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Release Notes
IP Business Gateways

AOS Release A2.04.00
August 3rd, 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 A2.04.00 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 A2.04.00

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

Summary of New Features

This section highlights the major features, commands and behavioral changes for AOS 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 A2.04.00.

CID does not work when fxs impedance is set to 600r

Issue Detail

- In rare cases, Caller ID tones were attenuated below specification when an FXS port was configured for the 600r coefficient. This caused Caller ID to not be detected by the station equipment. This behavior is only observed when using 600r. This issue has been addressed.

Transparent proxy updates incorrect user in database when IP address of user changes

Issue Detail

- Duplicate proxy user database entries created by a user registration with a new IP address caused the proxy to select the wrong entry when updating the database. In this case, responses to registration messages were sent to the original IP address instead of the newly updated IP address. Manually clearing the proxy user database would resolve the issue. This issue has been addressed.

ISDN B channel stuck in active/restarting

Issue Detail

- If a b-channel isn't idle within 60 seconds after a call is terminated, the Adtran will proactively re-start the b-channel. In rare cases, the recovery timer used to restart the b-channel would hold the channel in the restart state, preventing that channel from being used. Resetting the PRI interface would resolve the issue. This issue has been addressed.

Problems with VRRP when transitioning from backup to master

Issue Detail

- When the Adtran transitioned from the backup router to the primary router, traffic destined for the virtual IP/MAC associated with the VRRP group would be dropped and not forwarded. This issue has been addressed.

Proxy user database not updated properly when relevant contact binding isn't listed first

Issue Detail

- A 200 OK response from the registrar contains a list of Contact fields showing all current bindings. If the Contact field contained more then one binding, the proxy user database would only update the registration request for the first entry listed. This issue has been addressed.

302 Message Fails to Redirect call

Issue Detail

- A 302 message used to temporarily redirect SIP traffic to a secondary server would cause the call to be torn down in the Adtran. This issue has been addressed.

Out of memory reboot due to SNMP memory leak

Issue Detail

- In rare cases, a memory leak related to SNMP could have caused the unit to reboot. This issue has been addressed.

"ip sip timer registration-failure-retry" is not maintained after a reboot

Issue Detail

- The command "ip sip timer registration-failure-retry" wasn't maintained after a reboot. This issue has been addressed.

900e only: Ethernet Interface stops forwarding frames after multiple collisions

Issue Detail

- If one of the Ethernet interfaces on the 900e was in half duplex and taking collisions, the Ethernet transmitter would stop forwarding frames until the unit was rebooted. This issue has been addressed.

MGCP only: Reboot caused by Video SDP

Issue Detail

- If the call agent sent an SDP to the Adtran that contained connection information for a video endpoint, the unit would reboot. This issue has been addressed.

200 OK to INVITE with sendonly media does not contain SDP attribute for recvonly or inactive

Issue Detail

- When an INVITE for an inbound call was received that contained an SDP attribute for a sendonly media stream, the corresponding SDP answer sent by the Adtran did not contain an SDP attribute for recvonly or inactive. This issue has been addressed.

Firewall incorrectly matches DNS queries from PCs on the LAN with queries initiated by the Adtran

Issue Detail

- If a DNS query sent from a PC on the LAN through the Adtran was assigned a UDP NAT port that was the same as the UDP port that the 900 had previously bound to a socket for DNS, the firewall in the Adtran would think that the DNS reply was destined for itself. This would prevent the reply from making it back to the PC. This issue has been addressed.

Secure socket reboot

Issue Detail

- In rare cases, HTTPS, SSH, or VPN could trigger an SSL related reboot. This issue has been addressed.

A-record hosts present in SRV record times out

Issue Detail

- If an SRV record response contained additional A-records that were not aliases for the SRV record, the derived A-records would not be refreshed properly. A change was made to properly parse all additional A-records associated with a SRV record response. This issue has been addressed.

