ADTRAN The Network Access Company

Corporate Office
ADTRAN, Inc.
901 Explorer Boulevard
Huntsville, AL 35806

U.S. Mail
P.O. Box 140000
Huntsville, AL 35814-4000

General Information
800 9ADTRAN
info@adtran.com
www.adtran.com

Pre-Sales
Technical Support
800 615-1176
application.engineer@adtran.com
www.adtran.com/support

Post-Sales
Technical Support
888 423-8726
support@adtran.com
www.adtran.com/support

ACES Help Desk
888 874-ACES
aces@adtran.com
www.adtran.com/support

Release Notes
IP Business Gateways

AOS Release A4.02.00
August 16th, 2010

Contents

Purpose and Supported Platforms

AOS Voice Products release A4 is a major system release that adds many new features and addresses customer issues that have been uncovered in previous code releases.

Release A4.02.00 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 and NetVanta 6300 series platforms.

Supported Platforms for A4.02.00

- **TA 900 Series** – VoIP Multiservice Access Gateway, single T1/ADSL interface
- **TA900e Series** – VoIP Multiservice Access Gateway, multi-T1 interface
- **NetVanta 6300 Series** - VoIP Multiservice Access Gateway, modular WAN

Summary of New Features

This section highlights the major features, commands and behavioral changes for AOS A4.01.00.

SIP Diversion and P-Asserted-Identity Header Enhancements

- When an ADTRAN SIP device is fronting a PBX, and the PBX does not support Redirecting Number (because it uses a call control other than PRI or because its PRI implementation does not support Redirecting Number), the ADTRAN SIP device previously had no way of adding an alternate identity header when it was needed by the softswitch to authenticate the origin of the call. In some situations, it is desired that a Diversion header or a P-Asserted-Identity header be added to every outbound call. In some cases, it may be useful to add a Diversion header or a P-Asserted-Identity header to an outbound call only when the Caller-ID of the call is not recognized as a number local to the PBX.

Templated Proxy Users

- Added support for endpoints or IP PBXs behind the 900 that do not register back to the softswitch or can only register one user for all phones. It is possible to create proxy "users" using the same wildcard methods used in accept templates on grouped trunks.

Increase Proxy BLF Support in IPBGs from 4 to 50 Lines

- Added support for up to 50 BLF users for customers who have phones that support higher BLF capability than four lines.

Ability to Specify Multiple SIP Signaling Ports

- Added the ability to listen on multiple SIP ports. This will allow the user to specify a unique SIP port for transparent proxy.

Additional Features merged from AOS 17.05.02

- VRRP
- VQM MIB
- Enhanced QoS and supporting MIB
- NQM MIB
- VAP synchronization for multiple NV 150 configurations
- Switchport scheduler for PoE interfaces
- Multicast support
- TWAMP and NTP
- LLDP-MED
- Enhanced QoS and traffic shaping for both WAN and Ethernet interfaces

Summary of Bug Fixes

This section highlights major bug fixes in AOS version A4.02.00

Reboot when using “show ip mroute” command

Issue Detail

- Intermittent reboots had been observed when using the ‘show ip mroute’ command after a routing update on a unit running PIM Sparse Mode and OSPF. This issue has been addressed.

Failures using DPT mode on analog FXO trunks

Issue Detail

- Applications with moderate to heavy load would potentially see calls fail when using DPT mode for FXO trunks. This issue has been addressed

Reboot when “show run | begin” command is issued

Issue Detail

- The ADTRAN would reboot when the command “show run | begin” was issues if a 20 character description was configured on a grouped-trunk. This would also cause a reboot if the configuration was saved. This issue has been addressed.

TA 900/900eL2 only: FXO port impedance values won’t initialize after reboot

Issue Detail

- In some cases, the FXO port would not initialize the correct impedance values after the ADTRAN was reset. This issue would remain until the unit was rebooted or a "shut" / "no shut" sequence was performed on the affected ports. This issue has been addressed.

