RELEASE NOTES

NetVanta 7000 Series Products
AOS version R10.3.1
September 24, 2012

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To the Holder of the Manual

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Introduction

AOS version R10.3.1 is a generally available feature release that also addresses customer issues that were uncovered in previous code releases. It has been subjected to both design verification and product qualification testing. Results obtained during this 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 15*.

A list of new or updated documents for this release appears in *Documentation Updates on page 17*. 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 the ADTRAN's IP telephony products.

Supported Platforms

The following platforms are supported in AOS version R10.3.1.

- 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 R10.3.1, 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	R1.3.16	1.3.12
Polycom IP 321/IP 331 phones	3.1.3c	4.1.2b
Polycom IP 335, IP 450, IP 550/560, IP 650/670, IP 5000, IP 6000, IP 7000 phones	3.1.3b	4.1.2b
Legacy Polycom IP 430, IP 501, IP 601, IP 4000 phones	3.1.7	-

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 R10.3.1**.**

- 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.
- IP 700 Series phones will not play the ringback tone when they receive a 180 Ringing response after a 183 Session Progress response.

- 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 R10.3.0.

- Added a configurable simple remote phone support option in the GUI that can be configured on a per user basis.
- Added the ability for a call queue member to log in and out of a call queue using the Personal Phone Manager GUI. Also added was the ability for a call queue member to check the login status of other members using the Personal Phone Manager.
- Added the ability to add local emergency numbers for international applications based on the configured country code.
- Added a refresh button on the existing Caller ID and the Dialed Number tabs in the Personal Phone Manager GUI. The refresh retrieves the up to date caller ID and dialed number for the logged-in Personal Phone Manager extension.
- Added configurable support for Polycom SpectraLink Wi-Fi Series phone models 8440, 8450, and 8452.
- Added the ability to configure email actions on the Personal Phone Manager GUI.
- Added support for Polycom's **donotdisturb** configuration change that allows shared line appearances (SLA) to have a silent ring on a per registration basis. By default, all shared line appearances on Polycom phones ring. Adding the **donotdisturb** configuration to customer-sip.cfg file can enable all, some, or none of the Polycom phones to ring silently on SLA.
- Added a GUI menu (**Voice > Extension List**) to display all configured extensions including aliases and DID numbers. This allows an administrator to see all of the configured extensions in one location. The extensions list can be sorted by extension, name, and account type.

This section highlights the major features, commands, and behavioral changes for AOS Version R10.2.0.

- Enabled VLAN filtering by default for Polycom phones.
- Added configuration and phone support for Polycom phone model VVX-500.
- Improved the FindMe-FollowMe user's experience by allowing configuration of contact groups that will match unknown caller ID types and handle the calls as the user desires.
- Enabled the display of Call Queue Statistics in Personal Phone Manager for non-administrator users.

- Call flow handling of ring group calls was improved to minimize answer delays. A call to a ring group on the NetVanta 7000 Series typically goes through a couple of reINVITEs. Depending upon whether the ring group users are (a) on the NetVanta 7000 Series unit itself or (b) on a branch office product connected to the unit, the number of reINVITEs can add up and lead to delayed audio. This feature improves user's experience by eliminating or reducing these delay cases.
- Added the ability to inform the SIP user on which trunk a call was received using either a different ring tone or other distinguishing feature.
- Added support for administrative configuration of forced account code entry on a per user basis. This feature applies to SIP and analog endpoints and supports applications that require an account code entry for every call.
- Added the ability to transfer directly to Voicemail using SPRE code *86 plus the voice mailbox number. This feature improves the receptionist user's experience for transferring calls directly to voicemail.
- Added the administrative ability to email Call Queue statistics for the last 24 hours using CLI options. This enables administrative reports to be pushed rather than having to be checked periodically.
- Improved the Personal Phone Manager user's experience by organizing Caller ID and Dialed Number lists to place the most recent entry at the top.
- Added administrative configuration of FindMe-FollowMe for each user on the system. The number of simultaneous calls using FindMe-FollowMe remains limited to 10 in order to preserve system resources.
- Enhanced system VoIP security by using randomly generated SIP authorization password for IP phones.
- Added support for deployments in Ireland with localized system prompts.
- Added enhanced features to allow the NetVanta 7000 Series hardware to support using a NetVanta 6355 firmware image. This enhances customer options for blending and extending NetVanta 7000 Series deployments with other ADTRAN products and hosted VoIP.
- Added support for G.711 a-law for music on hold and system prompts. This feature enhancement improves international CODEC interoperability.
- Added administrative configuration of release timers for analog trunks. This new configurable support allows adjustments to call disconnect and trunk release timers for the specific application and country requirements.
- Added configuration support for multiple versions of Polycom phone firmware. This feature allows administrators to select the optimal firmware version for each Polycom phone model, based on their application.
- Added GUI support for enabling and disabling Ring Group Call Waiting.
- Added the ability to override the inbound Caller ID name/number when delivering FindMe-FollowMe to external coverage. This feature helps the called party quickly identify whether the call being received is a FindMe-FollowMe call.
- Added the **Press to Accept** option (by default) for FindMe-FollowMe when using analog trunks. This setting allows for a positive notification that a call was answered.

