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Release Notes
IP Business Gateways

AOS Release A2.07.00
June 25, 2010

Contents

Purpose and Supported Platforms

AOS Voice Products release A2 is a major system release that adds many new features and addresses customer issues that have been uncovered in previous code releases.

Release A2.07.00 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 Total Access 900/900e series platforms and the Netvanta 6310. Netvanta 7100 release notes are available on the ADTRAN knowledge base:

Supported Platforms for A2.07.00

- **TA 900 Series** – VoIP Multiservice Access Gateway, single T1 interface
- **TA 900e Series** – VoIP Multiservice Access Gateway, multi-T1 interface
- **NetVanta 6310** – VoIP Multiservice Access Gateway, modular WAN

Summary of New Features

This section highlights the major features, commands and behavioral changes for AOS A2.07.00.

Enhancement to Feature Group D trunks

- Added the ability for the DNIS number in a feature group D message to match a dial plan entry rather than waiting for the trailing *. Also, the digits sent as DNIS from the PBX will now use the global inter-digit timeout.

Enhancement to SIP timers

- Previously, the SIP rollover timer would apply to both INVITES and REGISTER messages. New commands have been added to allow for the independent configuration of the rollover timer for registration (e.g. ip sip timer rollover register)

Additions for IP Business Gateways in A2.06.00

Enhanced E911 support

- Allows support for a 911 call center to call back the same analog or IP phone that made the original 911 call.

Additions for IP Business Gateways in A2.05.00

Two B-Channel Transfer

- Two B-Channel Transfer (TBCT) enables a PBX connected to a PRI interface to connect two independent calls together.
- Once the ADTRAN accepts the request, the PBX is released from the call and the two call legs are transferred on the SIP trunk using the REFER method.
- TBCT is the National ISDN variant of ISDN Explicit Call Transfer Supplementary Service.
- See Appendix B for a link to the quick configuration guide.

RFC 4904 Trunk Group Identifier

- This feature aims to implement RFC 4904 compliant transportation of originating (ingress) trunk group parameters. Trunk group parameters (TGPs) are exchanged between Internet signaling entities to identify originating and terminating trunk groups. Originating TGPs are established based upon the ingress trunk group. Terminating TGPs represent routing decisions downstream to a specific egress trunk group.
- To enable both originating and terminating trunk group parameters, RFC 4904 specifies that originating TGPs are supplied within the SIP Contact header and terminating TGPs are identified in the Request URI.
- See Appendix B for a link to the quick configuration guide.

ETSI PRI Overlap Signaling

- This feature allows an ISDN PRI interface to accept a SETUP message with a missing or incomplete called party number information element. If there are no called party digits in the SETUP message, the ADTRAN generates dialtone. Dialtone is cancelled after one or more dialed digits have subsequently been received.
- This feature doesn’t require any configuration and is for use with ETSI PRI interfaces on the NV 6300 series only.

Additional Features

- Added support for DNIS out-pulsing over ring-groups for FXS users.

Additions for IP Business Gateways in A2.04.00

Enhancement to DSO leveling for TDM trunks

- Added an enhancement to DSO leveling on a TDM/PRI trunk to allow for 1dB increment level adjustments in both directions.
- The new config options are configured on the voice trunk with the commands “rtp tx-gain x” and “rtp rx-gain x”, where x is the attenuation/gain value of -6 to 14.
- This feature is not supported in the 1st gen 900/900e series.

DNIS out-pulsing over FXS

- FXS interfaces will now be able to send dialed digits to devices on an FXS interface via DTMF.
- This feature is helpful when sending calls to an attached fax server with analog interfaces.
- It can be enabled on the Voice User associated with the FXS interface with the command “dnis-digits x” where “x” is 1-16 digits.
- The DNIS digit out-pulse can be configured with delays before sending digits (after the call is answered).