Reboot due to invalid RTP packet

Issue Detail

- A reboot might occur if the ADTRAN received a packet that was identified as RTP but contained a header that was shorter than the required length. This issue has been addressed.

T38 reboot after receiving 200 OK with no media attributes

Issue Detail

- A reboot might occur if the ADTRAN received a 200 OK with T38 specified but no other media attributes defined. This issue has been addressed.

"debug sip stack messages" is truncated when initiated from the GUI

Issue Detail

- If a user browsed to the Utilities->Debug Unit page in the GUI and added a filter for ‘sip stack messages’, the SIP messages would be partially truncated when displayed. This issue has been addressed.

One-way Audio due to Codec Negotiation Problem

Issue Detail

- If the ADTRAN received an SDP where the codec preference order in the media field had DTMF relay before G.711 or G.729 (i.e. m=audio 31794 RTP/AVP 101 0), media wasn’t sent properly, resulting in one way audio. This issue has been addressed.

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

Issue Detail

- If the ADTRAN received a signal request for off-hook warning tone, a fast busy tone was played instead. An off-hook warning tone (a.k.a howler tone) is now correctly played. This issue has been addressed.

Radios / VAPs page in GUI returns 404 not found

Issue Detail

- Navigating to the ‘Radios/VAPS’ page in the GUI would result in a ‘404 Not Found’. This issue has been addressed.

Fax tone list commands broken

Issue Detail

- If the ‘fax-tone’ command was used to create a fax tone list for modem-passthrough or T38, the commands entered into the CLI to create the list would not be added to the running config and therefore would not take effect. This issue has been addressed.

NTP associations do not appear in the GUI

Issue Detail

- NTP associations wouldn’t appear in the GUI when the "view associations" box was checked. The text "There are no associations." appeared even though there were associations available. The synchronization statistics appear correctly in the GUI. The associations could be viewed via the CLI. This issue has been addressed.

Proxy Template BYE is sent in the wrong direction

Issue Detail

- If a BYE message without a contact header was sent through the SIP proxy with the proxy configured using proxy user templates, the ADTRAN would incorrectly forward the message to the originator of the BYE. This issue has been addressed.

Caller-id name for loopback accounts does not save properly

Issue Detail

- Quotations were not saved properly when configuring a Caller-ID name on a voice loopback account. This could prevent the name from restoring on reboot if the name contained a space. This issue has been addressed.

BYE not matching proxy user database entry

Issue Detail

- A BYE would not match a proxy user database entry if the BYE didn’t contain a Contact header. A Contact header is normally used for matching with the proxy user database. 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 A4.02.00 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 (A4.01.00) separated by hyphens instead of decimals.

From the 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 primary boot image. The router should now be rebooted with the privileged command

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

When the unit reboots, it will be running AOS version A4.02.00

Appendix A – Errata for A4.02.00

The following is a list of errata that still exist in A4.02.00

Field Issues

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.

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 A4.03.

“show 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.

“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.

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.

6355 only: Overhead Paging doesn’t work
Calls to the overhead paging extension 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 A4.03.

Ethernet interface is always associated with a bridge group when running IRB
When running integrated routing and 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.

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.

Fax negotiation failure when running T.38

T38 calls against Lucent LCS or Audiocodes gateways may fail to negotiate properly.

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.

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 then routed 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.

Reboot if T.38 is negotiated during call setup

If the initial INVITE used to setup a call contains T.38 parameters, there is a small chance that the ADTRAN will reboot. In most circumstances, T.38 is negotiated once a fax call is detected on a line. This issue only occurs if T38 is contained in the first INVITE received. This issue will be addressed in A4.03.

Proxy failover SLA NOTIFY spoofing not working properly

When the proxy receives a SUBSCRIBE for an SLA line when in failover mode, the NOTIFY spoofed by the ADTRAN may terminate the SLA subscription instead of granting a line seizure. This will be addressed in A4.03.

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

In A2.05.00 or later, the onboard FXO ports may not detect ring voltage if the ring cadence is not the standard 2s on, 4s off. This issue will be addressed in A4.03.

