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Release Notes
AOS IAD Products

AOS Release A1.03.00
June 25, 2008

Contents

Purpose and Supported Platforms

AOS Voice Products release A1 is a major system release that adds many new features and addresses customer issues that have been uncovered in previous code releases.

Release A1.03.00 is Generally Available code, meaning that 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 Appendix A.

A listing of available documents for this release appears in Appendix B. Configuration guides, white papers, data sheets, and other documentation may be found on Adtran's knowledge base, http://kb.adtran.com.

The contents of these release notes will focus on the Netvanta 6355 and the Total Access 900/900e series platforms. Netvanta 7100 release notes are available on the Adtran knowledge base:

http://kb.adtran.com/article.asp?article=2299&p=2

Supported Platforms for A1.03.00

- **TA 900 Series** – VoIP Multiservice Access Gateway, single T1 interface
- **TA900e Series** – VoIP Multiservice Access Gateway, multi-T1 interface
- **NetVanta 6355** – Multiservice Access Gateway
- **NetVanta 7100** – Converged IP Office in a Box

Summary of New Features

This section highlights the major features, commands, and behavioral changes for AOS A1.03.00. For a list of related documents, please see Appendix B.

Additions for All Voice Products in A1.03

Sip Diversion

The Sip Diversion feature in A1.03 converts Redirecting Number IEs on a PRI trunk into Diversion headers on the SIP trunk, allowing the calling party information to be preserved. This functionality is performed automatically after upgrading to A1.03.

Additions for 2nd Gen TA 900/900e series in A1.02

MGCP – Media Gateway Control Protocol

MGCP is a newly added VoIP call control method available for 2nd generation TA 900s. FXS ports are the only available endpoints for MGCP. No T1 CAS or T1 PRI support will be provided in this release. Media will still be transported over RTP, as it currently is with our SIP implementation. SIP and MGCP can coexist on the same IAD at the same time (i.e. SIP used for PRI delivery while MGCP used for FXS delivery).

3-Way Conferencing

Local 3-way conferencing is now supported in the 2nd gen TA 900 series. 3-way conferencing is only supported in 2nd generation TA 900s. The only local conferencing participants currently supported are FXS voice users. The IAD is limited to only 3 separate 3-way conferences at a time.

Additions for All Voice Products in A1:

VQM – Voice Quality Monitoring

Ability to make VoIP quality measurements on RTP media flows terminated or passed through the IAD. Results will estimate MOS scores for particular calls as well as jitter buffer performance.

Loopback Accounts

When the loopback account call is connected, the RTP audio is looped back. This will provide an easy method to verify proper operation and configuration during install, and can be used with VQM to troubleshoot network issues. This feature includes the ability to initiate SIP calls via the CLI.

AWCP - ADTRAN Wireless Control Protocol

AWCP is a Layer 2 Control and Management protocol that enables the platform to act as a Wireless Access Controller for the Netvanta 150 Access Point. Up to eight Netvanta 150 Access Points can be configured and managed by the AC.

Top Talkers

The NetFlow flows that are generating the heaviest system traffic are known as the "top talkers." The NetFlow MIB and Top Talkers feature can be used for security monitoring or accounting purposes for top talkers, and matching and identifying key users of the network.

Full PRI support

All 23 B channels of a PRI can be used simultaneously. The number DSP resources are increasing from 16 to 24 channels on the RoHS-E1 units.

Top visited websites

The top websites feature is designed to report top websites requested by users to system administrators. This feature is intended to be used in conjunction with the *ip urlfilter* command so that customers without a Websense server will have a simple URL filtering package.

Additional Features

- VRF aware DHCP server
- VRF aware firewall
- Portal-List(s) to assign username access to different portals (system applications such as http, telnet, ssh, ftp, and console). Previously, a username for ftp could also telnet or ssh into the unit.

Summary of Bug Fixes

This section highlights major bug fixes in AOS version A1.03.00.

