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Release Notes

AOS IAD Products

AOS Release A1.07.00
March 16, 2009

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 A1.07.00 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 at kb.adtran.com

Supported Platforms for A1.07.00

- **TA 900 Series** – VoIP Multiservice Access Gateway, single T1 interface
- **TA900e Series** – VoIP Multiservice Access Gateway, multi-T1 interface
- **NetVanta 6355** – Multiservice Access Gateway

Summary of New Features

This section highlights the major features, commands, and behavioral changes for AOS A1.07.00. For a list of related documents, please see Appendix B.

Additions for All Voice Products in A1.07

Enhancement to caller-id generation for FXS users

Previous to A1.07, caller-id was generated out an FXS user 1000ms after the end of the first ring cycle. A config option was added for A1.07 that makes the amount of delay configurable from 500ms to 2000ms, with 1000ms being the default. The configurable delay was added to improve interoperability between legacy PBXs and the Adtran IPBGs.

Additions for TA 900/900e series in A1.05

Enhancement to SPRE code modes

Added enhancements to SPRE code modes to allow individual SPRE codes to function in a different mode then the mode that is globally defined. Locally handled SPRE codes can also be remapped to different functions.

Additional Features

- Added config option “voice disconnect-mode fast-busy” to play reorder tone instead of dialtone after an analog call is disconnected by the remote party.
- Added config option “ip sip proxy failover accept-registrations” to allow the SIP proxy in the Adtran to respond to REGISTER messages when in permanent failover mode.

Additions for All Voice Products in A1.04

Enhancement to Match / Substitution templates

The match / substitution templates now have the ability to match on *. This will allow for the substitution of any dial string containing a *.

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 or later.

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 A1.01

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 Top Talkers feature can be used for security monitoring or accounting purposes for top talkers, and matching and identifying key users of the network. This information can be exported into a NetFlow 9 CSV file.

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 A1.07.00.

Spre-mode override causes dial tone to be presented

Issue Detail

- If a *voice spre-mode override* was configured, dial tone was presented after receiving network handled SPRE codes. If no voice spre-mode overrides were configured, there was silence while waiting for the interdigit timeout (this is the expected behavior). This issue has been addressed.

Redirect with *302 Moved Temporarily* fails

Issue Detail

- If the contact field in the 302 didn't contain the transport method (i.e. UDP), the Sip parser would attempt to resolve the contact as an FQDN. This issue has been addressed.

Sip diversion header fails to authenticate

Issue Detail

- If the Sip server tried to authenticate an INVITE that contained a SIP diversion header, the Adtran would try and authenticate with the original calling party user. The original calling party user is now sent in the reINVITE to the authentication request. This issue has been addressed.

RTP inbound to loopback account blocked by firewall unless explicitly allowed

Issue Detail

- Unless explicitly allowed, inbound RTP to a media loopback account would not pass through the firewall. The only workaround was to allow UDP ports for any/all ranges on which RTP may have been sent. This issue has been addressed

MGCP only: Caller-ID name "O" not properly translated

Issue Detail

- If a caller-ID name of "O" was received from the MGCP call agent, the correct reason code for "Unavailable" was not sent to the connected analog device. This issue has been addressed.

Comfort noise is only applied at first occurrence of speech

Issue Detail

- Comfort noise is only applied during the first occurrence of speech. Subsequent occurrences of speech / active NLP will not result in comfort noise being injected. This issue has been addressed for 2nd gen products only.

Adtran Proxy ID not removed from contact when sending response to proxied phone

Issue Detail

- When "ip sip proxy grammar proxy-id contact-user" was configured, the Contact user was properly modified with the Adtran proxy ID for REGISTER messages sent through the proxy. 200ok responses sent back to the proxied phone contained the Adtran Proxy ID, which should have been removed from the contact header. This issue has been addressed.

603 decline is sent instead of *486 user busy* for users with per-call-waiting disabled

Issue Detail

- If call-waiting was disabled on a per-call basis with *70 (with SPRE mode override configured to handle the SPRE code locally), the Adtran would respond with a *603 Decline* when a second inbound call was placed to that user. The unit now responds with a *486 user busy*. This issue has been addressed.

Stateful proxy prevents calls in failover mode if DNS entry for FQDN times out

Issue Detail

- If a unit configured for stateful proxy was in failover mode, meaning it lost connectivity to the SIP server, calls would work to the backup interface until the DNS entry associated with the proxy user timed out. This would only happen if connectivity was lost to the DNS server. DNS entries are now cached if connectivity to the DNS server is lost. This issue has been addressed.

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 A1.07.00 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 (A1.07.00) 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 A1.07.00.

Appendix A – Errata for A1.07.00

The following is a list of errata that still exists in A1.07.00.

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 issue has been addressed in A2.

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 has authenticated. This issue has been addressed in A2.

Erroneous problem with Public Security zone

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

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 issue has been addressed in A2.

