RELEASE NOTES

NetVanta 7000 Series Products
AOS version R10.5.1
January 29, 2013

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Introduction

AOS version R10.5.1 is a maintenance release that addresses customer issues that were uncovered in previous code releases.

This release is generally available code. Results obtained during internal testing have been evaluated and the code has been determined to be ready for general availability. Caveats discovered during testing but not addressed in this build are listed in *Errata on page 10*.

A list of new or updated documents for this release appears in *Documentation Updates on page 13*.

Configuration guides, white papers, data sheets, and other documentation can be found in the ADTRAN Support Community, https://supportforums.adtran.com. The contents of these release notes will focus on ADTRAN's IP telephony products.

Supported Platforms

The following platforms are supported in AOS version R10.5.1.

- NetVanta 7100 – IP Communication Platform
- NetVanta 7060 – IP PBX

For a list of the software and firmware requirements, refer to the table in *Minimum Software or Firmware Summary on page 6*.

To confirm the Boot ROM version of the ADTRAN unit, telnet or console to the unit and issue the **show version** command. In the command output, the Boot ROM version will be listed as **Boot ROM version XX.XX.XX**. If you require a Boot ROM upgrade, please contact ADTRAN Technical Support (support@adtran.com or 888-423-8726) for assistance.

Hardware Requirements and Limitations

In an effort to maximize customer experience, whenever possible and applicable, ADTRAN will advertise the minimum hardware requirements for running the recommended software versions. While ADTRAN strives to support the newer software revisions on existing hardware, due to CPU, RAM, and other hardware limitations, it may not always be possible. In such instances, customers are advised to upgrade the hardware (including phones, NetVanta 7000 Series chassis, and accompanying networking gear) while upgrading their software, because performance issues and erratic behavior could cause certain product features to become nonfunctional. ADTRAN provides field advice whenever possible in these cases. Resellers and customers are advised to periodically check with ADTRAN Technical Support and field staff for these advisories, especially when upgrading to newer software revisions.

NetVanta 7100 Hardware

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

There have been two revisions of NetVanta 7100 hardware. These are denoted by different part numbers: 1200796L1 (older) and 1200796E1 (newer). Beginning with AOS release A2.04, ADTRAN does not recommend using newer AOS versions on the older 1200796L1 units. These units continue to be field worthy and would continue to perform as expected for their useful lifetime on software revisions prior to A2.04. However, due to differences in hardware, some or all of the new features might not be supported on the older hardware (1200796L1).

The 1200796L1 is explicitly NOT recommended for use for the following features or firmware releases:

- For any firmware release R10.x or higher
- Support for greater 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 NetVanta 7100 to perform 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 are no further known constraints for other features at this time, keep updated on any future advisory by ADTRAN. The recommended hardware for the AOS A2.05 and later features is 1200796E1. Contact your ADTRAN representative about the options available to you if you have a 1200796L1 unit, and want to use a newer release.

IP Phone Models

Beginning with release A4.x, the legacy Polycom phones (IP 430, IP 501, IP 601 and IP 4000) do not support all the features available in the current AOS and phone firmware releases. Customers could experience sluggish behavior on these older generation phones when used in conjunction with newer software releases. If you experience sluggish behavior after an upgrade, contact ADTRAN Technical Support for a solution. This could involve either upgrading the phone hardware (to the equivalent newer generation phone, such as IP 450, IP 550, IP 650, or IP 6000) or scaling back the feature load on the legacy phones.

Software Requirements and Limitations

This section defines the recommended firmware/software versions necessary for the related aspects of the NetVanta Unified Communications solution.

AOS Firmware Image Storage

AOS firmware images can be stored on flash/non-volatile random access memory (NVRAM) as well as on CompactFlash® memory. However, it is recommended that the primary firmware image be stored on flash/NONVOL and the backup firmware be stored on CompactFlash.

