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Release Notes
IP Business Gateways

AOS Release A4.05.00
March 11th, 2011

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 A4.05.00 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 Total Access 900/900e and NetVanta 6300 series platforms.

Supported Platforms for A4.05.00

- **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 A4.05.00.

Bad Splice Detection on EFM NIM2s

- Add the ability to detect a bad splice on the EFM module (Netvanta 6310/6330 only). This feature is a merge from the functionality in the NetVanta 800 series.

Tscan support on the EFM NIM2s

- Added Tscan support on the SHDSL EFM NIM2 (Netvanta 6310/6330 only). This feature is a merge from the functionality in the NetVanta 800 series.

Adaptive Training support on the EFM NIM2

- Added Adaptive Training support on the SHDSL EFM NIM2 (Netvanta 6310/6330 only). This feature is a merge from the functionality in the NetVanta 800 series.

Single port ADSL2+ ATM NIM2

- ADSL2+ ATM NIM2 support for Netvanta 6310 and 6330.

International FXS Impedance and Tones - Australia, Ireland, UK, and Mexico

- Added additional impedances and tones to support international applications in Australia, Ireland, UK, and Mexico. This feature request applies to IPBG, Gateways, and NetVanta 7000 series products.

International FXO Support - Australia, Ireland, UK, and Mexico

- Used specifically for international applications where the near end (ADTRAN CPE) must detect the release tone in order to hang up the call. Support for IPBG/UCAS in Australia, Ireland, UK, and Mexico.

Additions for IP Business Gateways in A4.03.00

SIP Media Loopback

- Support SIP medial loopback as specified in media-loopback IETF draft

Caching DNS for Transparent Proxy Phones

- Caching DNS will allow IP phones using an FQDN to re-register and place calls with the ADTRAN running transparent proxy during a network outage.

Auto ERL Tool on FXO Interfaces and NIM2 Modules

- This feature adds the same automatic echo return loss tool that is available today on the FXO ports of the NetVanta 6355 and 7000 series. This will aide in using the NetVanta 6310 and 6330 as analog PSTN gateways with NetVanta UC.

SIP Proxy w/ Aliases

- Addresses survivability issues for applications where DNIS digits received do not match proxy user extensions. This feature also allows support for removal of alphanumeric characters to enable proper handling in failover scenarios.

The following NIM2 modules are now supported in the NV 6310/6330

- 2W/4W SHDSL ATM NIM2

The following VIM modules are now supported in the NV 6355

- E1 ETSI PRI VIM

Additions for IP Business Gateways in 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 A4.05.00.

IP RTP firewall-traversal port-range not reserved on boot-up

Issue Detail

- On bootup, the ports reserved by **ip rtp firewall-traversal** were not skipped by the firewall when doing source NAT.

Forwarded call on a PRI causes a reboot

Issue Detail

- If a Redirecting Number IE was present in a received SETUP message and the Unknown flag was set, a reboot could occur.

Reboot when displaying unsupported codecs

Issue Detail

- When the **debug voice switchboard** command was enabled, printing the name of some unsupported CODECs may have triggered a reboot.

Problems sending larger packets over PPPoE with ADSL NIM2 Module

Issue Detail

- Packets larger than 1410 bytes were dropped by the ADSL2+ NIM2 module when using PPPoE on NetVanta 6310/6330 Series units.

Interop issue with Polycom phones using LLDP-MED

Issue Detail

- There is an interoperability issue that existed between ADTRAN voice products and Polycom phones using OS 3.3.0.1098. The LLDP system capabilities TLV sent by the ADTRAN unit has the TELEPHONE capability bit set. This causes the Polycom phone to reject the LLDP packet. The system capabilities TLV is an optional TLV, and this is not an error within the ADTRAN unit. As an interoperability workaround, the following configuration item was implemented to exclude setting the TELEPHONE capability bit within the LLDP system capabilities TLV: **lldp system-capabilities exclude telephone**

show sip statistics command output typo

Issue Detail

- The output for **show sip statistics** displayed **Response Restransmits transmitted** instead of **Response Retransmits transmitted**.

Reboot due to an SNMP memory leak

Issue Detail

- Performing an SNMP GET could result in a memory leak if the respective interface had a MAC address equal to **00:00:00:00:00:00**. Over time, repeated GETs could consume all memory in the box.

