RELEASE NOTES

NetVanta 7000 Series Products
AOS version R10.9.5
October 17, 2014

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Contents

Introduction *4*
Supported Platforms *4*
Hardware Requirements and Limitations *4*
Software Requirements and Limitations *6*
Important Notices *7*
System Notes *8*
Features and Enhancements *9*
Fixes *9*
Errata - System Management *10*
Errata - Call Control *12*
Errata - Audio *13*
Errata - Endpoint *13*
Upgrade Instructions *14*
Documentation Updates *14*

Introduction

AOS version R10.9.5 is a maintenance release that also 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 - System Management on page 10*.

A list of new or updated documents for this release appears in *Documentation Updates on page 14*.

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 R10.9.5.

- 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 Required for Interoperability 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.

ADTRAN branded VVX phones (model names ADTRAN VVX 300, ADTRAN VVX 310, ADTRAN VVX 400, ADTRAN VVX 410, ADTRAN VVX 500, and ADTRAN VVX 600) work with NetVanta 7000 series AOS release version R10.8.0 and beyond without requiring an additional license key purchase. The equivalent Polycom branded phones will not work with release R10.8.0. If you are currently using the equivalent Polycom branded phones with the NetVanta 7000 series, you will need to either remain on a pre-R10.8 release version or use the ADTRAN branded version of the VVX phones until a licensing mechanism can be added to allow the use of Polycom branded VVX models.

The rest of the Polycom family of supported IP end points continue to remain unaffected. Either an ADTRAN branded model or the equivalent Polycom branded models of these IP phones can be used with R10.8 and beyond. See the following table to determine AOS release R10.8 compatibility with ADTRAN and Polycom branded phone models.

Table 1. Release R10.8 Phone Compatibility

Model	Part #	Compatibility with AOS Release R10.8 and Beyond
ADTRAN VVX 300	1200853G1	Yes
ADTRAN VVX 310	1200853G1#GB	Yes
ADTRAN VVX 400	1200854G1	Yes
ADTRAN VVX 410	1200854G1#GB	Yes

ADTRAN VVX 500	1202856G1	Yes
ADTRAN VVX 600	1200856G1	Yes
Polycom branded VVX 300, 310, 400, 410, 500, and 600	Multiple	No. R10.7 is the last supported AOS version for these phones.
ADTRAN branded and equivalent Polycom branded SoundPoint IP 321, 331, 335, 450, 550, 560, and 670	Multiple	Yes. There are no restrictions when using these models.
ADTRAN branded and equivalent Polycom branded SoundStation IP 5000, 6000, and 7000.	Multiple	Yes. There are no restrictions when using these models.

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 R10.9.5, an upgrade to bootcode version A2.06.B1.01 is required. Check the table in *Minimum Software or Firmware Required for Interoperability 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 Required for Interoperability

Product or Phone Model	Minimum Software or Firmware	Minimum Boot ROM
Remote NetVanta 7000 Series (when networking to another 7000 series device)	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 R10.8.0 and Later

Beginning with AOS version R10.8.0, the syntax of certain commands was modified from previous AOS versions (such as AOS R10.5.x, R10.7.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 R10.8.0 or higher to a previous AOS version (such as AOS R10.5.x, R10.7.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 R10.8.0 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 R10.8.0 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

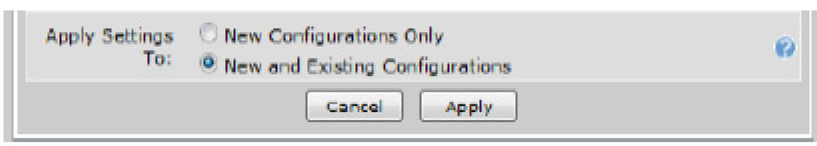
The NetVanta 7000 Series products can maintain a maximum of 3000 voicemails per system. The implementation of voicemail message expiration allows the system to remain within the defined limit. Upgrading the CompactFlash card to a larger card is not supported and will not result in more voicemail storage. 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.