To copy the current image from flash/NVRAM to CompactFlash, use the **copy flash** <*filename*> **cflash** <*filename*> command.

Required AOS Bootcode Version

When upgrading to AOS version R10.5.1, an upgrade to bootcode version A2.06.B1.01 is required. Check the table in *Minimum Software or Firmware Summary on page 6* to verify you have the required minimum Boot ROM. Contact ADTRAN Technical Support for this bootcode version and instructions for loading it.

Minimum Software or Firmware Summary

Product or Phone Model	Minimum Software or Firmware	Minimum Boot ROM
NetVanta 7000 Series	A4.10 or later	A2.06.B1.01
NetVanta 6355/Total Access 900(e) Series	A2.06 or later	-
NetVanta UC Server (as part of BCS)	UCS 5.0.1	Not applicable
ADTRAN IP 706/IP 712 phones	R2.3.0	2.1.0
Polycom IP 321/IP 331 phones	3.2.7	4.1.2b
Polycom IP 335, IP 450, IP 550/560, IP 650/670, IP 5000, IP 6000, IP 7000 phones	3.2.7	4.1.2b
Legacy Polycom IP 430, IP 501, IP 601, IP 4000 phones	3.1.8	4.1.2b

These files can be downloaded from http://www.adtran.com/support, select **Software Downloads**, and choose the appropriate phone model from the **IP 700 Series**. Contact ADTRAN Post Sales Technical Support at (888) 423-8726 or email: support@adtran.com, if you are unable to download these files.

Important Notices

The following important notices are provided in addition to the previous *Supported Platforms*, *Hardware Requirements and Limitations*, and *Software Requirements and Limitations* sections to ensure successful deployment.

Upgrades to AOS version R10.2.0 and Later

Beginning with AOS version R10.2.0, the syntax of certain commands was modified from previous AOS versions (such as AOS A2.x, A4.x and A5.x) by either removing or adding the **ip** keyword. In general, when the **ip** keyword appears in a command, it signifies that the command is only applicable to IPv4 functionality. As more features introduce IPv6 support, the **ipv6** keyword is added to signify the command is only applicable to IPv6 functionality. The **ip** keyword has been removed from several commands to signify that the command has both IPv4 and IPv6 functionality.

Due to this syntax change, downgrading a NetVanta 7000 Series product configured in AOS version R10.2.0 or higher to a previous AOS version (such as AOS A2.x, A4.x and A5.x), could cause service disruption because the new syntax might not be recognized by the previous version. Upgrading a unit from an older AOS version to AOS version R10.2.0 or later will not cause service disruption because both the old and the new syntaxes are accepted. **It is recommended that a full copy (data and voice settings) of the configuration be saved prior to upgrading to AOS R10.2.0 and above**. This can be done from the **Utilities > Configuration** page in the GUI.

For more information on specific commands, refer to the *AOS Command Reference Guide* available at https://supportforums.adtran.com.

Please note that the NetVanta 7000 series does not support IPv6 at this time. If you envision needing any IPv6 features natively on the NetVanta 7000 series, then contact your ADTRAN representative with your request. In general, we recommend using an IPv6 capable ADTRAN router with the NetVanta 7000 series for any IPv6 features.

Default Firewall Configuration Changes

Changes were made to the default firewall configuration to increase security of voice platforms when connected to the Internet. These changes can impact remote phones and SIP trunking applications, but do not impact local phones on the NetVanta 7000 Series.

- In AOS versions A2.01.00 through A2.03.00.SC, the default Public access control policy (ACP) 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 ACP to allow SIP traffic from remote sites and SIP trunking providers.
- Units that were shipped with AOS versions through A2.03.00.SC contain a default configuration that allows inbound SIP traffic (destined for UDP port 5060). These configurations should be modified before deployment. Guidelines for this configuration are given in the *NetVanta 7000 Series Security Guide* available from the ADTRAN Support Community, https://supportforums.adtran.com.