When using SRV records, the Adtran does not try a secondary server if the primary does not respond

Issue Detail

- When using SRV records with a primary and backup server, if the primary server does not respond, the Adtran does not retry using the backup server. This issue has been addressed.

Data LED only References PPP Interface

Issue Detail

- A change was made to include the status of Frame Relay and HDLC interfaces with the data LED. This issue has been addressed.

Unable to call Proxy phones in a ring-group

Issue Detail

- In a survivability scenario where proxy users are assigned to a ring group, a call from an FXO or PRI trunk will ring all of the proxied phones correctly. When a proxied phone answers an inbound call, there would be no talk path. This issue has been addressed.

Copying a TFTP config over running-config disables PRI

Issue Detail

- Copying a config that connected an isdn-group to an existing PRI would result in a crossconnect to an invalid tdm-group. This issue has been addressed.

MGCP only: Notification Requests not handled as line package events for NCS

Issue Detail

- In NCS, digits are sent as line package events as opposed to DTMF package events with MGCP 1.0. The Adtran would only handle a Notification Request for digits when the DTMF package was used. This issue has been addressed.

MGCP only: FQDN as MGCP local-domain-name causes packets to improperly use 255.255.255.255 as the layer 3 source address

Issue Detail

- Using an FQDN as MGCP local-domain-name caused packets to use 255.255.255.255 as the layer 3 source address. This issue has been addressed.

Problems with alias command on PPP interface

Issue Detail

- The "alias link" command on a PPP interface had no effect on the running configuration. This issue has been addressed.

SNMP traps not sent to all defined servers

Issue Detail

- When a unit was configured with multiple SNMP servers, traps were not sent to all of the defined servers. This issue has been addressed.

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

Issue Detail

- If a unit configured for stateful proxy was in failover mode, calls would failover to the backup interface until the DNS entry associated with the proxy user timed out. This issue has been addressed.

Proxy changes TO user in SUBSCRIBE message, causing BLA subscription to fail

Issue Detail

- BLA subscriptions would fail through the proxy if the first line on a phone is a private extension and the second line is a BLA line. This problem only occurred if the phone was not the first phone to register through the proxy. This issue has been addressed.

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

Issue Detail

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

MGCP only: Failover does not revert back to primary call-agent

Issue Detail

- When the Adtran fails to receive a response from the primary call-agent, it correctly fails over to the secondary call-agent. If the primary and secondary servers were using the same FQDN in the notified entity field, the Adtran would continue sending messages to the secondary server, regardless of whether or not the primary call-agent came back online. This issue has been addressed.

Proxy matches wrong proxy user database entry on outbound NOTIFY from Aastra phones

Issue Detail

- NOTIFY message from Aastra IP Phones were matching the wrong proxy user database entry. This was due to the fact that the Aastra phones only include the transport parameter in the contact field of REGISTER requests. This resulted in Aastra not receiving BYE and failure of line seizure notifications for BLA lines. This issue has been addressed.

Proxy survivability changes Contact domain on non-INVITE messages

Issue Detail

- During station to station survivability in transparent proxy mode, the Adtran incorrectly changed the Contact field for all non-INVITE messages. The Contact domain in the INVITE changed from one IP to another, but the Contact domain for non-INVITE messages changed from an IP address to a FQDN. This issue has been addressed.

Call forwarded from an IP phone fails through the proxy

Issue Detail

- When a call was forwarded by an IP phone, the stateful proxy in the Adtran was removing the forwarded number from the contact field in the “302 Moved Temporarily”. This issue has been addressed.

