Group Access Configuration Guide -
http://kb.adtran.com/article.asp?article=3016&p=2

- Guides you through configuring the system to allow users to dial “7” instead of “9” as the preceding digit.

Configuring the NetVanta 7000 Series for a Bandwidth.com SIP Trunk -
http://kb.adtran.com/article.asp?article=3057&p=2

- Provides configuration examples specifically for use with SIP Trunks from Bandwidth.com (http://www.bandwidth.com).

Configuring the NetVanta 7000 Series for a COVAD SIP Trunk -
http://kb.adtran.com/article.asp?article=3036&p=2

- Provides configuration examples specifically for use with SIP Trunks from COVAD (http://www.covad.com).

Configuring AOS Devices for Forwarding Calls out SIP Trunks -
http://kb.adtran.com/article.asp?article=3045&p=2

- Shows how to configure Trunk Accounts so that calls that are forwarded out SIP Trunks are accepted by service providers.

Configuring SIP Networking over VPN in AOS -
http://kb.adtran.com/article.asp?article=3046&p=2

- Outlines the configuration of SIP Networking via a VPN & GRE Tunnel.

Configuring and Using Hands Free Auto-Answer Intercom -
http://kb.adtran.com/article.asp?article=3051&p=2

- Outlines the configuration and operation of hands-free intercom features for use phone-to-phone

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