Notice of Defined Voicemail File Limit

The NetVanta 7000 Series products can maintain a maximum of 3000 voicemails per system. The implementation of voicemail message expiration allows the system to remain within the defined limit. Upgrading the CompactFlash card to a larger card is not supported and will not result in more voicemail storage. Should you need to replace a failed CompactFlash card, contact ADTRAN Technical Support for assistance.

Updates to Web Interface Pages

On occasion, changes are made to web pages in the NetVanta 7000 Series web interface that may require files in the browser cache to be purged. This can be done in most browsers by deleting the browsing history or by pressing Ctrl-F5 in most cases.

System Notes

This section outlines known caveats for AOS version R10.5.1**.**

- The **match ani** command used for ANI substitution will match on the received ANI prior to any global ANI substitutions. The **match ani** command used for adding or substituting diversion headers will match on the modified ANI after the global ANI substitutions are applied.
- During conferences that use the conference bridge in UC Server, when one member in a conference places the call on hold, music may stream to all members that have joined the conference.
- Caller ID does not display on pickup *52xxxx*.
- The Personal Phone Manager's User Status monitoring list may return the list from the previous user's browser session if more than one user shares the desktop browser.
  The work around is to delete all cookies and restart the browser.
- Calls with caller IDs that contain special characters can be disconnected when placed on hold by an Advatel IP Console.
- Adding a T1/E1 link to an existing Multilink PPP bundle using the GUI causes the PPP link to bounce when applied. The PPP link will go down and immediately recover; however, some packets could be lost. To work around this issue, a T1/E1 can be added using the CLI, and the link will stay up while the addition is applied.

- Calls using the G.729 CODEC are limited to 25 calls for E1 PRI.
- FindMe-FollowMe treats all calls from the auto attendant as internal calls.
- SNOM M3 phones do not support attended transfer at this time. This and other caveats will be documented in a future configuration guide for using the SNOM phones with the NetVanta 7000 Series.

Features and Enhancements

This section highlights the major features, commands, and behavioral changes for AOS Version R10.5.0

- Added support for two E1/PRIs along with a SIP trunk. One PRI connects to the PSTN and recovers timing, and the other connect to a PBX and sources timing.
- Added the option for PRI calls to automatically append a # at the end of the dial string, and mark the information element Screen_indicator in the Called number as Network based.
- Added support for passing # as a valid dialed digit from a phone user to a PRI trunk.

Fixes

This section highlights major bug fixes in AOS version R10.5.1

- Clear text password is displayed in the event log
- On an AOS unit acting as a DNS proxy, the unit could reboot when a client attempted to resolve a domain name and the DNS servers could not reached.
- Sending a DNS query for an empty host name caused the AOS unit to reboot.
- In rare cases, a PRI interface will remain in a link down state if the T1 inteface link state goes down and the back up in a short period of time.
- When using a PRI network role, if the ISDN T303 timer expired, a reboot occurred.
- The backup python script did not present a progress indicator while running.
- If a unit configured with the **sip-server rollover service-unavailable-or-timeout** command received a 503 Service Unavailable response to a SIP REGISTER message, no additional registrar servers would be contacted.
- Default Polycom configuration files were not placed in the CompactFlash root directory upon boot.
- SCAs indicated that a call was active when there was not an active call on the line.
- Python backup script instructions were difficult to read and follow.
- For any device that did not support Multi-VRF for SIP (i.e., NetVanta 7000 Series, NetVanta 3100, and NetVanta 600 Series), SIP access classes would block all SIP traffic.
- Issuing the command **no voice num-rings** did not successfully restore the default value of 4 on the NetVanta 7100.
- Received Allow-Events headers that improperly used semicolons as a delimiter were not being properly corrected.
- The output of the **show run interface pri** <*slot/port*> command did not display the administrative state for the interface.