Fixes

This section highlights major bug fixes in AOS version R10.3.1**.**

- After disabling port mirroring to a particular port, a reboot was required before the port would receive packets.

- Units upgraded from another version of R10.x to R10.3.1 required an update of the sip_31x.cfg and sip_32x.cfg files. This is accomplished by deleting the two files in the Polycom folder on CompactFlash and then rebooting the unit. The unit will replace them upon reboot with the new versions.
- Added improved notification of how to synchronize the SIP Auth Password when adding an existing user to a phone configuration.
- The Polycom 550 and 650 would not display dialed digits from the idle screen.
- The IP Phone Config menu added multiple SLA keys when the user changed dial strings in IP Phone Globals menu.
- The SendVM button on Polycom phones did not function.
- DNS entries used by voice services were not being communicated properly to the VoIP Name Service. This caused those entries to not get refreshed properly before their TTL expired, which would result in the first call failing once the TTL expired.
- When VLAN filtering was set for VLAN ID 1 Polycom phones did not receive packets. A brief note was added to the GUI to explain the recommended settings to work with VLAN filtering.
- A 503 Service Unavailable response would be received when creating a Loop start RBS voice trunk.
- The second caller to the Call Queue extension would not ring any call queue member phones.
- Both the IP Phone Globals>Global Files>Polycom customer-sip.cfg and ADTRAN adtran_customer.txt files had Administrator Guide links that were out of date.
- After upgrading from A5 to R10.3.0, a new User Account/Phone Config could not be created because some files used in that process were not updated to R10.3.0 versions.
- For Auto Attendants, the Dial by extension digit action was automatically changed to Collect Digits regardless of how it was configured.
- Replication slot leaks or failures caused paging group calls to fail.
- Debugging status groups sometimes displayed garbled ASCII characters as output.
- Unneeded dial plan entries were contained in the factory default startup configuration that either: 1) overlapped with the new voice emergency-services list, or 2) were not applicable to the configured system country.
- In AOS R10.3.0, it was possible for SLAs/SCAs to enter into a state where the phones were not updated with the current status of the SLA/SCA. The result was that a phone with the SLA/SCA would show as idle when it was busy, calls on public hold would not show as available to be retrieved, and an idle SLA/SCA would show as busy.
- When FindMe-FollowMe was configured to ring the extension and an external number simultaneously, users were unable to deselect the check box for Accept which prompts the called party to press 1 to accept the call.
- After a caller in a call queue pressed a digit to be transferred out, a system reboot would occur.
- The help dialog stating the allowable range for the SNTP wait time was incorrect on the GUI.
- The command **ip sip user-matching require contact-match** did not function in R10.X.
- When the packet capture feature was in use, memory would leak.
- The emergency services list was displayed in the configuration and was removable, and the dial plan pattern ID help text and inputs were not being correctly checked.
- The user directory list was not building in the PPM GUI.

- Received SIP UPDATE messages were rejected with a 503 Service Unavailable response. The proper response was 200 OK with SDP.