Possible reboot due to race condition between resources

Issue Detail

- A race condition existed between CPU resource utilization monitoring via SNMP and Auto-Link. In rare scenarios, this race condition could cause a reboot.

DNS client sources queries from ports that should be reserved for the DSP

Issue Detail

- DNS query source port randomization was added in AOS A4 as a security measure. The source port randomization feature did not take into account ports that were reserved by other services, such as the DSP. If a source port was chosen that is reserved by the DSP, all DNS queries would fail until the unit was rebooted. This would result in inbound and outbound call failures. For A4.05, the range of ports used for DNS query source port randomization has been restricted.

Issue with ring group coverage

Issue Detail

- Changes were made in A4 that caused an issue with the standard linear ring group configuration. The result was that every other voice user was skipped as the call moved through the ring groups.

ISDN PRI never recovers from busy-out monitor

Issue Detail

- When using busy-out monitor on a PRI to monitor the status of a track, if the track failed and then subsequently recovered, the D channel was never brought back up from the fail state.

Incorrect port inserted into Contact header of proxied INVITE

Issue Detail

- When using the SIP proxy, if the soft switch was using a port other than 5060 for SIP, the Contact header in a 200 OK could get updated with the wrong port when the 200 OK was sent to the phone.

Current and 24-hour ADSL Tx/Rx blocks counters reversed

Issue Detail

- The Tx and Rx blocks count in a **show interface adsl 0/1 performance-statistics Total-24-hour** was reversed from the output of **show interface adsl 0/1** on the ADSL Total Access 900 products.

Netvanta UC interoperability issue: No ringback for transferred calls

Issue Detail

- Ringback was not be heard by the calling party in the following scinero: A call was received on the PRI interface of an IPBG that was being used as a PSTN gateway for Netvanta UC. The call was answered by a SIP phone connected to the UC server. The phone then blind transferred the call back out of the PRI on the IPBG to an external user. The transfer worked properly, but the calling party did not hear ringback. If the call was forwarded instead of blind transferred, then ringback was heard.

MGCP only: Possible reboot when receiving a ModifyConnetion

Issue Detail

- In rare occurrences, receipt of a MDCX could cause a reboot due to an interaction with the timing of a sequence of events.

Calling party number not presented on PRI

Issue Detail

- In some cases, the calling party number received via SIP was not presented in the SETUP message sent on a PRI if an ISDN number template was not present.

ip sip access-class not working after a reboot

Issue Detail

- SIP access classes were not restored to the configuration after a reboot.

FXS impedance changes not properly configured in the GUI

Issue Detail

- Changing FXS impedance values in the GUI did not properly adjust the impedance values on the interface. The impedance values were correctly shown from the CLI.

Event message is generated when using ring groups and accept $ on a grouped trunk

Issue Detail

- When using **accept $** on a grouped trunk, ring group calls would generate an event message to the console.

Cannot specify a MEF Ethernet subinterface for the snmp-server source-interface command

Issue Detail

- Attempting to specify a MEF Ethernet subinterface with the **snmp-server source-interface** command would fail on the NetVanta 6310/6330 Series units.

ISDN trunk account goes into an incorrect state, preventing B channels from being used

Issue Detail

- During high traffic conditions, several processes must compete for available trunk resources. In rare cases, it was possible for multiple trunk appearances to reserve the same B-channel. As a result, the reserved B-channel would not be available until a **shutdown**/**no shutdown** sequence was applied to the PRI interface.

Incorrect ISDN cause code used for busy trunks

Issue Detail

- When the ADTRAN unit received a SETUP message on a particular active B channel, it responded with a CALL_PROC and then a DISCONNECT with a cause code of 21 (CALL_REJECTED). The ADTRAN will now send a more accurate response code of 17 (USER_BUSY).

Reboot after modifying ISDN trunk resource selection

Issue Detail

- In rare cases, changing the resource selection on a PRI trunk would cause a reboot.

Auto-link updates too often if commands are re-entered

Issue Detail

- If the **auto-link** command was entered multiple times, the ADTRAN unit would keep any previous auto link timers active instead of resetting the timer each time the command was entered. This resulted in multiple auto link updates to an N-command MSP server.