“ip sip qos dscp x” needs a value verified option

Issue Detail

- Alphanumeric values can be entered as a DSCP value. Numeric values are appropriately checked to verify that they are in the appropriate range.

Corrective Action

- Added a check for non-numeric values.

6355 only: T-1 NIM loses LBO setting after a reboot

Issue Detail

- T-1 NIM loses LBO setting after a reboot. This problem is specific to the NIM rather than the VIM.

Corrective Action

- Modified several framer driver files to resolve the issue.

Reboot during reINVITE when a call is forwarded locally

Issue Detail

- If a call is placed into a ring group and that call is in the middle of being transferred locally from one ring group member to another, the unit will reboot if the IAD receives a reINVITE from the softswitch.

Corrective Action

- Made a change to switchboard timing to accommodate the reINVITE.

Time between ringing pattern & FSK tones for CID incompatible w/some customer PBXs

Issue Detail

- The time between the end of first power ringing pattern and start of FSK should be 0.5 – 1.50 seconds per GR-30-CORE. Our previous implementation was at the low end of the spec (500 ms). We have seen that some PBXs have problems detecting CID at the lower end of the spec.

Corrective Action

- Added a config option to the FXS interface config to make the CID timing configurable: “caller-id delay <500 – 2000>”

Record-Route address ignored during call setup

Issue Detail

- When a Record-Route is received, messages are sent to the layer 3 Contact address instead of the address in the Record-Route field.

Corrective Action

- Added logic to check to see if the previous request had a record-route.

“IP unnumbered” for PPP interface not parsed correctly during bootup

Issue Detail

- A previous fix that was made to prevent allowing multi-access interfaces from being unnumbered caused a different issue upon bootup or when configuring a fresh interface.

Corrective Action

- Changed rule to prevention mechanism for allowing unnumbered addresses on multi-access interfaces.

Match-sub with empty substitution isn’t maintained after a reboot

Issue Detail

- When using “match XXXX substitute “”” to remove DNIS digits from a call, the match/sub will not be maintained in the running config after a reboot. The persistence viewer kills this command on a reboot so the user will always have to re-enter the command.

Corrective Action

- Changed trunk manager check to accommodate empty quotes for match/substitutes.

900 ADSL Only: 503 error when trying to navigate to the physical interfaces page on the GUI

Issue Detail

- The user will get a 503 http error when trying to access the “Physical Interfaces” page on the System tab of the web GUI.

Corrective Action

- Made appropriate changes to GUI architecture to handle ADSL interface status.

Authentication fails for inbound calls to the IAD due to empty username field

Issue Detail

- If an inbound call is placed to the IAD that requires authentication in the softswitch (either when sending a BYE to tear down the call or when sending a reINVITE for a fax call), the username field sent by the IAD will be empty. This is not an issue if the number is registered or if the initial call was outbound.

Corrective Action

- Made changes to CLDU to get the username correctly from the running config.

Invalid Extra Space on CID sent to trunk / user when FROM CID doesn’t contain quotations

Issue Detail

- The IAD adds an extra space to CID taken from the SIP FROM field if it's not enclosed in quotations. The caller-id name should be "abcdefghijklmno", and not "abcdefghijklmno ".

Corrective Action

- Made changes to CID parser is the SIP trunk manager to better handle cases where CID is sent without quotations.

Invalid User-agent field sent from IAD

Issue Detail

- In A1, the IADs started sending the generation information in the user-agent field: ADTRAN_Total_Access_916e_(1st_Gen)/A1.02.00.E. This violates RFC 3261.

Corrective Action

- Removed parenthesis from user agent field to comply with the RFC.

Encrypted passwords larger then 62 characters won't be maintained after a reboot

Issue Detail

- If a password larger then 62 characters is used for any config parameter that is encrypted, the added encryption characters will cause the new hash to be larger then 64 characters. This causes an issue with the Persistent Viewer on bootup.

Corrective Action

- Made changes to the length check used when passwords are encrypted.