- If a reINVITE was received shortly or immediately after the ACK for the initial INVITE, the ADTRAN unit would respond with a 491 Request Pending. This caused a delay in the connection of two-way audio.
- The ADTRAN unit rebooted if VQM was disabled while a call was active on the system.
- When using FindMe-FollowMe to ring two local extensions in succession and then an external number, the call resulted in no audio when answered on the external number.
- The match statement for DNIS or ANI substitution considers [0-9] as 0123456789 which caused strings to exceed the maximum length.
- Users were unable to use click-to-dial from the Personal Phone Manager Caller ID List.
- Python backup scripts did not function reliably when the firewall was enabled.
- Changing the clock/timing source on a T1 VIM caused an exception report to be generated the unit was rebooted.
- The ring group help text for the **max-inbound** command implied incorrectly that 9 is the maximum number of calls when 10 was correct.
- SIP syntax error events were logged even if they were automatically corrected by the unit.

This section highlights major bug fixes in AOS version R10.5.0.

- Port Authentication would not work on the NetVanta 7100.
- The Session ID included in the SDP for SIP messaging sent to IP phones did not always increment properly.
- An attended transfer to a ring group extension caused a DSP resource leak.
- In rare cases, it was possible for the ADTRAN unit to reboot if an INVITE with a malformed replaces header was received.
- Changes made to boot settings on the IP Phone Globals > Boot Settings GUI menu caused an error if the customer was previously running R10.2 code and upgraded to R10.3+.
- An inbound call on a user role PRI could fail if caller ID name was received after the initial ISDN Setup message.
- When editing a phone configuration, the wrench column was too large, which obscured the ability to add line keys.
- When media anchoring was configured and the group ring type was set to Executive Ring, calling the ring group from a SIP phone resulted in one-way audio when the executive member answered.
- A reboot would occur when all of the following conditions were met:
    - A received SIP message (most likely an INVITE) contained a Diversion header.
    - The user portion of URI in the Diversion header was in E.164 format.
    - The country code in the Diversion user did not match the one configured on the unit.
    - The length of the Diversion user (including the '+') was less than 16, and the combined length of the configured IDD prefix and the diversion user (excluding the '+') was greater than or equal to 16.
- The incorrect CODEC was chosen for FindMe-FollowMe calls when the user was set to provide ringback. This did not occur when using only CODEC G.711.
- The Switchboard was unable to find a target for call routing, causing a reboot.

- The **ip sip grammar from user international** command did not change the FROM field in the SIP header to the E.164 format.
- The **polycomftp** factory default username was placed in a portal list named **phones** to restrict access to FTP logins only.
- Modifying a user on the GUI resulted in two duplicate User successfully updated messages.
- When VQM was enabled, there was no audio for external calls on a simple remote phone configuration.
- On the GUI DHCP Config > Numbered Options tab, the Value field was not completely visible.
- Using pickup groups with media anchoring resulted in one way audio.
- Deleting all voicemails in a user account using the PPM was very slow and would not completely empty the voice mailbox.
- Using the PPM to delete a large number of voicemails resulted in a error dialog box that persisted.
- The NetVanta 7100 failed to send a NOTIFY to the Advatel IP console after a status group subscription if the group contained 32 members.
- When media anchoring was configured and the group ring type was set to Executive Ring, calling the ring group from a SIP phone resulted in one-way audio when the executive member answered.
- It was not possible from the GUI to set the number of rings for a user with a call coverage list.
- In FindMe-FollowMe actions, where the accept option was available (enabled) but the number of seconds before disconnect was set to a low value, the call disconnected before the user could hear the caller's name or have a chance to press 1 to accept the call.
- Modem tone detection did not function on ring group calls.