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 Adtran contains the requested port number, but the destination port for the SIP packet still contains the original port number. This issue has been addressed in A2.

Problems with URI matching for SIP Proxy User Database

If a SIP endpoint sends a REGISTER through the proxy containing an r-instance value but it doesn’t send the r-instance in the INVITE, the Proxy User Database lookup in the proxy won’t correctly match the two contact headers. This issue has been addressed in A2.

Improper detection of V.8 ANSAM without phase reversal

In the rare case that a fax/modem device sends a V.8 ANSAM event without phase reversals, the Adtran will initially, incorrectly, report V.8 ANSAM w/ phase reversal followed by V.8 ANSAM w/ no phase reversal when the tone completes. This will cause the Adtran to send a reINVITE for the improperly detected tone. This issue has been addressed in A2.

Trunk appearance leak on E&M wink trunk w/ dialtone enabled

Enabling dialtone on E&M wink trunks requires a reboot to prevent the trunk appearance from locking up. After the initial reboot, all E&M wink trunk will function correctly. This issue has been addressed in A2.

Question marks used for URL-based options are interpreted as context-sensitive help in the CLI

Question marks can’t be used in URL-based DHCP options. They are incorrectly interpreted as context sensitive help. This issue has been addressed in A2.

PPP: 'no shutdown' on enabled interface causes it to bounce
Entering the “no shut” command on a PPP interface that has already establish a PPP connection will cause the interface to bounce and re-negotiate. This issue has been addressed in A2.

MGCP Confirmation tone (g/cf and l/cf) does not work
When the TA 900 receives an S: g/cf or an S: l/cf to play a confirmation tone, nothing is played out. This issue will be addressed in A4.

A single T.38 FAX call does not work without disabling plc on the voice user
Plc must be disabled for T.38 FAX to work properly. “no plc” on the voice user or trunk will disable packet loss concealment. This issue will be address in A2.

MGCP Inbound calls fail if LocalConnectionOptions are not present in CRCX
If a create connection is sent from the call agent without the LocalConnectionOptions field in the MGCP header, the subsequent call will fail. This issue will be address in A2.

SIP to MGCP Ringback issue
While placing a call from a SIP user to an MGCP endpoint on the same Adtran with both lines registered, the SIP user will not hear ringback. 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.

1st gen 900/900e only: Possible issue with DTMF generation under heavy call load
With more than 18 simultaneous calls connected on a 1st gen TA 900 or TA900e series IAD using G.729 codec, it is possible that DTMF tones will not be recognized by the terminating CPE due to an issue with the Adtran generating frequencies at 2804 Hz or higher under heavy call load. This degradation in tone generation cannot be heard by the human ear. With 23 simultaneous calls, the call completion rate to terminating CPE is approximately 99.5%. With 24 simultaneous calls, the call completion rate drops to approximately 97%.

DNS proxy failover failure
If a DNS request is sent through the DNS proxy of the Adtran to the primary public DNS server and the DNS server responds with a "destination unreachable", the secondary DNS server won’t be queried. This issue has been addressed in A2.

900e only: Channels on 2nd PRI fail to establish voice path
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
For all calls into an E&M wink trunk, a DSP resource is reserved for RTP during call setup. This could prevent a DSP resource from being available to generate DTMF for DNIS if 23 calls are already active. This only effects 1st gen TA 900 series products.

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

The Adtran 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.

503 error on system summary page of GUI

If a multi-link Frame Relay interface is configured, the System Summary page of the GUI will error out with a 503. This issue will be addressed in A2.

"copy http" command only works with FQDN when FQDN is not in the host table

The copy http command only works with a FQDN if it is not currently in the host table. If the FQDN is already in the host table, the command will immediately exit with no error message. This issue will be addressed in A2.

Modem-passthrough does not work for ring-group users

On calls placed into a ring group, no fax/modem detection resources are allocated. This will prevent the Adtran from automatically being able to switch to data mode for fax or modem calls. This issue will be addressed in A2.

Lost packets on "show voice quality-stats" doesn't match the "show media-gateway channel" stats

The number of lost packets in the "show voice quality-stats" doesn't match up with "show media-gateway channel" stats in the following scenario: 2 G.729 calls are brought up over a 2xT1 MLPPP bundle. With both calls up, the media-gateway counters were cleared and then t1 0/1 was pulled and then immediately plugged back in (it wasn't unplugged long enough to cause any errors to display on the console). After the T1 was plugged back in, the lost packets in the "show voice quality-stats" are different than the lost packets in the "show media-gateway channel" stats. This issue will be addressed in A4.

Number of lost packets is larger than the number of expected packets

In rare cases, the number of lost packets logged by the "show voice quality-stats" could be larger then the number of expected packets for a given call. This issue will be addressed in A4.

"busy all now" on a SIP trunk does not work properly

If the user enables "busy all now" on a SIP trunk, the first inbound call will receive a 486 Busy Here, but the second call will ring normally. Outbound calls result in an INVITE being sent out the SIP trunk, even though it should be busied out. This issue has been addressed in A2.

