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Release Notes
IP Business Gateways

AOS Release A2.05.00
November 25th, 2009

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 A2.05.00 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, the Netvanta 6310, and the Total Access 900/900e series platforms. Netvanta 7100 release notes are available on the Adtran knowledge base:

Supported Platforms for A2.05.00

- **TA 900 Series** – VoIP Multiservice Access Gateway, single T1 interface
- **TA900e Series** – VoIP Multiservice Access Gateway, multi-T1 interface
- **NetVanta 6310** – VoIP Multiservice Access Gateway, modular WAN
- **NetVanta 6355** – VoIP Multiservice Access Gateway

Summary of New Features

This section highlights the major features, commands and behavioral changes for AOS 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 A2.05.00.

Quality stats display incorrect jitter buffer mode after a fax/modem call

Issue Detail

- Once modem-passthrough detects a fax/modem call, the jitter buffer mode switches to fixed. The output of "show voice quality-stats [id]" displayed the jitter buffer mode as adaptive, regardless of the actual state of the jitter buffer. This issue has been addressed.

Unsupported packetization periods are not rejected

Issue Detail

- If an SDP offer/answer contained a packetization period other then 10, 20, or 30ms (the values currently supported), the Adtran would still respond with a *200 OK*. This could result in one way audio. A change has been made to reject any unsupported packetization period with a *488 Not Acceptable Here*. This issue has been addressed.

Receiving SDP with multiple media fields results in no media

Issue Detail

- If NAT is enabled and SDP is received with multiple media fields (m=), a corrupt firewall association was created resulting in no audio. This issue has been addressed.

Auto-link failure prevents device from connecting with N-command MSP

Issue Detail

- In some cases, auto-link would not connect properly with MSP. This was caused when the volume identifier was appended to a longer than average firmware image name (e.g. T900G2A-A2-03-00-SC-E.biz) in the boot config, resulting in a string longer than 32 characters. This issue has been addressed.

GUI: opening system summary page automatically clears CPU max load

Issue Detail

- Clicking on the system summary page in the GUI would reset the max CPU utilization. This issue has been addressed.

Unattended transfer with analog user fails to send REFER

Issue Detail

- An unattended transfer, or transfer on hangup, would fail if the hang up was initiated before the 18x provisional response was received from the soft switch. Instead of sending a REFER, a CANCEL was sent, and the original call leg was torn down. This issue has been addressed.

MGCP only: Caller-id name delivery sent with quotations

Issue Detail

- When using MGCP, the calling party name field of the MDMF message was sent in quotations. This may cause caller-id detection problems on some caller-id devices. This issue has been addressed.

Invalid firewall association causes reboot

Issue Detail

- The Adtran may reboot if a firewall association for RTP was created at the same time that the layer one interface (T1, Ethernet, ADSL) dropped. It is highly unlikely for this issue to be seen in the field. This issue has been addressed.

Loose routing parameter not added to Record-Route header for SIP traffic through proxy

Issue Detail

- In stateful and outbound proxy modes, SIP proxy failed to add the ";lr" parameter for Record-Route header. This issue has been addressed.

"show tech terminal" doesn't produce a show tech file

Issue Detail

- The 'show tech terminal' and 'show tech' commands create the file showtech.txt. The output for 'show tech terminal' was discarded, resulting in showtext.txt failing to be written to the flash. This issue has been addressed.

Disconnect on the trunk account during a reINVITE causes a reboot

Issue Detail

- There was a possible race condition where the unit would reboot if the SIP trunk account was generating a reINVITE at the same time as it received a BYE to tear down the call. This issue has been addressed.

Firewall association for RTP stream removed through Proxy when SDP is present in an ACK

Issue Detail

- If FastFlow and NAT were enabled, the inclusion of SDP in an ACK would cause a disassociation between the firewall and the FastFlow entries. This would result in no media through the firewall. This issue has been addressed.

Stateful Proxy will not resend SIP messages if a reINVITE is received during setup

Issue Detail

- If a reINVITE was received during call setup before the initial transaction was complete, SIP proxy would terminate the initial call leg. This issue has been addressed.

maddr: field used for legacy Record-Route results in failed call

Issue Detail

- If a 200 OK used the "maddr=" parameter in the contact URI, the corresponding ACK was sent back to the Adtran instead of the remote SIP device. The "maddr=" field for legacy Record-Route is now supported. This issue has been addressed.