900e/6355 only: Ethernet port may stop transmitting if there is a duplex mismatch between CPE

Issue Detail

- When high traffic-rates force late collisions, due to duplex setting mismatches between the IADs Ethernet port and the directly connected device, the port may cease transmitting; although, it remains up and receives traffic. A reboot will be required to restore the Ethenet interface to normal operation.

Corrective Action

- Modified restart procedure for Ethernet driver when receiving late collisions.

Possible reboot when SETUP received from PBX over PRI when "debug isdn verbose" is running

Issue Detail

- The IAD could potentially reboot when trying to display FACILITY IE in the SETUP message from the PBX. This could only occur if "debug ISDN verbose" was running when the IAD receives the SETUP.

Corrective Action

- When the IAD tried to format the debug output, the position of the cursor was not being saved and restored correctly for Facility IE length less than 14 characters. Made appropriate changes.

Fax calls are being sent to primary SIP server even after failover

Issue Detail

- If the IAD loses contact with the primary SIP server, it properly fails over to the secondary server. If a fax call is then placed, the original call leg goes through to the secondary server as it should. When the IAD detects the fax tones and sends a reINVITE, it sends it to the primary server and not the secondary.

Corrective Action

- The IAD now maintains the failover mechanism throughout the duration of the call.

6355 only: Wrong default config loaded after "erase start"

Issue Detail

- The 6355 pulls its default config from flash. Updated changes to the default config are stored in the .biz file for the firmware. This can cause the 6355 to load an older default config instead of the most recent one in the .biz file.

Corrective Action

- The 6355 now correctly pulls the default config from the .biz file.

Possible duplicate Call-IDs on REGISTERs after rebooting the IAD

Issue Detail

- It is possible that a Call-ID on multiple REGISTERs could be the same across a reboot of the IAD. This causes a situation where the registrations after a reboot may fail.

Corrective Action

- A limitation was found in the mechanism used to generate Call-IDs. Made the appropriate changes.

Missed digits when dialing from certain analog key systems / stations

Issue Detail

- Digit detection in the IAD can be problematic with older analog stations.

Corrective Action

- Changes to the DTMF detection libraries in the DSP will increase the reliability of digit detection in A1.03.

RTP DSCP Options for MGCP endpoints showing up incorrectly in config

Issue Detail

- The DSCP value entered for RTP traffic originating from an MGCP endpoint displays improperly in the running configuration, even though the correct value is being placed in the RTP packet. This does not effect performance. It is purely cosmetic.

Corrective Action

- Made the appropriate changes to print the correct value to the output of the running config.

Call type INTERNATIONAL no longer prepended with 011 outbound on SIP trunk

Issue Detail

- In previous versions of code, the IAD would prepend a 011 to the dialed number for calls out the SIP trunk if the call type in the SETUP from the PRI was an international call. This would only work if an isdn-number-template was configured to match on international call type. This functionality was broken in rev 14.

Corrective Action

- The Trunk manager wasn't properly checking the call type field of the SETUP on the PRI. Added the appropriate check.

Cannot manage IADs running A1 in N-Command

Issue Detail

- The IADs show up as "Unmanaged SNMP Devices" in N-Command when running any version of A1. This is a problem with how N-command parses the system description field in the IAD. Upgrading IADs to A1 does work without any loss of IAD configuration, but the management functionality is lost.

Corrective Action

- The has been addressed and will be updated automatically by n-command.

Upgrade Instructions

Several steps need to be taken to assure a valid upgrade. First, save your existing configuration to a tftp server. The command to execute this step is

```
Router# copy start tftp
```

You will be prompted for file names and the server address in the process.

Next, download AOS version A1.03.00 from the ADTRAN support website. When properly installed on your tftp server, the file will have the form "*product-version.*biz*"* where *product* is the platform name, and *version* is the AOS image version (A1.03.00) separated by hyphens instead of decimals.