Not sending "NOT ENDtoEND ISDN" in ALERTING message on PRI to SIP calls in response to a 180

The Adtran currently doesn't send "Description:NOT ENDtoEND ISDN" in the ALERTING out the PRI to the PBX in response to a 180 Ringing from the SIP side. This will be addressed in A4.

6355 only: Overhead Paging doesn't work
Calls to the overhead paging extensions do not work properly.

T1 in Yellow Alarm Causes 503 on System Summary page of GUI
If one of the T1s on the Adtran is receiving a yellow alarm, the system summary screen sends back a 503 server error. Once the alarm clears, it works as it should. This issue will be addressed in A4.

Problems with Total Access Config Wizard in the GUI
The Total Access Config Wizard in the system menu of the GUI won't complete past the VoIP section.

RTP is not allowed through the firewall when NAT is performed on inbound calls to a SIP user with no SDP in the INVITE
If the SDP for an inbound call is not sent until the SIP Server ACKs the 200 OK when the called party answers, the firewall will not open a hole for RTP resulting in no audio.

"Voice Quality-Stats" Jitter Buffer Average is greater than max value
In rare instances, the output of a "show voice quality-stats" could show that the average size of the jitter buffer is higher than the max value.

T.38 stack does not send any packets when a Super G3 fax is used
When a Super G3 fax machine is used behind an Adtran, T38 will not function properly. This issue will be addressed in A2.

One-way Audio - Audio Codec Negotiation Problem
If the Adtran receives an SDP where the codec preference order in the media field has DTMF relay before g.711 or g.729 (i.e. m=audio 31794 RTP/AVP 101 0), media won't be sent properly resulting in one way audio.

Jitter Buffer Mode shows adaptive after modem-passthrough detects a data call
After modem-passthrough has switched the user to data mode, the jitter buffer mode still shows adaptive instead of fixed under "show voice quality-stats [id]".

http secure server could become unresponsive
In rare cases, the secure http server in the Adtran could become unresponsive, preventing https access. CLI access is not affected.

MGCP Inbound calls fail if LocalConnectionOptions is not present in CRCX
If local connection options aren't sent in the MGCP header or if no codecs are present in the local connection options, calls will fail. This issue will be addressed in A2.

MGCP possible issue with 3-way conferencing
The issue occurs in the following scenario: Phone A calls phone B, then phone B flashes and calls phone C. If phone B flashes BEFORE phone C answers (so that A and B can talk while waiting for C), the three-way conference will fail. After Phone C answers, phone A and B will continue to hear ringback. If phone B flashes AFTER phone C answers, then three-way conference works.

Calls to a UCD ring group will result in no talk path

Calls to a ring group with uniform call distribution configured will result in no talk path in either direction.

900e / 6355 only: Possible problem with VPN connection between Ethernet ports

Under heavy load, the Adtran cannot service packets at the same rate needed for the packets to be encrypted, causing the unit to drop packets. Input decryption errors are reported to the terminal due to encrypted packets missing in the sequence. Throughput performance is slightly affected.

"debug IP packet VRF <vrf>" provides no output after Fast Flow enabled on interfaces

"Debug IP packet vrf <vrf>" on the the non-default vrf does not display any data after "ip ffe" is enabled on the ethernet and MFR interfaces. "Debug ip packet" on the default vrf will continue to relay information to the terminal.

H.323 video conference fails with H.323 ALG

Incorrect passives are formed when the media address specified in an openLogicalChannel or openLogicalChannelAck H.245 message differs from the sender of the H.245 signaling.

Possible reboot on 1st gen 900 series

The reboot that could occur is a direct result of some changes made to reduce memory overhead in the first gen products. The issue will be addressed in A2.

BGP failure to re-advertise reachable prefixes after Ethernet interface flap

When an Ethernet interface bounces, a withdrawal UPDATE is sent to all BGP peers. If the interface comes back online immediately, then the Adtran will not re-advertise the prefix. Generally this will only occur if the interface comes back online before the withdrawal is sent. This issue will be addressed in A2.

Tcl script not run properly when tied to a probe/track

Tcl scripts will not run when used in conjunction with a probe/track. The only way to run the script is to manually execute it from the CLI. This issue will be addressed in A2.

T.38 Fax call fails when switching from G.711 to T.38

When a T38 fax call is detected on an active channel using G.711, the subsequent T38 call will fail due to packet loss concealment being enabled on the G.711 channel. This issue will be addressed in A2.

VQM stats are reset for calls where the media port is changed during a call

If the media port is changed after a call has already been established, the RTP monitoring stats in VQM will treat the new media stream as a separate call leg. This issue will be addressed in A2.

Fax/modem detection is not enabled for analog ring group users

If a call is placed to an analog FXS user through a ring group, a modem-passthrough detection resource is not allocated. This prevents fax calls from switching to analog data mode. This issue will be addressed in A2.

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 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