SIP Proxy reboot with "ip sip proxy grammar proxy-id contact-user"

Issue Detail

- With the ‘ip sip proxy grammar proxy-id contact-user’ option configured, the proxy appends the proxy-id in the contact following the first occurrence of the "@" symbol. If that symbol doesn’t exist in the contact field, a reboot would occur during this operation. Using ‘ip sip proxy grammar proxy-id contact-parameter’ is an acceptable workaround as the “;pname=pvalue” value for the proxy-id is correctly appended it to the end of the contact. 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 A2.03.00.SC 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 A2.03.00.SC

Appendix A – Errata for A2.04.00

The following is a list of errata that still exists in A2.04.00

MGCP Confirmation tone (g/cf and l/cf) does not work

When the TA 900 receives an S: g/cf or an S: l/cf to play a confirmation tone, nothing is played out. This issue will be addressed in A4.

SIP to MGCP Ringback issue

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

1st gen 900/900e only: Possible issue with DTMF generation under heavy call load

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

6355 only: Overhead Paging doesn’t work

Calls to the overhead paging extensions do not work properly.

T1 in Yellow Alarm Causes 503 on System Summary page of GUI

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

RTP is not allowed through the firewall when NAT is performed on inbound calls to a SIP user with no SDP in the INVITE

If the SDP for an inbound call is not sent until the SIP Server ACKs the 200 OK when the called party answers, the firewall will not open a hole for RTP, resulting in no audio. 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.

Jitter Buffer Mode shows adaptive after modem-passthrough detects a data call

After modem-passthrough has switched the user to data mode, the jitter buffer mode still shows adaptive instead of fixed under "show voice quality-stats [id]". This is purely a cosmetic error. 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.

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.

“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.

access class applied to line telnet / ssh not displaying number of matches in show access-lists

“show access-list” doesn’t show the number of matched packets for a standard ACL applied to a line telnet or line ssh access class. 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.

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

If a change is made to the configuration of a PRI interface that has active calls, the Adtran disconnects the calls and bounces the D-channel. No indication is given to the SIP server that calls were disconnected. 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 Websence 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.

“show tech terminal” does not function properly

When issuing the “show tech terminal” command, the tech file isn’t written to flash. This will be addressed in the next release of A2.

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

When a description is configured on a PPP interface, the CLI will accept a command to set the alias but it will not appear in the running-config or SNMP interface walk. This will be addressed in the next release of A2.

DNIS match/sub to blank number is ignored

When trying to match a DNIS number and substitute it with a blank number to prevent DNIS digits from being played out an interface, the blank used as the substitute is ignored. An example would be ' match dnis "3000" sub "" '.

Proxy not adding the ";lr" to Record-Route headers

When in stateful or outbound proxy mode, the Adtran does not add the ";lr" to the Record-Route header.

MGCP only: Caller ID name delivery with MGCP units is inconsistent with units configured for SIP

Caller-id name is delivered in quotes for inbound analog calls over MGCP. This may cause an issue with caller-id name detection for some phone systems.

MGCP only: ‘local-domain-name IP’ setting causes “500 Unknown Endpoint” to be returned when a DLCX keepalive is received

When the MGCP local-domain-name is set to an IP address instead of media-gateway, the Adtran will respond to DLCX keepalive with a “500 Unknown Endpoint” error instead of a 250 OK. Using media-gateway as the local domain name works correctly.

403 sent in response to RE-INVITE when should be responding with 481

If a re-invite 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.

GUI: opening System Summary page automatically clears CPU Max Load

Opening the system summary page in the GUI causes the Max CPU stat to clear. This should only clear when clicking the "Clear CPU Max Load" button.

Interval in ‘Voip Name-Service Name-Table’ Not Updated Correctly

The Interval field in the ‘show voip name-service name-table’ doesn’t change, even though the actual entry is timing out correctly. This issue is purely cosmetic.

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.

With NAT configured, receiving video SDP in B2BUA app causes 1 way audio

Calls containing a video SDP will experience one way audio if network address translation is configured on the Adtran. The firewall properly creates a NAT session for audio but the source port reserved is overwritten by the video SDP session.

Replacing/deleting a tracked TCL script causes lockup

The Adtran will lock up when trying to edit or delete a TCL script that is being run on a track.

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 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".

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