Errata

The following is a list of errata that still exist in AOS version R10.5.1**.**

- The unit's FTP server does not respond properly to a LIST command with the required carriage-return/line-feed formatting. This results in certain FTP clients failing to connect.
- The NetVanta 7100 and NetVanta 6355 platforms fail to reset QoS map statistics for applied QoS maps when the **clear counters** command is executed.
- If **ip mcast-stub fixed** is removed from an interface configuration which also has an IGMP static group configured, the mroute is not deleted as it should be. This only occurs when static groups are configured. It does not occur with dynamically learned groups.
- Users may receive a 503 Service Unavailable response when adding or removing a voice user from a CODEC group in the GUI. The add or remove operation will have been successful, therefore this issue is cosmetic.
- When adding a new User Account, the GUI may incorrectly display an FXS port as available when it is already assigned.
- Call duration in **show voice call summary active** command output is reset after receiving a SIP reINVITE.
- The CLI command **no description** does not successfully remove a description from a ring group configuration.
- Inbound calls from a Megapath (Broadsoft) SIP trunk fail to be delivered by FindMe-FollowMe to external number. Calls roll to next Call Coverage item when answered by the external number.

- When configuring Call Queues via the CLI, if an attempt is made to configure more than the maximum number of queues, an error will be shown. Following this error, no configuration commands can be entered on other queues until the configuration mode is exited.
- Calls into a ring group that has a simple remote phone user as a member may have no audio.
- Hair pinning calls may have no audio after a transfer.
- SLAs will incorrectly briefly transition to idle before returning to the active state when a call is disconnected locally.
- Voicemail Operator Assist on a ring group dials 0 even when configured with a different value.
- In IE9, when trying to apply a user account change, a database error will be displayed.
- Inbound SIP calls fail when the command **max-number-calls** <*value*> has been set.
- FindMe-FollowMe may not properly populate the SDP in either the INVITE or ACK when using SIP PSTN trunks.
- When a simple remote phone user places a call destined to a either ring group, FindMe-FollowMe, or a call queue, and a local phone user answers, the local phone will not be able to hear the remote user.
- Transport=TCP is incorrectly included in the Contact header on a UDP SIP trunk.
- When using FindMe-FollowMe to contact two external numbers simultaneously, one-way audio will occur when one of the external number answers.
- Forwarded voicemail messages may report a date one month prior to the actual date of the message.
- An external call from an analog phone may result in choppy hold music when the caller places the call on hold.
- Voice quality can be degraded when all 23 channels on a PRI are in use.
- Analog phones that are members of a ring group may have no audio on ring group calls.
- If a call proceeds through FindMe-FollowMe to an Auto Attendant and is then transferred to another number, the unit will reboot.
- Adding a description to a status group on the Status Groups GUI menu may result in a 503 Service Unavailable response.
- Issuing the **factory-default** command writes default values to the startup configuration, but it does not ensure that the current boot configuration is reset to the startup configuration.
- Systems with greater than 16 simultaneous G.729 encoded SIP calls to a PRI trunk may experience voice quality degradation. It is recommended that customers who require greater 16 simultaneous SIP to PRI trunk calls configure the system to use G.711 encoding which is not affected.
- .SipPhoneInfoDatabase is not included in the MSP backup of NV7100
- SLA accept/reject templates do not affect calls sent using the SLA.
- SIP trunk calls may not hear ring back when calling into an Auto Attendant and reaching a ring group with a linear hunt type. Ring back may stop after the second phone of the ring group is presented the call. If the ring group is called directly (not through an Auto Attendant) or if the group is configured as ring-all type this will not occur.
- When call coverage is set to internal, it will still allow calls to be routed out an external trunk.
- During an internal SIP-to-SIP call, if the caller places the called party on hold, and the called party places the caller on hold, and then the caller takes the called party off hold, both parties will have no audio.
- Single Number Reach may fail to detect fax tones from certain fax machine models.