MGCP only: Modem calls fail to train at higher speeds

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, will not properly disable the echo-canceller. This could prevent higher speed modems from connecting. This issue will be addressed in A4.03.

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.

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.

Websense client fails to communicate with server

Problems with the buffered server packet accounting within URL filtering prevents the Websense client from working properly. This issue will be addressed in A4.03.

No Audio on Nortel CS2K Network Conferencing

During a Nortel CS2K 3-way conference, it is possible to receive re-INVITEs back-to-back so quickly that the ADTRAN responds with a *491 Request Pending* in response to the second re-INVITE because it has 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 will be addressed in A4.03.

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

Currently, configuring the field to which E.164 formatting should be applied does not work properly. E.164 formatting will always be sent in the To, From, and Contact headers, regardless of the grammar configuration. This issue will be addressed in A4.03.

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

When a call is received on an endpoint in a call waiting scenario, a blank caller-id number and unknown name is sent even though no caller-id information was presented from the call-agent. This issue will be addressed in A4.03.

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

Received SNMP packets with invalid authentication credentials are dropped without notification. This results in a failure to both generate adequate debug information and SNMP authentication-failure notifications. This issue will be addressed in A4.03.

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

If the ADTRAN receives a MDCX without a SignalRequest line while it is already playing a tone such as ringback, the tone will be improperly terminated. This issue will be addressed in A4.03.

“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 properly show the performance totals for the previous 24 hour period.

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

Inbound SIP calls with *anonymous* sent in the From header are sent to the connected PBX with ”anonymous” incorrectly set as the Calling Party Number IE of the ISDN SETUP message. The calling party name is also sent as “presentation restricted”. This issue will be addressed in A4.03.

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

Calls between local FXS ports on the ADTRAN will not hear ringback if the ADTRAN is connected to a CS2K. This issue is unique to the CS2K because of the manner in which it sends MDCX messages. This issue will be addressed in A4.03.

“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.

Possible reboot when running VPN

If the ADTRAN enters a state where an inbound security association (SA) does not have a corresponding outbound SA, the unit may reboot. This issue will be addressed in A4.03.

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').

Hyphen not sent in SIP headers

A hyphen contained in the SIP identity of a user does not translate into SIP messages generated by the ADTRAN. This issue will be addressed in A4.03.

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.

503 error when adding/deleting ANI or DNIS substitution

The web GUI could generate a 503 error when the user attempts to add or delete an ANI or DNIS substitution entry. This issue will be addressed in A4.03.

503 error on SIP Trunk page with 17 character authentication password

The web GUI will intermittently generate a 503 error when a SIP authentication password is configured that is more than 17 characters long. This issue will be addressed in A4.03.

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. As a result, the reserved B-channel will not be available until a "shut" / "no shut" sequence has been applied to the PRI interface.

Mu Dynamics security test can cause reboot

Several SIP and TCP vulnerabilities were discovered when running the Mu Dynamics security suite against the IPBGs. These issues will be addressed in A4.03.

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

If PPP LCP packets are received before PPPoE session confirmation (PADS) messages, the PPPoE session will be terminated. This could cause a large amount of delay in PPPoE negotiations. This issue will be addressed in A4.03.

Outbound calls could cause a reboot due to memory corruption

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 will be addressed in A4.03.

Voice Quality Stats show invalid characters

If a call duration is extremely short (less than a couple of seconds), the delay in "show voice quality-stats" could be displayed as "-(". This issue will be addressed in A4.04.

"no-alg" config parameter not showing up in config

The 'no-alg' parameter will not be applied to an Access Control Policy entry even though the command is accepted. This issue will be addressed in A4.04.

Problems with processing REFER from Netvanta UC

A race condition occurs when the Netvanta UC Server sends a BYE immediately upon receiving a NOTIFY. This has the potential for a trunk account being transferred to tear down the call before initiating the new one. This issue will be addressed in A4.04.

Possible reboot with multicast traffic

In certain scenarios, the ADTRAN will reboot if IGMP packets are 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 will be addressed in A4.03.