Configuring bridging on the IPBGs could result in a core dump

Issue Detail

- Currently, bridging is not supported on AOS voice products. Enabling these commands by mistake would cause the unit to reboot. The bridging command set has been removed.

debug isdn l2-formatted command doesn't properly decode SETUP messages with a calling party name type of 0x82

Issue Detail

- When running **debug isdn l2-formatted**, if the calling party name in a received SETUP message was presented with a type of 0x82, the remaining debug would not display properly.

Removing CoS from a user breaks User Accounts page

Issue Detail

- If a user was removed from a class of service (CoS) using the Classes of Service menu in the GUI, the User Accounts menu would no longer work properly. Removing **cos no-access** from the user account in the CLI would restore access. Changes made to the CoS did work in the User Accounts menu.

Reboot when shutting down PRI interface

Issue Detail

- If a PRI interface was shut down while calls were active, the ADTRAN unit may reboot. The timing window in which this could occur is very small, therefore the likelihood of this reboot is low.

FTP cannot use default AAA list

Issue Detail

- FTP authentication requests for AAA would 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**).

NetVanta 6310: T1 0/1 is only configurable as data in web GUI

Issue Detail

- The NetVanta 6310 GUI would not allow the user to configure T1 0/1 for PRI. However, it could be configured using the CLI.

aaa authentication enable default line enable will not failover correctly

Issue Detail

- **aaa authentication enable default line enable** would not failover to the enable password method if the line password was not configured (in the case of console or Telnet) or if it was not available (in the case of SSH). This means that SSH users cannot pass enable authentication with this configuration.

NetVanta 6310 builds configuration even when provisioning is invalid

Issue Detail

- When no subtended-host provisioning was defined on the Total Access 5000, the NetVanta 6310 would still create a configuration on receipt of the invalid provisioning. The command **subtended-host mode disabled** has been added to allow the unit to discard all subtended-host provisioning received from the Total Access 5000.

Fast busy tone is not played when a 503 Service Unavailable is received

Issue Detail

- The ADTRAN unit would issue a forward disconnect to an analog voice user when a *503 Service Unavailable* was received in response to an INVITE for an outbound call. This prevented the user from hearing a fast busy signal. Now the fast busy will be played for 15 seconds before the forward disconnect is issued.

MGCP only: Confirmation tone (S: g/cf) does not work

Issue Detail

- If an MGCP SignalRequest to play a confirmation tone (S: g/cf) was received, the tone was not played.

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 A4.05.00 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 (A4.05.00) 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 A4.05.00

Upgrade Instructions for NV 6310/6330 using EFM Nims

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.

Before upgrading, make sure the Adtran is running version A3.01.00 boot code. You can verify the boot code by looking at the *Boot ROM version* line in the output of “*show version*”. If the Adtran is not running A3.01.00 boot code, please contact technical support.

Next, download AOS version A4.05.00 application and NIM code to the desired device. The EFM NIM firmware is independent of the application code. It is important to remember to transfer both files to flash before proceeding with the upgrade.

When using tftp, enter the following from the privileged prompt:

```
Router#copy tftp flash
```

During the tftp download you will be prompted for the tftp server name, the 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 sixty-four megabytes of files, whether AOS or configuration files.

Now, apply A4.05.00 NIM code to the EFM NIM. With the NIM connected to the Adtran, issue the following command at the configuration prompt (where x is the slot where the NIM resides):

```
Router#copy flash <filename> int mef-ethernet x/1
```

Depending on which NIM in installed, the filename will either be EFMT1A-A4-05-00.biz or EFMDSLA-A4-05-00.biz for the T1 and SHDSL NIMs respectively. When the NIM upgrade is complete, there will be a message indicating if the upgrade was successful.

Next, configure the system to use the A4.05.00 application code by entering the following command:

```
Router(config)#boot system flash NV6300A-A4-05-00-E.biz
verify
```

The router should then be rebooted with the following command:

```
Router#reload
```

After rebooting, the Adtran will be running AOS version A4.05.00.

To verify that the NIM successfully loaded A4.05.00, enter the following command (where x is the slot where the NIM resides):

```
Router#show int mef-ethernet x/1 version
```

You can verify the application code by looking at the *OS version* line in the output of “show version”.