This section highlights major bug fixes in AOS version R10.3**.**

- Setting a DHCP lease time of more than 9,867 days prevented the ADTRAN unit from storing the binding correctly.
- With password encryption enabled, BGP authentication passwords were encrypted a second time on reboot.
- Outbound route maps configured with the **set local-preference** command and advertised to iBGP peers were not backwardly compatible.
- DNS proxy would cause the pcconfig process to lock up which caused traffic to stop passing and the console to stop responding.
- The processing of non-TCP broadcasts with the firewall enabled (excluding DHCP broadcasts), caused a memory leak that eventually lead to a reboot.
- In units running 18.01.03 with TACACS configured, the unit would reboot during authentication.
- Phone configurations could not be edited after moving a shared line to the first button.
- When using the force delivery option in the call queue with most idle agent call distribution, the most idle agent received multiple successive calls.
- The AOS Packet Capture feature eventually caused a reboot or lockup condition.
- Auto attendant, voicemail, FindMe-FollowMe, and other voice applications did not source RTP from the correct IP address when the media gateway was a loopback.
- If the ADTRAN unit received an SDP offer with the RFC 3264 sendonly attribute to place the call on hold, it would not begin sending RTP again when a sendrecv attribute was received to take the call off of hold.
- Adding to a status group using the GUI would return an error on every mailbox that was being monitored.
- RTCP reports generated from the loopback account were incorrect making it impossible to calculate round trip delay.
- In a PSTN gateway application, if an attended transfer is initiated towards the ADTRAN gateway the transfer-or will send a REFER with Replaces to the gateway. The ADTRAN gateway should be responding with an INVITE with Replaces back out the SIP trunk. If the transfer-or was the party that initiated the original call to the transferee, the INVITE is not sent. However, if the transferee initiates the original call, the INVITE is sent and the transfer executes properly.
- When using SIP over TCP, the source port displayed in the output of **debug sip stack messages** command was incorrect.
- On the IP Phone Configs menu, when adding phone MAC addresses to the staging area some phone models could not easily be selected from the drop-down list because the list scrolled off the menu.
- Polycom configuration files referenced an incorrect version of the SIP configuration file. This prevented the phones from booting due to a mismatch in the SIP configuration versions. This only occurred if a NetVanta 7000 Series unit was upgraded to R10.2.0 and then the configuration of a Polycom phone was modified.
- NetVanta 7100 units rebooted when a SIP call was placed between a NetVanta 7100 with NAT and another NetVanta 7100 without NAT.

- After upgrading to R10.1, the VQM reporter grammar options always show up after booting, even if they are not configured.
- The informational text on the Route Table GUI menu did not display correctly.
- When attempting to create a new extension and phone configuration file using the IP Phone menu on the GUI, the password for the voice user did not match the password in the IP phone configuration.
- The Upload Phone Firmware dialog box for uploading ADTRAN firmware would not display when using Internet Explorer version 8.
- In the GUI, a 503 Service Unavailable response was delivered when selecting any trunk account type other than SIP.
- If a local user was transferred to a remote user with a 10-digit number using attended transfer, the resulting connection had one way audio. If the local, 4-digit extension is used, or if a blind transfer is used, the call would function as expected.
- Uploading a CSV file that contained Polycom SoundPoint IP 550/560 (a deprecated phone type) caused the GUI to become unresponsive.
- Under very specific circumstances (for example, a specific DNS server and name to resolve), the unit would reboot.
- In some cases, the nonce count was improperly parsed by the SIP stack, which caused authentication failures when using the SIP proxy.
- The ring group help text for the **max-inbound** command implied incorrectly that 9 is the maximum when 10 was correct.
- Random SIP authentication password would not function when adding a second extension to an IP phone configuration.
- Using the auto attendant to change the Dial By Name prompt language to Irish was only partially successful. Some of the prompts would remain in English.
- The NetVanta 7100 would not properly terminate the previous SLA state subscription if the phone responded with a 500 to the NOTIFY with Subscription-State: Terminated.
- In certain cases, the ISDN caller ID name was not delivered when configured for delivery in a Facility message after the Call Proceeding message instead of a Setup message.
- Configuring a nondefault PRI response code mapping for a 403 Forbidden response to a 100 PRI cause code would not function properly.

This section highlights major bug fixes in AOS Version R10.2.1.

- With password encryption enabled, BGP authentication passwords were being encrypted a second time on reboot.
- The NetVanta 3120 would not respond to SNMP polls for VQM.
- Browsing to the Spanning Tree menu in the GUI could return a 503 Service Unavailable response.
- Generated checksums for the **show startup-config** and **show running-config** would not match when the configuration files were identical.
- Changing PoE settings using the GUI could cause a 503 Service Unavailable response.