Distinctive Call Waiting

- This feature will enable distinctive call waiting tones.
- The tones will trigger on Bellcore Alert-Info headers.
- The feature will be enabled for any voice user configured with ‘special-ring-cadences’

SIP Proxy Shared Line Appearance

- The purpose of this feature is to extend failover functionality to cases in which phones are using SLA lines to originate and accept calls.
- This feature is required in order to distribute a request to multiple proxy users that have been registered with the same dial string.
- To enable this feature, the command “ip sip proxy duplicates-allowed” setting must be configured in global config.

Modem-Passthrough Auto Call-Waiting Disable

- MPACD is a feature that will automatically disable call-waiting, triggering on an incoming fax or modem call.
- The call-waiting is disabled for the duration of this call only.

Additional Features

- Added support for Ground Start with MGCP.
- 16 digit alphanumeric passwords are now supported for registration to SIP voice users. The default passwords remain the same.
- Added SNMP trap support for both CPU and heap utilization. Once the threshold values have been exceeded for the specified time interval, an SNMP trap will be sent.

Additions for IP Business Gateways in A2.03.00.SC

Enhanced ANI Substitution

- Enhanced ANI substitution allows the user to change both the number and the name (if the trunk supports ANI name information) of the calling party on a per-trunk basis for outbound trunks.
- Additionally, ANI substitution allows the per-trunk configuration of ANI replacement based on DNIS. This is a one-to-one replacement that occurs on outbound trunks that support ANI. Both the name and number of the calling party are optionally affected, but it does not affect the called party information in any way.
- Although the Total Access 900 and NetVanta 6000/7000 Series support both the traditional and enhanced versions of ANI substitution, it is important to remember that the traditional ANI substitution is configured globally on inbound trunks, and the enhanced ANI substitution is configured on a per-trunk basis for outbound trunks

Source and ANI Based Call Routing (SABR)

- SABR is a feature on AOS voice products that enhances call routing services by routing calls based on either source or ANI information. It can also restrict the access of certain trunks (sources) and certain users (ANI) to a configured trunk group. For example, using SABR allows faxes and modems to be limited to user-specified trunks for connections, as well as restricting the types of calls certain users are allowed to dial, while maintaining full access for others. SABR can allow certain users (hotel guests for example) to be able to only dial certain numbers out a specified trunk group (911 for example) while allowing other users (front desk personnel for example) full access to the trunk group.

Dial Plan Named Timeout

- Configuring Dial Plan Named Timeout allows the user to extend the period of time before a dial plan entry is matched by the switchboard. This will allow for 7 and 10 digit dial plan entries to co-exist on the same system without having to specify special characters for routing the calls. By default, a call is routed as soon as the calling party dials the last matching digit of a dial plan entry. With a Dial Plan Named Timeout defined and applied to a dial plan, the switchboard will wait to route the call when a dialed number is matched to a dial plan until the defined timer has expired.

Enhancement to DSP capabilities

- The 2nd gen 900 series now supports up to 4 simultaneous T38 sessions. The 1st gen 900s are still limited to a single T38 call.
- The 2nd 900/900e series now support up to 30 and 60 DSP resources respectively.

Enhancement to caller-id generation for FXS users

- In previous revisions, caller-id was generated out an FXS user 1000ms after the end of the first ring cycle. A config option was added for A2.03 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.
- Added support for *single data message format* for caller-id.

Added support for DSX trunk audio leveling

- DS0 Leveling attenuates the audio level of received packets before being transmitted out a TDM interface. The direction of leveling occurs in the packet to TDM path only and never in the reverse direction. DS0 Leveling attenuates to a fixed level of -16dBm0, -19dBm0, and -22dBm0.
- The new DSO leveling config options are an extension of the existing alc command configured on the trunk interface. If no level is specified, the default of -16dBm0 will be enabled.

Added support for Virtual Router Redundancy Protocol

- Virtual Router Redundancy Protocol (VRRP) allows load sharing and provides seamless redundancy to networked end-host systems. The result is a fault tolerant, easily managed system where the responsibility for availability is managed by the ADTRAN.

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.