Considerations Before Upgrading Related to SPRE Code Support for SLA

1. Local SPRE code dialing from an SLA requires phone dial plan changes. After upgrading to R10.6.0 software, newly created phone configurations will have the proper dial plan settings applied. For upgrade cases where SLA was already configured on an existing phone, the dial plans will be modified to support this new functionality. Please review the changes under the IP Phone configuration page and regenerate the phone configurations by using the admin login and browse to Voice>IP Phone Globals>Default Settings>, select “New and Existing Configurations” and select Apply.



2. SPRE code dialing from an SLA could interfere with existing configurations if SPRE codes were used on SLA’s prior to this release. Plese review your configuration to determine if SPRE codes were allowed prior to the upgrade (check SLA dial plans) and if so, you will need to configure the following command **voice spre-mode override <*xx>** using the appropriate codes in place of **xx**.

System Notes

This section outlines known caveats for AOS version R10.9.5.

- 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

No new features or enhancements were included in this release.

Fixes

This section highlights major bug fixes in AOS version R10.9.5.

- An analog call to a SIP user configured for Find Me-Follow Me (FMFM) would result in one way audio if Ringback Only and No Press to Accept were enabled.
- Applying a change to the IP Phone Globals page would reset all remote phones to local.
- An extension could be added that exactly matched an emergency services number. The system will now prevent the creation of such an extension when emergency services is enabled.
- When configured with a user role PRI, if the local exchange sent progress indicator #2 (PI2) to indicate the presence of inband audible ringback on a SIP to PRI call, a 183 Session Progress with SDP was not sent on the SIP call leg.
- Inbound calls in the call queue with a caller ID of Unknown could not be answered by queue members
- A FindMe-FollowMe external call could not be completed if there was a CODEC mismatch between the original and the external call. Transcoding will now occur to resolve this issue.
- The Update Directories action produced an error message when there were a large number of directory entries.
- The NetVanta 7000 Series allowed configuration of a forward to a number that already existed as an alias. This resulted in a looped call that occupied all system resources, and would eventually cause a reboot.
- Attempting to retrieve an active call on a SLA via the held-call-pickup number would reboot the NetVanta 7100.
- When modifying an existing SIP trunk using the GUI, the transport mode of that trunk was changed from UDP to TCP, breaking communication on that trunk.
- Attempting to retrieve a call using the park retrieval SPRE code on a Simple Remote Phone resulted in no audio if internal music on hold was configured.
- The Email Action on the VM Settings tab in the GUI could not be set for operator or ring groups.
- Users were able to bypass Forced Account Codes by dialing the external number without the trunk access code.

- When viewing the call queue Membership Status in the Person Phone Manager, the call queue sorting order was reset after updating or auto-refreshing the GUI menu.
- If **no notify-ringing** was set for a status group, a NOTIFY message would still be sent when call queue members were receiving a call.
- When using Internet Explorer 9 or earlier, the Apply button on the Voice > IP Phone Globals > Default Settings tab would not function.
- Assisted transfers over a SIP trunk to a call queue resulted in a disconnect.
- Intermittent inbound calls on SLA lines would ring indefinitely without the switchboard routing the call.
- Hairpinned calls failed to have audio after a transfer.
- Transferring a call from an external SIP trunk back out the same trunk via an auto attendant action resulted in a failure to populate the SDP portion of the SIP message in either the INVITE or the ACK message.

This section highlights major bug fixes in AOS version R10.9.4.