- In a PSTN gateway application, if an attended transfer was initiated towards the ADTRAN gateway, the transferor would send a REFER with Replaces to the gateway. The ADTRAN gateway should respond with an INVITE with Replaces back out the SIP trunk. If the transferor was the party that initiated the original call to the transferee, the INVITE was not sent. However, if the transferee initiated the original call, the INVITE was sent and the transfer executed properly.
- When using SIP over TCP, the source port displayed in the output of **debug sip stack messages** command was incorrect.
- After upgrading to R10.1.0, the VQM reporter grammar options always showed up after booting, even if they were not configured.
- IPv6 traffic destined to 0:: was forwarded to the default gateway instead of being dropped.
- A QoS policy could not be configured on a demand interface.
- The IPv6CP protocol state could occur even when IPv6 was disabled on a PPP interface.
- When 802.1q encapsulation was disabled on an Ethernet interface, the interface could not be configured for **port-auth supplicant** mode.
- The Setup Wizard for a NetVanta 3120 became unresponsive on the System Info menu.
- Removing a PPP cross connection and then adding it back to a SHDSL interface caused the PPP interface to remain down, unless the SHDSL interface was disabled and then re-enabled.

This section highlights major bug fixes in AOS Version R10.2.0.

- After an external call was blind transferred and then retrieved by a Pickup Group, no audio was heard.
- Calls from an auto attendant to an extension could not be retrieved using Directed Call Pickup.
- The GUI would allows a user name with an apostrophe to be added to a status group. An error should have been presented since apostrophes are not supported in a status group.
- Using the Update Directories action from the GUI deleted the speed dial setting from Polycom IP phones.
- Event logs showed false positives for 911 emergency calls.
- The Rebuild All and Update Directories actions caused all Polycom phones to be removed from IP Phone Configs GUI menu.
- When an E1 configuration in the GUI was modified, the unit could have become unresponsive and rebooted.
- A 503 Service Unavailable response to a Notify message caused unattended transfers to fail.
- Altering the line key default values in the GUI would not change the value in the config files for ADTRAN 7xx Series IP Phones.
- Polycom phones could have locked up when resubscribing to a particularly large status group.
- Directed Call Pickup failed when a Pickup Group was not configured on the unit.
- Callers could not be forwarded to toll-free numbers from Call Coverage unless toll-free numbers were added to the dial plan templates as a long-distance number.
- The NetVanta 7100 would not properly clear calls with an invalid call state.
- Parked calls were dropped if a member of a Pickup Group attempted pick them up.

- The talk path was lost on calls to or from an ISDN trunk when **modem passthrough** was enabled and the calls were received while other calls were active.
- Turning on a large amount of debug information would adversely affect the performance of the unit.
- The NetVanta 7100 leaked a miniscule amount of memory with every inbound call to a queue.
- INVITEs sent from the NetVanta 7100 contained an extra quotation mark which caused the IP phones to not respond to the INVITE.
- The **ring-voltage** command was removed as an option for an FXS interface.
- The **ip sip inbound-trunk-matching prefer trunk-routing** command interfered with the ability to use a remote gateway as a transparent proxy and PSTN trunk gateway simultaneously.
- On the Personal Phone Manager FindMe-FollowMe menu, modified Ring External durations were not applied correctly. The value was accepted, but reverted to the previously configured value after the user applied the changes.
- When the ISDN Layer 1 bounced, it could have caused a packet leak.
- Portal lists with invalid characters could be created in the GUI.
- The NetVanta 7100 would add an extra set of quotation marks to the FROM header for certain call flows.
- When a voice user's standard greeting was stored on the CompactFlash and became corrupted, the voicemail system would not failover to the voice user's default greeting.
- The NetVanta 7100 could have rebooted after issuing the **show voice call summary active** command.
- The **group-ring-call-waiting** command was ignored by station-to-station calls to a ring group.
- If the second leg of a trunk-to-trunk B2BUA call negotiated to a lower **session-expires** than the first leg, the value of the second leg timer was used for the first leg as well. This behavior only affected user agents that prevented reINVITEs more often than the minimum **session-expires** value.
- The firewall would not block traffic from a device on the LAN that spoofed the NetVanta 7100's WAN IP address.
- Loss in connectivity between a NetVanta 150 Access Point and a NetVanta Access Controller could have caused the Access Controller to reboot.
- If the IP Interfaces GUI menu was used to delete a MLPPP interface, the interface was deleted from the configuration, but was not fully torn down in the unit. This caused problems when creating new MLPPP interfaces.
- The GUI option to Upload Firmware from Default Firmware selection tab under IP Phone Globals > Boot Settings failed with an error.
- New phone configurations created using the manual input method for existing users with nondefault SIP authentication passwords would not be created with the correct password.
- The clock source for a WAN T1 could not be configured from the GUI.
- A 200 OK SIP response to SIP INVITE with the sendonly media attribute did not contain SDP attribute for recvonly or inactive.
- The Slot LED for an FXO/FXS VIM always remained red.
- In the VQM RTP Monitoring menu, the refresh button refreshed the displayed graphic, but it also duplicated information in the lower part of the menu. Also, when the cursor hovered over a data point, it displayed multiple instances of the same data.