Summary of Bug Fixes

This section highlights major bug fixes in AOS version A2.07.00

Delay in PPPoE session caused by out of order PPP LCP packets

Issue Detail

- If PPP LCP packets were received before PPPoE session confirmation (PADS) messages, the PPPoE session would terminate. This caused a large amount of delay in PPPoE negotiations. This issue has been addressed.

Possible reboot if T38 is negotiated during call setup

Issue Detail

- If the initial INVITE used to setup a call contained T38 parameters, there was a small chance that the ADTRAN would reboot. In most circumstances, T38 is negotiated once a fax call is detected on a line. This issue only occurred if T38 was contained in the first INVITE received by the ADTRAN. This issue has been addressed.

Proxy failover SLA NOTIFY spoofing not working properly

Issue Detail

- When the SIP proxy received a SUBSCRIBE for an SLA line when in failover mode, the NOTIFY spoofed by the ADTRAN may have terminated the SLA subscription instead of granting a line seizure. This issue has been addressed.

NV 6355 only: Ringing not detected on built in FXO port

Issue Detail

- In A2.05.00 or later, the onboard FXO ports would not detect ring voltage in some scenarios with non-standard ring cadences. This issue has been addressed.

MGCP only: Modem calls fail to train at higher speeds

Issue Detail

- When the remote gateway detects a fax/modem call, it will send a modify connection to disable the echo-canceller in the ADTRAN. The ADTRAN, however, would not properly disable the echo-canceller. This may have prevented higher speed modems from connecting. This issue has been addressed.

SIP messages without contact header not matching PUD entry

Issue Detail

- SIP messages received by the SIP proxy in the ADTRAN would not match a SIP proxy user database entry if the message didn't contain a Contact header. A Contact header is normally used for matching with the SIP proxy user database. This issue has been addressed.

No Audio on Nortel CS2K Network Conferencing

Issue Detail

- During a Nortel CS2K 3-way conference, it is possible to receive re-INVITEs back-to-back so quickly that the ADTRAN would respond with a *491 Request Pending* to the second re-INVITE because it was not finished handling the first. The CS2K would correctly handle the retry for the 491, but the media for the call would drop. This issue has been addressed.

Cannot specify which header fields to be used as E.164 number

Issue Detail

- Configuring the field to which E.164 formatting should be applied did not work properly. E.164 formatting was always sent in the To, From, and Contact headers, regardless of the grammar configuration. This issue has been addressed.

MGCP only: Caller-ID sent during call waiting even when no Caller-ID is sent by call agent

Issue Detail

- When a call was received on an endpoint in a call waiting scenario, a blank caller-id number and unknown name was delivered by the ADTRAN even though no caller-id information was presented from the call-agent. This issue has been addressed.

“debug snmp packet” does not provide output on authentication failures

Issue Detail

- Received SNMP packets with invalid authentication credentials were dropped without notification. This resulted in a failure to both generate adequate debug information and SNMP authentication-failure notifications. This issue has been addressed.

MGCP only: Ringback prematurely terminated without a request from the call agent

Issue Detail

- If the ADTRAN received a MDCX without a SignalRequest line while it was already playing a tone such as ringback, the tone was improperly terminated. This issue has been addressed.

Anonymous Caller-ID on SIP to PRI calls not handled properly

Issue Detail

- Inbound SIP calls with “anonymous” sent in the From header were sent to the connected PBX with ”anonymous” incorrectly set as the Calling Party Number IE of the ISDN SETUP message. The calling party name was also sent as “presentation restricted”. This issue has been addressed.

MGCP only: Local ringback fails on port to port calls against CS2K

Issue Detail

- Calls between local FXS ports on the ADTRAN would not hear ringback if the ADTRAN was connected to a CS2K. This issue is unique to the CS2K because of the manner in which it sends MDCX messages. This issue has been addressed.

Possible reboot when running VPN

Issue Detail

- If the ADTRAN entered a state where an inbound security association (SA) did not have a corresponding outbound SA, the unit would reboot. This issue has been addressed.