Invalid login credentials for AAA causes a memory leak

Issue Detail

- When trying to login using the craft port, a small amount of memory was leaked each time a AAA session was dropped for invalid login credentials. Over an extended period of time, this could cause a reboot. This issue has been addressed.

MGCP only: Ringback tone generation fails when using g/rbk

Issue Detail

- If the call agent sent a signal request to the Adtran for g/rbk after the initial call dialog had been established, the Adtran would not play ringback. This issue was not present if g/rt was used for ringback generation. This issue has been addressed.

“dial-string source to” causes memory leak

Issue Detail

- If ‘dial-string source to’ was configured on a SIP trunk, a memory leak would occur that could cause the Adtran to reboot. This issue has been addressed.

D-channel locks up on PRI interface

Issue Detail

- The D-channel on a PRI trunk could disconnect and lock up if the SIP trunk received a 180/183 with SDP immediately (ms) followed by a 4xx level message. This particular call flow has only been identified against a Nortel CS2K when calling an invalid number. This issue has been addressed.

NV 6355 platform displays as “BAD” in “show version”

Issue Detail

- When issuing a ‘show version’, the platform version would display as “BAD”. The hardware ID for new FXO hardware was not updated in firmware loaded prior to shipment. This issue has been addressed.

“ip sip proxy failover match-digits” not matching called party number

Issue Detail

- When in failover mode, SIP Proxy would not match a 10-digit request URI to a SIP Proxy user that had registered using 4-digits when using the "ip sip proxy failover match-digits 4" command. This issue has been addressed.

End packets not sent for RFC 2833 talk-off events

Issue Detail

- When using RFC 2833, the Adtran would not send end packets for DTMF talk-off events. This resulted in some gateways continuing to play the talk-off event for the duration of the call. This issue has been addressed.

Proxy survivability fails when using a port other than 5060 for SIP

Issue Detail

- When in proxy failover mode, calls sent to the Adtran would fail if the UDP port for SIP was changed to a value other than 5060. The proxy was using the UDP port in the contact header instead of the configured UDP port. This issue has been addressed.

A simultaneous disconnect by SIP and PRI user causes reboot

Issue Detail

- If both ends of a SIP to PRI call hung up simultaneously, there was a possible race condition where the ISDN interface would attempt to clear ISDN call variables that had already been cleared. This would result in a reboot. This issue has been addressed.

Adding a description to PPP interface prevents SNMP alias from being set

Issue Detail

- If a description was configured on a PPP interface, the CLI would accept a command to set the alias but it would not appear in the running-config or SNMP interface walk. This issue has been addressed.

PRI: No Disconnect message is sent before bringing down the D-channel

Issue Detail

- If a change was made to the configuration of a PRI interface that had active calls, the Adtran disconnected the calls and bounced the D-channel with no indication given to the SIP server. A BYE is now sent to the SIP server. This issue has been addressed.

DNIS match/sub to blank number is ignored

Issue Detail

- When trying to match a DNIS number and substitute it with no number to prevent digits from being played out an interface, the “” used as the substitution was ignored. This issue has been addressed.

Interval in “voip name-service name-table’ not updated correctly

Issue Detail

- The interval field in the ‘show voip name-service name-table’ never changed even though the actual entry timed out correctly. This issue was purely cosmetic. This issue has been addressed.

Replacing/deleting a tracked TCL script causes lockup

Issue Detail

- The Adtran would lock up when trying to edit or delete a TCL script that was to be executed based on a track. This issue has been addressed.

Network conferencing-mode mode missing from NV 6355

Issue Detail

- The "voice conferencing-mode network" command was missing from the NV 6355. The default operation for the NV 6355 should be network conference, similar to the TA 900 series. A change has been made to make network the default conferencing mode. This issue has been addressed.

Changing resource selection on a PRI interface connected to multiple trunk groups causes a reboot

Issue Detail

- Changing the resource selection for a single PRI interface that was connected to multiple trunk groups would result in a reboot. This issue has been addressed.

MGCP ground start fails to release line from Tip Group on DLCX

Issue Detail

- On a ground start call using RFC2833 signaling, the FXS port would not correctly release the tip ground at the end of a call. This caused the line in the PBX to be remain off-hook. This issue has been addressed.

MGCP fax calls may fail if echo-cancelation is not disabled via local connection options

Issue Detail

- Previously, the Adtran would only disable the echo-canceller if directed to do so in the local connection options (LCO). Now, if a fax or modem tone is detected, the echo canceller will be disabled, regardless of the state received in the LCO. This issue has been addressed.