- Polycom phone configurations were created with incorrect dial plans.

Errata

The following is a list of errata that still exist in AOS version R10.3.1.

- The unit's FTP server does not respond properly to a LIST command with the required carriage return / line feed formatting. This results in some FTP clients failing to connect.
- Port authentication does not function on the NetVanta 7100.
- 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 than 16 simultaneous SIP to PRI trunk calls configure the system to use G.711 encoding which is not affected.
- SIP trunk calls may not hear ring back when calling into an AutoAttendant 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 AutoAttendant) or if the group is configured as ring-all type this will not occur.
- Single Number Reach may fail to detect fax tones from certain fax machine models.
- 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.
- The Switchboard may be unable to find a target for call routing, causing a reboot.
- 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.
- The **ip sip grammar from user international** command is not changing the FROM field in the SIP header to the E.164 format.
- 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.
- Modifying a voice user on the GUI results in two duplicate User successfully updated messages.
- 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 VQM is enabled, there is no audio for external calls on a simple remote phone configuration.
- 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.
- On the Numbered Options tab in the GUI, the Value field does not display completely.
- Using pickup groups with media anchoring may result in one way audio.

- Deleting all voicemails in a user account using the PPM is very slow and may not completely empty the voice mailbox.
- Using the PPM to delete a large number of voicemails may result in a error dialog box that persists.
- The NetVanta 7100 fails to send a NOTIFY to the Advatel IP console after a status group subscription if the group contains 32 members.
- Call Coverage incorrectly states that it goes to Busy for some System Modes.
- The Request-URI field in the INVITE sent to a remote phone is being encrypted.
- When media anchoring is configured and the group ring type is set to Executive Ring, calling the ring group from a SIP phone results in one way audio when the executive member answers.
- The Configuration Successfully Saved dialog box does not appear when saving the configuration from the System Summary GUI menu.
- The PPM does not automatically make the Ring External Number option the last option that you can configure. Instead, it lets the user configure other additional options afterward and saves the configuration without problem.
- 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.
- It is not possible from the GUI to set the number of rings for a user with a call coverage list.
- 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.
- In FindMe-FollowMe actions, where the accept option is available (enabled) but the number of seconds before disconnect has been set to a low value, the call may disconnect before the user can hear the caller's name or have a chance to press 1 to accept the call. The workaround is to increase the action time setting to allow additional time to accept the call.
- 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.
- Modem tone detection does not function on ring group calls.
- 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 refreshed the displayed graphic, but it also duplicated information in the lower part of the menu. Also, when the cursor hovered over a data point, it displayed multiple instances of the same data.
- T.38 FAX call tests fail after T1 PRI loss and system timing shifts. A reboot is required to clear the condition.

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 R10.3.1 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*
- *NetVanta Dual SFP+ XIM Quick Start Guide*
- *NetVanta Dual SFP XIM Quick Start Guide*
- *NetVanta Dual Stacking XIM Quick Start Guide*
- *NetVanta 1544 Series Gigabit Ethernet Switch Quick Start Guide*
- *NetVanta 1600 Series Gigabit Ethernet Switch Hardware Installation Guide*
- *NetVanta 1534 Series Gigabit Ethernet Switch Quick Start Guide*
- *NetVanta 1230 Series Fast Ethernet Switch Quick Start Guide (2nd gen)*