Hyphen not sent in SIP headers

Issue Detail

- A hyphen contained in the SIP identity of a user would not translate into SIP messages generated by the ADTRAN. This issue has been addressed.

grouped-trunk with 20 character description will cause a reboot when config is saved

Issue Detail

- Trunk groups with names and descriptions greater than 20 characters in length caused the ADTRAN to reboot when the config was saved. This issue has been addressed.

Possible reboot with multicast traffic

Issue Detail

- In certain scenarios, the ADTRAN would reboot if IGMP packets were received on a non-default VRF. The IGMP packets would have to be sent to a broadcast address or one of the ADTRAN's unicast addresses. This issue has been addressed.

Problems with URL filtering prevents Websense from working correctly

Issue Detail

- Downloads from certain HTTP servers would stall and timeout with HTTP URL filtering enabled. This issue has been addressed.

Mu Dynamics security test can cause reboot

Issue Detail

- Several SIP and TCP vulnerabilities were discovered when running the Mu Dynamics security suite against the ADTRAN. These issues have been addressed.

RFC 2833 debug not showing needed information for inbound packets

Issue Detail

- "debug voice dsp voip 0/x y rfc2833" did not work properly for inbound RFC 2833 events. This issue has been addressed.

Reboot due to an authentication race condition on the SIP trunk

Issue Detail

- In a SIP authentication call scenario, if an inbound call was created and cleared immediately, it was possible that the outbound SIP trunk would enter an invalid state when creating the second INVITE with authentication. This issue has been addressed.

503 Service Unavailable error on SIP Trunk page with 17 character authentication password

Issue Detail

- The web GUI would intermittently generate a 503 error when a SIP authentication password was configured that was more than 17 characters long.

503 Service Unavailable error when adding/deleting ANI or DNIS sub

Issue Detail

- The web GUI would generate a 503 error when the user attempted to add or delete an ANI or DNIS substitution entry.

Possible security vulnerability with http secure-server

Issue Detail

- By default, the ADTRAN would allow a browser to fallback to SSL 2.0 if it didn't support SSL 3.0 or TLS 1.0. This caused a security hole due to known weaknesses with SSL 2.0. A change was made to prevent the ADTRAN from falling back to SSL 2.0 unless explicitly enabled in configuration.

Outbound calls could cause a reboot due to memory corruption

Issue Detail

- In some cases, outbound calls could cause a reboot due to memory corruption. The "hotline" command and high call volume may exacerbate the problem. 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 A2.07.00 from the ADTRAN 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 (A2.07.00) 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 filename, and finally the name you want to give the file once it is transferred to the on-board flash. Now from the Configuration prompt:

```
Router(config)#boot system flash <filename.biz> verify
```

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 A2.07.00

Appendix A – Errata for A2.07.00

The following is a list of errata that still exist in A2.07.00

NV 6310 only: Poor audio quality when running G.711 a-Law

When running G.711 a-Law, calls originating or terminating on a TDM endpoint (FXS or PRI) will have poor audio quality. This is not an issue when running G.711 µ-Law.

ISDN trunk not using available B channels

During high traffic conditions, several processes must compete for available trunk resources. In rare cases, it is possible for multiple trunk appearances to reserve the same B-channel. This issue will be addressed in A2.08 and A4.02.

IP 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: g/rt@$). This is also only an issue on hairpin calls.

TA 900eL5 only: Voice quality degradation when running multiple G.729 and T.38 calls

Poor PESQ scores and loss of path confirmation were experienced during testing when running 12 729 E&M to SIP calls plus 4 T.38 faxes. This issue is specific to the 900eL5 only. This issue will be addressed in A4.

1st Gen. TA 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%.

TA 900e only: Channels on 2nd PRI fail to establish voice path

Due to how resources are allocated from the DSPs on the TA 900e, only 39 simultaneous calls will connect over 2 PRI interfaces if the first 23 calls are brought up on T1 0/4. The next 16 calls that connect on T1 0/3, for a total of 39 simultaneous calls, will work correctly. Any subsequent simultaneous calls (more then 39) 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.