Hairpin calls between proxy users can cause lockup in firewall

Issue Detail

- When hairpin calls were made between two transparent proxy users and VQM was enabled, the ports used to advertise each user's RTP stream to the public softswitch were locked in the firewall. This would prevent the user from placing calls until the Adtran was rebooted. This issue has been addressed.

First inbound call fails if FQDN for SIP server and MGCP call agent are the same

Issue Detail

- If the same hostname is used for both the SIP server and the MGCP call agent, the SRV records for SIP weren't refreshed and would expire from the host table. This resulted in a call failing when the name service client couldn't resolve the SIP server from the local cache. This would only affect the first call after the SRV records associated with the SIP server timed out. This issue has been addressed.

Firmware upgrade fails when using N-command MSP

Issue Detail

- Firmware update jobs in MSP could fail due to the AOS device not responding to a TCP keep alive message from the MSP server. This issue has been addressed.

MGCP failover does not revert back to primary call-agent

Issue Detail

- If the Adtran had a primary and secondary call-agent statically defined, it would correctly failover to the secondary server if the primary server became unresponsive. When the primary server came back online, however, the 900 would not fall back to the primary and would continue to send traffic to the secondary server. 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 A2.05.00 from the public FTP server. 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 (A2.03.00) 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 A2.05.00

Appendix A – Errata for A2.05.00

The following is a list of errata that still exists in A2.05.00

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.

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.

NV 6310 only: IAD reboots when invalid or incorrectly formatted ECT ISDN messages are received

Invalid or incorrectly formatted ETSI PRI channel transfer (ECT) messages that should result in a reject will cause the NV 6310 to reboot.

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 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.

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.

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 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, 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.

6355 only: Overhead Paging doesn’t work
Calls to the overhead paging extensions do not work properly. 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 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. This issue will be addressed in A4.

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. This issue will be addressed in A4.

"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. 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 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.

"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.

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. This issue will be addressed in A4.

URL filter does not relay DNS reply
With URL filtering enabled, the Adtran will buffer any replies from a DNS server until it receives an allow message from the Websense server. Once the response from the Websense server is received, the buffered reply is dropped by the Adtran instead of being sent to the PC. The next time the PC queries the URL, the reply is sent correctly. This issue will be addressed in A4.

403 sent in response to reINVITE when 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.

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.

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 a Lucent LCS gateway 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".

Cannot configure a TCP VQM reporter

A VQM collector configured for TCP will always use UDP for PUBLISH messages sent to N-command MSP. This issue will be addressed in A2.06.

Two B-Channel transfer initiated after a forward or unattended transfer will fail to a Broadsoft

If a call originating from the PSTN is forwarded by a PBX connected off the PRI of the Adtran, the calling party number in the From: header will be a PSTN number. The diversion header is used to include the redirecting number field from the PBX. Broadsoft uses the number in the diversion header to authenticate the user, but it does not use the number in the diversion header to anchor the call leg. When a REFER is sent from the Adtran to transfer the two parties together, Broadsoft will reject the REFER. In order for the Broadsoft to recognize the second call leg as being anchored from the diversion header user, the SIP header must contain the transferring party as a user in one of several other header fields, of which one is p-asserted identity. In A2.06, Adtran will include this user in the p-asserted identity field. This is an interop issue between Adtran and Broadsoft. All attended transfers work correctly.

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 900eL3 only: FXO missing impedance values

FXO ports 1-8 on the TA 900eL3 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. 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 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.

Calling party number delivered as "unknown" in MDMF

If the CID number received in an INVITE is "Unknown", then the text string "Unknown" is sent to the connected TDM device instead of the proper reason-for-absence code. This could cause a display issue with certain Caller-ID devices. This issue will be addressed in A2.06.

“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.

Failure to fall back to G.711 on a PRI trunk after receiving a *488 Not Acceptable Here*

If the soft switch responds to a T.38 request by the Adtran with a *488 not acceptable here*, the Adtran will respond by tearing down the call with a BYE. This only affects SIP to PRI calls. This issue will be addressed in A2.06.

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.

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.

DNS queries to secondary server with no valid route causes a lockup

If the configured primary DNS server stops responding to queries, the Adtran will failover to the secondary DNS server. If there is no route to the secondary DNS server present in the route table, the Adtran will lockup when trying to query the secondary server. If both DNS servers are routable by the default route in the Adtran, this issue will never occur.

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