- It was not possible to configure **ip rtp udp** with a range that included UDP ports 63018 and 63019.
- The inability of spanning tree to allocate packets for transmission resulted in a reboot.
- When creating a new user in the GUI, DID numbers and Aliases were not saved if the Edit Config button was pressed followed by the Apply button.
- When a user account was deleted it could not be removed from the global directory for Polycom phones.
- Incoming SIP trunk calls which were locally transferred would stop remote ringback during coverage.
- If a call using a 3.1 kHz audio bearer capability was received on a PRI, the AOS unit would not wait for calling party name to be sent in a FACILITY message after being instructed to do so in the SETUP message.
- Editing a SIP trunk that was configured for TCP in the GUI would change the transport to UDP.
- If an assisted transfer was performed to a virtual user with coverage configured to an external number that began with the Trunk Access Code digit, the system would incorrectly strip two digits before routing the call.
- GUI pages would not correctly load when TCP port 556 was used for the connection to the HTTP server.
- Upon a system reboot, dynamically added SIP registrations would be lost and not immediately added back.
- The GUI would incorrectly state that System Mode coverage could go to Busy, when in actuality it would use the Default Mode coverage.

This section highlights major bug fixes in AOS version R10.9.3.

- When the French Canadian language setting was configured in the GUI, the navigation links did not load correctly.
- When using the Enhanced ANI Substitution feature to add a Diversion header, two Diversion headers were added.
- When using an ISDN trunk, if the RELEASE_CMP for a previous call was received after the SETUP for a new call but before the new call connected, the new call would fail.

- When configuring call queues using the CLI, if an attempt was made to configure more than the maximum number of queues, an error was shown and no further configuration commands could be entered.
- The CLI command **no description** could not be used to remove a description from a ring group configuration.
- Transcoding failed for calls to voicemail that were received on an ISDN trunk.
- In certain scenarios, external calls going to a voicemail account did not generate an SMDR log to indicate the call had ended.
- If auto-link was configured, but the configured auto-link target could not be contacted, the AOS device would eventually become unresponsive.
- Call queue prompt audio was sent from the wrong port after a call was transferred from an auto attendant.
- Park Retrieval from a VVX 500 was displayed as being on hold.
- Removing a VIM would not correctly generate an exception report.
- When NetVanta 7100 was running NetVanta 6355 firmware, the System Summary GUI menu did not display.
- When NetVanta 7100 was running NetVanta 6355 firmware, the GUI displayed the platform as unknown.

This section highlights major bug fixes in AOS version R10.9.2.

- The installation wizard would incorrectly disable DHCP in the Polycom phone configuration files.
- In a transfer scenario, if a SIP Refer was received with a null Called Party Number a reboot would occur.
- Adding a description to a status group on the status groups page of the web GUI may have resulted in a 503 Server Error.
- An error was presented when adding a SIP Identity (or alias) to a User Account that began with the Trunk Access Code (TAC).
- DNS queries created by an AOS device could be sent using source port UDP 4500, which could have prevented responses from being properly received if "ip crypto" is enabled.
- Email notification could have failed when TLS was required by the mail server.
- Attempts to ring external numbers through FindMe-FollowMe would fail and disconnect the caller if the press-to-accept option was disabled.
- Monitoring user status in the Personal Phone Manger could have caused high CPU conditions.
- Assigning the language on the Default Settings tab under IP Phone Globals GUI menu would not change the language for Polycom SoundPoint IP phones.

This section highlights major bug fixes in AOS version R10.9.0.

- The GUI IP Phone Configs Update Directories command would only merge the first 21 system directory entries.
- The **factory-default** command would not change the Primary System Configuration file to the startup configuration if it had changed.
- Manually editing phone configuration files resulted in the phones disappearing from the IP Phone Configs GUI page.
- Paging group calls originating from ADTRAN IP 7XX phones had no audio.