1st Gen. 900 only: 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 T1 as local media gateway

The ADTRAN is limited to 18 FXS hairpin calls when the MGCP voice gateway is pointed out the T1 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 TA 900 Series products. The output of “show crypto” displays 200 for IKE and 400 for IPSEC.

Possible MGCP 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.

MGCP only: Analog calls may fail to operate correctly following a PPP link loss

After a PPP link loss and recovery, it is possible that an endpoint will not hear dial tone after going off hook or that the user may pre-maturely hear a busy signal. If calls were attempted from an endpoint while the PPP link was down, that endpoint may not be able to place or receive calls until the ADTRAN is rebooted. This issue exists in all previous versions of firmware.

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. This issue will be addressed in A4.

MGCP Confirmation tone (g/cf) does not work

When the TA 900 receives an S:g/cf to play a confirmation tone, no tones are played out the FXS interface. This issue will be addressed in A4.

Lost packets count 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.

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.

“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. This issue will be addressed in A4.

One-way Audio - Audio Codec Negotiation Problem

If the ADTRAN receives 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. This issue will be addressed in A4.

“debug ip packet vrf <vrf>” provides no output after Fast Flow is 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. This issue will be addressed in A5.

SDP not sent in answer when INVITE is sent without SDP offer

If an INVITE received in B2BUA mode doesn’t contain an SDP offer, the corresponding SDP answer in the “180 Ringing” received from the called party will not be passed through to the calling party.

6355 only: Overhead Paging doesn’t work

Calls to the overhead paging extensions do not work properly.

“Unknown” sent as Call-ID when using VQM reporter

When the ADTRAN records stats for a call in B2BUA mode, the Call-ID is shown as "unknown" in the PUBLISH packet sent to the VQM collector. This issue will be addressed in A4.

Weight isn’t respected for SRV Records with the same priority

When using SRV Records with the same Priority, the Weight should have an affect on balancing which A Record to use. Currently, the first A record received is used 100% of the time. This issue will be addressed in A4.

Ethernet interface is always associated with a bridge group when running IRB

When running integrated route bridging, Ethernet 0/1 is automatically assigned to a bridge group. This prevents the interface from being a routed interface. Bridging and IRB are not currently supported on voice products. This issue will be addressed in A4.

403 sent in response to reINVITE when it should be responding with 481

If a reINVITE is sent to a voice user for an unknown call appearance and the proxy is enabled, the ADTRAN will reply with a *403 Registration Required* instead of an *481 Call Leg/Transaction Does Not Exist*. If the proxy is disabled, the ADTRAN correctly responds with a 481. This issue will be addressed in A4.

MGCP only: fast busy improperly played for off-hook warning tone

When the ADTRAN receives a signal request for off-hook warning tone, a fast busy tone is played instead. This issue will be addressed in A4.

SSH does not permit multiple login attempts against local auth DB
When AAA is on, SSH logins do not allow multiple login attempts against the local database. The user is disconnected on the first or second attempt. If the user is presented with a second attempt, not even the correct credentials will work. This issue will be addressed in A4.

Fax negotiation failure when running T.38 against Lucent LCS gateway
T38 calls against Lucent LCS or Audiocodes gateways may fail to negotiate properly.

Possible issues with TACACS+ Authorization
When trying to use TACACS+ to authorize specific commands, the ADTRAN is sending "TAC+ TX: arg:cmd=version" instead of "TAC+ TX: arg:cmd-arg=version". This issue will be addressed in A4.

Loopback plug on Ethernet port with LLDP enabled causes reboot
If an Ethernet loopback plug is connected to an Ethernet interface on an ADTRAN with LLDP enabled, the unit will reboot. Disabling LLDP prevents the reboot. This issue will be addressed in A4.