Appendix A – Errata for A4.05.00

The following is a list of errata that still exist in A4.05.00

Reboot when receiving Diversion Header formatted for E.164
A reboot will occur if the ADTRAN unit receives a SIP message with the Diversion Header formatted for E.164 (i.e.+12565550001).This issue was introduced in A4.01.

NV6310 - "show arp" will not show complete mef-ethernet interface number
The **show arp** command will not show the complete interface number for mef-ethernet interfaces if the subinterface is more than one digit.

Stateful proxy doesn't replace hostname
When in stateful mode, the SIP proxy will not replace a hostname in the Request-URI, From, and To headers with the configured sip-server when the received hostname resolves to a local IP address.

Invalid static routes are incorrectly added to the route table as "null"
Adding a static route in an ADTRAN unit that points to a gateway address that falls within the subnet included in the same route will create a null route in the IP routing table for the subnet referenced in the static route.

RFC 2833 events reference an incorrect RTP channel
debug voice dsp voip 0/1 channel verbose rfc2833 displays incorrect DSP Channel in debug messages that are printed to the screen.

Analog initiated transfer fails
In A4.03 and later, analog initiated transfers on the ADTRAN unit use the Contact header information of the second call in the Referred-To header of the REFER. This can cause transfers to fail in some networks and the transferred call to appear as a second call on the target phone. Prior to A4.03 the ADTRAN unit used the "To:" header of the second call in the "Referred-To:" header.

LocalAddr and RemoteAddr in VQM PUBLISH packets are reversed
In the PUBLISH messages generated by VQM reporter, LocalAddr/RemoteAddr and LocalURI/RemoteURI are reversed.

Intermittent error configuring IP unnumbered
In some cases, it is possible that the user will see the following error message on the CLI when assigning an unnumbered address to an HDLC or PPP interface: **%Point-to-point (non-multi-access) interfaces only**. A workaround would be to assign the unnumbered address to a loopback interface and then assign it to the desired Ethernet interface.

Proxy failover mechanism not working properly
In a failover scenario, the SIP proxy may retry the INVITE to the SIP server a second time before routing the call via the switchboard, resulting in additional delay before the call is connected.

B-channel restarts sent when using 5ESS even when they are disabled
When using the 5ESS ISDN switch type, B channel restarts may be sent in some scenarios, even if they are disabled.

Feature group D prefix does not work
The prefix statement on the **dnis-digits** command on a feature group D trunk does not properly prepend digits to the DNIS number. A workaround would be to use the match/sub feature on the feature group D trunk.

ISDN Caller ID name intermittently works when set to deliver after Proceeding
In some cases, ISDN Caller ID name will not be delivered when configured for delivery in a Call Proceeding instead of a setup message.

Reboot may occur when removing voice trunk
In rare cases, the ADTRAN unit may reboot if a voice trunk is removed from the configuration.

T.38 failure when receiving NSF
In some cases, receiving T.38 NSF packets may cause fax failures.

MGCP stack reboot
In rare cases, it is possible for the ADTRAN unit to reboot if the MGCP stack is disabled and then re-enabled.

503 error when adding an IP host entry through the GUI
Attempting to add an **ip host** entry through the GUI results in a 503 error and the entry is not added.

HTTP and SSH TACACS+ authentication packets do not properly populate the Remote-Address field
SSH and HTTP/HTTPS connections do not populate the Remote-Address field of the TACACS+ authentication request. This results in the field always being populated with **0.0.0.0**. Telnet connections are not affected.

The proxy ID is not removed from contact binding when the grammar proxy-id is set to contact-param
When using the **ip sip proxy grammar proxy-id contact-param** command, the proxy ID will not be removed from the 200 OK sent to the proxied phone.

Possible lockup when modifying outbound proxy
In some cases, a lockup may occur when modifying the outbound proxy address for a SIP trunk in the GUI.

After reboot, FXS ports that were administratively shut down do not show as disabled
If an FXS port is administratively shutdown and the unit is rebooted, the two-wire status may show as onhook instead of disabled after the unit comes back up.

Named digit timeouts in dial plan entries do not work on CAS trunks
Named digit timeouts in dial plan entries are ignored when receiving digits from a CAS trunk.

RTSP ALG blocking session
The RTSP ALG may interfere with Windows Media Player streaming sites. The workaround is to use the **no-alg** option on the NAT statement for any ports that utilize RTSP, typically port 554.