RFC 2833 debug not showing needed information for inbound packets

'debug voice dsp voip 0/x y rfc2833' will not work properly for inbound RFC 2833 events. This issue will be addressed in A4.03.

Reboot due to an authentication race condition on the SIP trunk

In a SIP authentication call scenario, if an inbound call is created and cleared immediately, it is possible that the outbound SIP trunk would enter an invalid state when creating the second INVITE with authentication. This issue will be addressed in A4.03.

Possible security vulnerability with http secure-server

By default, the ADTRAN will allow a browser to fallback to SSL 2.0 if it doesn't support SSL 3.0 or TLS 1.0. This causes a security hole due to known weaknesses with SSL 2.0. A change will be made to prevent the ADTRAN from falling back to SSL 2.0 unless explicitly enabled in configuration. This issue will be addressed in A4.03.

NV 6310/6330 only: Reboot when shutting down efm-group

If all links are removed from an efm-group, and the efm-group is then administratively shutdown, the ADTRAN will reboot. If at least one link is present in the efm-group, the ADTRAN will not reboot. This issue will be addressed in A4.03.

PRI interface does not allow "connect" statement without switch-type defined

If a new PRI interface is configured, it can not be connected to a T1 until an ISDN switch-type is configured. Not only can it not be connected, but "connect" is not even an available option. This is a minor issue since a switch-type is required for a working configuration. This issue will be addressed in A4.03.

ISDN SETUP message with restricted number causes display error in debug

If the calling party number is restricted and the calling party name is restricted with a value of "anonymous", the isdn l2-formatted debug won't display anything in the SETUP message after the calling party name information. This is purely cosmetic and doesn't affect the SETUP message in any way.

Can not assign media-gateway to loopback interface from GUI

If a media-gateway is applied to a loopback interface, the GUI will display that the loopback is selected as the address type but it will not save this value once it is applied.

Nortel 3-way conferencing with conferencing-uri fails

Network conferencing against a Nortel CS2K will fail when using a conferencing URI.

Problem with processor utilization during VPN tunnel re-negotiation

VPN tunnel re-negotiation can cause the packet routing processor queue to spike to 100%. This spike in processor utilization could cause momentary voice quality issues.

Fax failures due to SIP glare

Calls may fail if the ADTRAN receives a re-INVITE at the same time it is attempting to re-INVITE the same call. This is known as SIP glare and can occur with modem-passthrough and T.38 re-INVITES since both sides may attempt to re-INVITE the call simultaneously. This can be worked around by changing the 'voice modem passthrough mode' to either inbound or outbound depending on which call direction you wish the unit to detect modem or fax tones. The default is both.

In survivability mode, calls from a private extension to SLA behind transparent proxy fail to stay connected

Calls can fail between a private extension on one phone and an SLA/BLA line on a second phone when both phones are behind transparent proxy in survivability mode. When the private extension dials the SLA extension, the SLA phone answers the call but is disconnected within a few seconds.

Unable to configure a startup-delay greater than 35 seconds

If a VRRP startup-delay is configured for more than 35 seconds, the timer will still expire in 35 seconds. If the delay is configured for less than 35 seconds, the timer will expire at the configured time.

Transparent Proxy doesn't work with TCP

Transparent mode for the proxy doesn't work when the phones are configured to use TCP.

NTP master command does not restore correctly

The CLI command 'ntp master' will not be restored after a 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 IPBG 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. 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%.

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 affects 1st gen TA 900 series products.

Issue with network conferencing against Broadsoft

With the ADTRAN set for "voice flash hook mode transparent," the conference originator must wait for the third party to answer before executing the flash hook to initiate the conference.

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.

Unresolved DNS queries generated by the IAD for VoIP Name Resolution create policy sessions that do not time out

Unresolved DNS queries generated by the IAD for VoIP Name Resolution do not time out per ' show ip policy-session' output. Over an extended period of time, the number of sessions continually increase until the DNS queries are resolved.