- A call placed to a remote user that that uses G.711 U-law or G.711 A-law will result in one-way audio if the call is routed out a trunk containing a CODEC list.
- The Update Directories action in the GUI does not properly update the directories for individual Polycom phones.
- FindMe-FollowMe fails with the Single Number Reach service in the NetVanta BCS.
- After upgrading from A4.X or A5.X to R10.2 or later, the GUI does not highlight the IP phone to indicate that the Polycom IP 550/560 phone configuration needs to be updated.
- Paging group calls fail when calling from a VVX 500.
- When Creating a New User in the GUI, DID numbers and aliases are not saved if the **Edit Config** button is pressed followed by the **Apply** button.
- When a voice user is configured for an empty caller ID number, the name is not being transmitted either.
- Polycom IP 320/321/330 phones do not display the users extension.
- In the GUI, when configuring the voicemail notification schedule for a user, the times Midnight and Noon are not listed in the correct order.
- Configuring a user to have Dialtone Only message waiting does not result in a SIP NOTIFY to SIP endpoints when a new message is waiting.
- Creating a new phone configuration results in an inapplicable sync dialog.
- The Polycom conference split feature does not function for Shared Line Appearances. The split can appear to be functioning, but there may be no talk path.
- When local packet capturing completes and while it is being exported, the voice quality may be adversely affected.
- The Polycom configuration files, sip_31x.cfg and sip_32x.cfg, are not up to date with the latest files provided by Polycom.
- Call Coverage incorrectly states that it goes to Busy for some System Modes.
- If a local user calls another local user with FindMe-FollowMe enabled, and the call is forwarded over a SIP trunk to a Metaswitch, the forwarded call may result in one-way audio.
- Inserting a CompactFlash card into an AOS unit while it is powered results in a reboot.
- A FindMe-FollowMe user, configured never to ring under call coverage, cannot be used as the destination to record prompts from the Audio Prompts menu on the NetVanta 7100 GUI.
- When using FindMe-FollowMe on a NetVanta 7100, internal calls forwarded to voicemail will function properly, but voicemails forwarded to an external server do not function properly. When Ringback Only is disabled on the NetVanta 7100, only the Refer the Call FindMe-FollowMe action can be used to direct inbound calls to a voice mailbox not located on that unit.
- When calling to a shared line appearance, audio may not be passed between the endpoints if one of them is behind NAT due to the RTP port being reported incorrectly in SDP to the phone. The issue can be avoided by not registering phones with shared appearances to an interface that is configured for NAT.
- In the VQM RTP Monitoring menu, the Source IPs and Interfaces menus have invisible data points that appear and display data when the cursor hovers over them. The invisible data point information duplicates a visible data point and can usually be found hidden above the visible data point.
- In the VQM RTP Monitoring menu, the refresh button duplicates information in the lower part of the menu. Also, when the cursor hovers over a data point, it displayed multiple instances of the same data.
- T.38 FAX calls fail after T1 PRI loss and system timing shifts. A reboot is required to clear the condition.

- When configuring call coverage, setting the Ring Extension to Never results in a three-second delay delivering voice traffic to the ADTRAN phone.

Upgrade Instructions

Upgrading ADTRAN products to the latest version of AOS firmware is explained in detail in the configuration guide *Upgrading Firmware in AOS*, available at https://supportforums.adtran.com. Firmware upgrades are available on the *Support/Software Downloads* section of the ADTRAN website at http://www.adtran.com.

Documentation Updates

The following documents were updated or newly released for AOS version R10.5.1 or later specifically for the AOS products. These documents can be found on ADTRAN's Support Forum available at https://supportforums.adtran.com. You can select the hyperlink below to be immediately redirected to the document.

- *AOS Voice International Configuration Guide*
- *Configuring Remote Phones with an AOS SIP Gateway*
- *Configuring Simple Remote Phones for the NetVanta 7000 Series*
- *Configuring SIP Trunking Gateway for Use with NetVanta ECS*
- *Configuring the NetVanta 7000 Series Personal Phone Manager*
- *Configuring User Accounts on the NetVanta 7000 Series*