Wrong source interface used in SDP answer with multiple frame relay PVCs
SIP Trunks will choose the wrong IP Address to use in the SDP if the contact header host and the via header host are routed out separate sub-interfaces. The ADTRAN will use the host of the Contact, rather than the Via, when searching for a route in order to identify the outbound interface. This issue will be addressed in A4.

“show ip route <ip address>” does not show connected or static routes
The ‘show ip route <ip address>’ command will not show specific connected or static routes for a particular host/network. This issue will be addressed in A4.

TA 900eL2 only: FXO missing impedance values
FXO ports 1-8 on the TA 900eL2 do not have the option for 600r or 900r impedance values. This issue will be addressed in A4.

Jitter buffer average delay reported by VQM is inaccurate
The average jitter buffer delay calculated by VQM is inaccurate due to a problem with the period used to calculate the average. This issue will be addressed in A4.

voip name-service attempts to refresh derived A record that is not listed in the SRV record
The DNS service strips the domain name from SRV targets received from the DNS server when the query to the first listed server times out. The workaround is to remove the “ip domain-name” configuration from the unit. This will be addressed in A4.

Fax/modem detection is not enabled for analog ring group users
Modem-passthrough will not identify modem/fax calls for users in a ring-group. This issue will be addressed in A4.

VQM - GUI - Downloading a large VQM CSV file via the web GUI sends CPU to 100% and usually fails
Downloading a large VQM CSV file via the web GUI will send the CPU to 100% and usually fails. The failure will usually occur at or above 500 VQM streams in memory.

"show qos map interface" doesn't show QoS maps applied to the interface
When trying to display a QoS map applied to an interface with the "show qos map interface <interface>" command, the output will always display "QoS Map not enabled for this interface". This issue is purely cosmetic and does not affect the performance of the QoS map. This issue will be addressed in A4.

TA 900 ADSL always shows unsaved config in N-command MSP
The interop-flag and phy-flag change constantly on the ADSL interface, resulting in the running config always being different that the startup config. This poses a problem when connecting to MSP. MSP compares the running config and the startup config in order to see if there are any unsaved changes. MSP will always categorize the unit as having an unsaved config in the Device Alert Dashboard. This issue will be addressed in A4.

An access class applied to line telnet / ssh doesn't display the number of matches
An access class applied to line telnet / ssh will not show the number of matches in the "show access-list" command. This issue will be addressed in A4.

PRI goes out of service when attempting ISDN to ground start trunk calls on FXO 0/1
When in failover mode, if calls are attempted from a PRI trunk to an FXO trunk that is configured for ground start, the D-channel will drop. This only affects the built in FXO port on the NV 6355 and the TA 900e.

AAA authentication banner not displayed before user and password prompt when connecting via SSH
If a AAA authentication banner is configured, users logging in using SSH will not see the banner when prompted for a login. This same configuration for Telnet users works correctly.

Blind transfers fail when using two IPBGs back to back
With one ADTRAN connected through another ADTRAN IPBG (SIP trunk to SIP trunk), the first IPBG will create unused call legs when the second IPBG performs a blind transfer. Over time, this would eventually cause all available resources on the SIP trunk to be unavailable. This issue will be addressed in A4.

OSPF Configuration link in the GUI gives 404 error
Users trying to access the OSPF configuration page on the web GUI will receive a 404 error, preventing them from configuring OSPF. OSPF can still be configured vial the CLI.

Receiving a 503 Service Unavailable response to an outbound call causes a forward disconnect
If an analog FXS user attempts to place an outbound call and the SIP server responds with a 503 Service Unavailable, the ADTRAN will perform a forward disconnect. This will cause the PBX to disconnect the trunk. The desired behavior would be to play a fast busy for several seconds before the forward disconnect so that the user knows that the service is unavailable.

"show interface t1 0/1 performance total" doesn't show 24-hour totals
In some cases, the 'show interface t1 0/1 performance total-24-hour' command will not show the performance totals for the previous 24 hour period.