From your privileged prompt:

```
Router# copy tftp flash
```

During the tftp download you will be prompted for the tftp server name, the tftp server file name, and finally the name you want to give the file once it is transferred to the on-board flash. The flash can hold up to eight megabytes of files, whether AOS or configuration files. Now from the Configuration prompt:

```
Router (config)# boot system flash filname.biz
```

The boot command tells the router which software on the flash to use as the boot up AOS. The router should now be rebooted with the privileged command

```
Router (config)# reload
```

When the unit reboots, it will be running AOS version A1.03.00.

Appendix A – Errata for A1.03.00

The following is a list of errata that still exists in A1.03.00.

MGCP only: One way audio after receiving a G/rt in a MDCX

If the remote gateway changes the connection mode from SENDRECV to SEND or RECV in a modify connection *and* that same modify connection also contains a ringback event (G/rt), the IAD will permanently stop sending audio in the RTP stream. This will be addressed in A1.04

MGCP only: Forward disconnect in MGCP when L/osi is received should be 900 ms

According to RFC 3660, the disconnect time when an osi is received from the call agent should be 900ms. Currently, the IAD removes battery for 500ms. This will be addressed in A1.04.

MGCP only: Distinctive ring and ring splash does not work properly

Distinctive Ringing signal requests (r0, r1, r2, r3, r4, r5, r6 or r7) from the call agent all ring the same pattern (2s on, 4s off). This will be addressed in A1.04

Start time for VQM RTP stats downloaded to .CSV file is inaccurate

The value used for start time is a UTC time value, not the normal system time expected by the user. This will be addressed in A2.01.

Erroneous problem reported with Public Security zone

If 802.1q subinterfaces are assigned to the ethernet port, the GUI system summary reports a problem with the Public security zone not being set correctly. The unit functions fine, but the GUI has an issue detecting the VLAN sub-interface when sanity checking the configuration. This will be addressed in A2.01.

GUI Call Quality Stats page shows codec as 'undefined'

In the GUI, Voice -> Call Quality Stats, codecs are displayed as 'undefined'. The CLI shows the correct output. This will be addressed in A2.01.

Caller-id name override doesn't work for inbound calls on a trunk

Caller-id number override for voice trunks works properly; however, the CID name field cannot be manipulated. This previously worked in version 12. This will be addressed in A2.01.

INVITE not properly changed after 302 'moved temporarily'

If a 302 is used to change the destination port for subsequent calls to a specific user, the URI generated by the IAD contains the requested port number, but the destination port for the SIP packet still contains the original port number. This will be addressed in A2.01.

URI matching for SIP PUD needs to be tweaked to handle certain SIP endpoints

If a SIP endpoint sends a REGISTER through the proxy containing an rinstance but does not send the ristnance in the INVITE, the PUD lookup in the proxy will think that the two contact headers don't match. This violates RFC 3261. This will be addressed in A2.01.

Improper detection of V.8 ANSAM without phase reversal

In the rare case that a fax/modem device sends a V.8 ANSAM without phase reversals, the IAD will report V.8 ANSAM with phase reversal as soon as the ANSAM is detected. However, when the end of the tone is detected, the IAD then could report V.8 ANSAM (no PR), causing the controller-side behavior to send another INVITE. This will be addressed in A2.01.

Trunk appearance leak on E&M wink trunk w/ dialtone enabled
If dialtone generation is enabled on an E&M wink trunk without saving the change to the startup-config and then rebooting the IAD, a trunk appearance could lockup and by unavailable. Rebooting the IAD will permanently clear up the issue. This will be addressed in A2.01.

aaa authorization command changes terminal length to 0
"aaa authorization exec default group tacacs+ if-authenticated" changes the terminal length to 0. Entering "terminal length <#>" will clear up the problem. This will be addressed in a future release of code.

DNS query initiated by SIP stack causes "ip host" entry to be removed from config
If an IP host entry is configured, and then the SIP stack initiates a DNS query for that FQDN, the ip host entry will be removed from the running config. This will be addressed in a future release of code.