- When using auto-link over HTTPS in AOS R10.8.0, check-ins would fail.
- Forwarded voicemail messages reported a date one month previous from the actual date of the message.
- Incoming SIP trunk calls that were locally transferred would stop remote ringback during any subsequent coverage.
- A 503 Service Unavailable response was received when attempting to edit a trunk account.
- The NetVanta 7100 rebooted when attempting a transfer during heavy call loads.
- Hot desking configuration files only supported the use of the default VLAN 2 IP address, 10.10.20.1.
- Click-to-Dial calls in a NetVanta BCS configuration were not being dialed.
- TFTP client connections caused memory to be lost and not recovered.
- Attempting to use a pickup group to pick up a call queue resulted in the call being dropped.
- When an external call was transferred to an internal extension which was configured for coverage to an external number, the diversion header was missing when the call was routed back out to the external number.
- An inbound ring group call that was cleared too quickly resulted in a reboot.
- When Music-on-Hold was set to external, Polycom IP phones were not able to blind transfer a call.
- Under heavy call load it was possible that successive calls could use the same memory location to store their B-channel information, which caused a reboot when the latter call disconnected.
- Enabling VQM caused audio to be lost when using the Simple Remote Phone feature.

Errata - System Management

The following is a list of System Management errata that still exist in AOS version R10.9.5.

- Simultaneously upgrading firmware on multiple VVX phones may cause the system to run out of memory and reboot. Workaround: Stagger the rebooting of the phones when upgrading phone firmware
- When a description is configured on an FXO interface, the progress dialogs of the ERL tool do not function correctly even though the ERL tool runs successfully.
- The Shared Line Accounts GUI menu incorrectly displays a configuration option for Codec Outbound Group. This is not a valid configuration option.

Errata - Call Control

The following is a list of Call Control errata that still exist in AOS version R10.9.5

- Successive reINVITE SIP messages to place a call on hold will be rejected with a 400 Bad Request response if incoming music on hold is enabled on the SIP trunk.
- Inbound calls from Megapath (Broadsoft) SIP trunks fail to be delivered by FindMe-FollowMe to external numbers. Calls roll to next Call Coverage item after being answered at the external number. **Workaround:** Enable Ringback Only and disable the Accept option in the FindMe-FollowMe configuration for the call to external party to be successful.
- Call coverage set to internal will still allow calls to be routed out an external trunk.
- Caller ID may not be correctly sent when an SLA/SCA call is transferred to an extension.
- transport=TCP is incorrectly included in the Contact header on a UDP SIP trunk.

- FindMe-FollowMe fails with Single Number Reach service in NetVanta BCS.
- When a voice user is configured for an empty caller ID number, the name is also not transmitted.
- When configuring call coverage, setting the Ring Extension to Never results in a three-second delay delivering voice traffic to the ADTRAN phone.
- SLA accept/reject templates do not affect calls sent using the SLA.
- T.38 fax call tests fail after T1 PRI loss and system timing shifts. **Workaround:** A reboot is required to clear the condition.

Errata - Audio

The following is a list of Audio errata that still exist in AOS version R10.9.5

- An external call from an analog phone may result in choppy hold music when placed on hold by the analog phone.
- A received SLA call that is answered and then attended transferred to a remote party will have one-way audio.
- If a Simple Remote Phone calls a user with FindMe-FollowMe configured as **ringback only**, **ring external**, and **press to accept**, the remote phone user will not hear audio. **Workaround:** Disable **ringback-only** in the FindMe-FollowMe configuration for the called user.
- When local packet capture completes and while it is being exported, voice quality may be adversely affected.
- During an internal SIP-to-SIP call, if the caller places the called party on hold, then the called party places the caller on hold, when the caller retrieves the called party from hold both parties will experience no audio.
- If a SIP is extension is blind transferred out an analog FXO trunk, noise is sometimes introduced in the audio.
- A call placed to a remote user that that uses G.711 Ulaw or G.711 Alaw will result in one-way audio if the call is routed out a trunk that contains a CODEC list.

Errata - Endpoint

The following is a list of Endpoint errata that still exist in AOS version R10.9.5

- A user account already registered (with a static registration or through the hot desk feature) to an IP phone may log into another hot desk phone. This causes the new hot desk phone to become the *active* phone for that user account. The original IP phone is no longer registered to that user account.
- Bria soft phones registered through a SIP security port always display in the Suspect list.

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 R10.9.5 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 Command Reference Guide*