“debug snmp packet” truncates T1 threshold traps
Running ‘debug snmp packet’ will only show the first 3 OIDs (and their values) for T1 threshold traps running on a T1 interface.

Removing cos from a user breaks User Accounts page
If a user is removed from a class-of-service using the Classes of Service page in the web GUI, the User Accounts page will no longer work properly. Removing “cos _no-access” from the user account in the CLI will restore access. To prevent this issue, any changes made to the class of service should be done under the User Accounts page.

RFC 2833 events w/ payload type 127 not detected
RFC 2833 events using payload type 127 will not be detected or processed by the ADTRAN.

Invalid character for “Avg Delay” in ‘show voice-quality-stats’
If ‘show voice-quality-stats’ is invoked before there is a enough of a sample size to calculate the average jitter buffer, the value for “Avg Delay” will be an invalid character.

Possible reboot when scrolling through CLI history
Using the ‘up arrow’ key to scroll through the available CLI history could cause a system reboot. The reboot only occurs when scrolling through long lines of text at the command prompt.

Reboot due to invalid RTP packet
A reboot could occur if the ADTRAN receives a packet that is identified as RTP but contains a header that is shorter than the required length.

Cannot configure MTU on Ethernet sub-interface
If 802.1q encapsulation is active on an Ethernet interface, it is not possible to set the MTU for an Ethernet sub-interface.

B-channel not torn down properly when D-channel bounces
B-channels for active calls are not properly cleared when the D-channel dropped on a PRI interface. Service messages should be used to take a B-channel out of service when the D-channel drops.

"debug sip stack messages" is truncated when initiated from the GUI
If a user browses to the Utilities->Debug Unit page in the GUI and adds a filter for ‘sip stack messages’, the SIP messages are partially truncated when displayed.

AAA enable ‘line’ method will not failover to ‘enable’ method if the line password does not exist
‘aaa authentication enable default line enable’ will not fail over to the ‘enable’ password method if the ‘line’ password is not configured (in the case of console or telnet) or if it is not available (in the case of SSH). This means that SSH users cannot pass enable authentication under this configuration.

NV 6310 only: T1 0/1 not configurable for PRI from web GUI
The GUI on the NV 6310 will not allow the user to configure T1 0/1 for PRI. It is, however, configurable via the CLI.

ifCounterDiscontinuityTime is not updated when “clear counters” is issued
When performing a ‘clear counters’ on the ADTRAN, the ifCounterDiscontinuityTime is not updated. This could cause incorrect data to be recorded when monitoring via SNMP.

FTP won't fail over to local authentication database

FTP authentication requests for AAA will not fall back to the local authentication database, even with "local" configured as the fallback method (e.g. 'aaa authentication login default group radius local').

Problem with TCL scripts could lockout console port

Continuously executed TCL scripts may stop running after a certain amount of time and yield the console unresponsive.

Invalid SDP answer when offer contains multiple connection data fields

If the unit receives an initial SDP offer that contains multiple media descriptions (m=) and specific connection data (c=) for each media description, the SDP answer will contain an invalid connection data (c=) field that is missing the IP address for the second media description. This issue will not occur if there is only a single global connection data field.

Appendix B – New and Related Documents

The following are documents related to the new features included in this AOS Release as well as other new documents that have been recently posted to the ADTRAN Technical Support Knowledgebase.

Feature Related Documents

Source and ANI Based Routing
http://kb.ADTRAN.com/article.asp?article=2510&p=2

Enhanced ANI Substitution
http://kb.ADTRAN.com/article.asp?article=2509&p=2

Configuring SPRE code override
http://kb.ADTRAN.com/article.asp?article=3048&p=2

Virtual Router Redundancy Protocol
http://kb.ADTRAN.com/article.asp?article=2155&p=2

Two B-Channel Transfer
http://kb.ADTRAN.com/article.asp?article=3116&p=2

Trunk Group Identifier
http://kb.ADTRAN.com/article.asp?article=3115&p=2