Inbound calls to DID 9111 match emergency-call dial plan
Calls received on a trunk (SIP, PRI, E&M, FXO) with a dialed number of 9111 will match the emergency call dial plan entry. The call will still process correctly; it will just be given an improper priority in the switchboard. This will be addressed in a future release of code.

Empty username sent in Authorization header of BYE message
When a BYE originating from the IAD is sent with authentication, the username field is empty in the Authorization header of the BYE message. This will be addressed in a future release of code.

Problems handling 0.0.0.0 SDP in RFC 2543 hold through the SIP proxy
If the proxy receives an INVITE to put a call on hold using RFC 2543, the associated 200OK will cause the IAD to create a firewall association for the non-existant RTP stream using an IP of 0.0.0.0. When this fails, the proxy never forwards the response on to the phone. This will be addressed in a future release of code.

Audio cuts out during call waiting beep on FXO to FXS call
When call waiting is disabled on a voice user and an inbound call hits the FXO trunk (destined for the voice user), the audio cuts out during the period where the call waiting beep would normally be played. This will be addressed in a future release of code.

SIP to MGCP Ringback issue
While placing a call from a SIP user to an MGCP endpoint on the same TA 900 with both lines registered, the SIP user will not hear ringback. This is a result of the separation of SIP and MGCP internally in the IAD. This is only an issue with a soft switch that instructs the MGCP endpoint to provide remote ringback in the signaling field of the MGCP message (i.e. S: rt@$). This is also only an issue on hairpin calls.

DNS proxy failover failure
If a DNS request is sent through the DNS proxy of the IAD to the primary public DNS server and the DNS server responds with a "destination unreachable", the secondary DNS server won't be queried.

SSH sessions are always listed as authentication in progress
The "show users" command does not list the idle time of any SSH sessions. They are always listed as "authentication in progress" whether or not the remote users have authenticated.

Channels on 2nd PRI fail to establish voice path (900e only)

Due to how resources are allocated from the DSPs on the 900e, only 32 simultaneous calls will connect over 2 PRI interfaces if the first 23 calls are brought up on T1 0/4. The next 9 calls that connect on T1 0/3, for a total of 32 simultaneous calls, will work correctly. Any subsequent simultaneous calls (more then 32) will experience no media cut through. Due to the order in which calls must be connected in order to experience this issue, there is a low risk of experiencing this problem in the field.

24th call cannot generate DTMF digits out DSX wink trunk due to resource limitation

A resource is reserved for RTP, and if a resource is not available for DTMF generation, that call out the DSX will fail. 23 calls work fine, as long as the 23rd call is not fighting for a resource with another call at that time.

MGCP limited to 18 FXS G711 Hairpin calls when using T-1 as local media gateway

The IAD is limited to 18 FXS Hairpin calls when the MGCP voice gateway is pointed out the WAN interface. It has been verified that 24 Hairpin calls work when the gateway is pointing out the ethernet interface.

Output of “show crypto” displays more VPN tunnels then are supported by the device

We currently support 30 VPN tunnels on the 900 products. The output of “show crypto” displays 200 for IKE and 400 for IPSEC.

Appendix B – Related Documents

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

AOS A1.01 Command Line Reference Guide (13MB file) –
http://kb.adtran.com/article.asp?article=2219&p=2

Voice Quality Monitoring Config Guide -
http://kb.adtran.com/article.asp?article=2262&p=2

[Video]Understanding Voice Quality Monitoring in AOS -
http://kb.adtran.com/article.asp?article=2296&p=2

Integrated Traffic Monitoring Config Guide (Top Talkers Support) -
http://kb.adtran.com/article.asp?article=2157&p=2

Multi-VRF Config Guide –
http://kb.adtran.com/article.asp?article=2156&p=2

URL Filtering/Top Websites Reporting Config Guide -
http://kb.adtran.com/article.asp?article=2158&p=2

AOS Wireless Config Guide –
http://kb.adtran.com/article.asp?article=2078&p=2