Reload scheduled in message does not appear in all scenarios
Using some authentication options, the **Reload scheduled in** message will not appear at login if a reload is scheduled.

RFC 2833 end packets are sent without a start packet
In some cases, RFC 2833 end packets may be sent without a non-end packet being sent first.

SIP to PRI cause code mapping failure
Configuring a non-default PRI cause code mapping for a *403 Forbidden* received from the SIP network does not work properly.

Local SIP and MGCP ports are not protected from use by NAT
When using NAT, the local SIP and MGCP ports are not protected from use by the firewall. Because of this, It is possible for a NAT session to hijack SIP or MGCP traffic destined for the unit. A workaround is to configure **no ip firewall nat-preserve-source-port**.

NAT source ports conflict with configured ip rtp udp range
With NAT source port preservation disabled, NAT sessions may fail once the source port being used reaches the reserved range configured with the **ip rtp udp** command.

Apparent glare condition results in a 400 Bad Request instead of attempting to backoff/retry or a 491
SIP glare when re-inviting to T.38 from both sides can result in a *400 Bad Request* being sent instead of a more correct SIP response to the received re-INVITE.

HDLC 1 interface menu of the GUI does not support an address type of unnumbered
Unnumbered IP settings are not reflected in the GUI and defaults to None. If the Apply button is selected, it will remove the IP address from an HDLC interface.

FXO impedance values don't apply properly on the GUI
FXO impedance values are not applied properly when changed in the GUI.

Unable to modify Ethernet subinterface via GUI
Modifying an Ethernet subinterface in the GUI will cause a 503 Server Error.

GUI fails to load
With NAT enabled on a non-default VRF, the GUI on the ADTRAN unit may become unresponsive. The GUI Is not supported on non-default VRFs.

GUI QoS wizard won't complete when adding map to a MEF Ethernet interface
QoS wizard can not apply a QoS map to a MEF Ethernet interface.

Hook flash not handled properly
If an FXS user places a call that is hair pinned out a PRI trunk on the same device, the call will disconnect if the FXS user attempts a flashook.

sess-id and sess-version changed in SDP origin field on receipt of a re-INVITE
Upon receipt of a re-INVITE, in some scenarios the sess-id and sess-version in the origin field of the SDP answer may change.

Calling party number not presented to FXS
If an INVITE is received from the soft switch in the E.164 format, the Caller-ID number will not be delivered to an analog FXS station.

Reboot when deleting a voice trunk with active calls
If a voice trunk is deleted with calls still active on that trunk, the unit will reboot.

MGCP only: DSP channels are not chosen or cleared correctly
In rare cases, DSP channels may not be cleared properly when an MGCP call is torn down between an ADTRAN IPBG and a Nortel CS2K.

Packets are lost in VQM / voice quality-stats when RTP stream changes
If the remote voice gateway changes the SSRC in an RTP stream received by the ADTRAN unit, and the sequence numbers are noncontiguous, VQM and the output of **show voice quality-stats** will log lost packets for the number of packets between the last sequence number of the first stream and the first sequence number of the new stream. This issue is purely cosmetic.

40 ms packetization period is not rejected in 183 or 200
If an unsupported packetization period is presented to the ADTRAN unit in an SDP answer, no indication that the presented ptime isn't supported by the ADTRAN unit will be sent to the remote user agent. This will result in no talk path.

NetFlow not reporting inbound RTP streams
Under certain conditions, inbound RTP streams for voice calls terminated by the ADTRAN unit cannot be exported to an external NetFlow collector.

SIP proxy not applying SIP grammar commands to Contact header
The Contact header for *302 Moved Temporarily* messages sent through the proxy will contain an IP address instead of the domain configured with **ip sip grammar** commands.

PRI does not acknowledge Connect or Disconnect
In rare cases, the PRI interface will get into a state where it will not acknowledge a CONNECT or DISCONNECT from the PBX. Performing a **shutdown** / **no shutdown** on the PRI resolves the issue.

T1 Degraded Minutes counter is inaccurate
In some cases, the T1 interface statistics will log Degraded Minutes although there are no other physical errors logged for that T1.

PRI states tied together for multiple PRIs in a single ISDN group
With multiple PRIs in the same ISDN group, bringing one PRI down will cause calls that should use the other PRI to fail. A workaround is to use two isdn-groups that only contain one PRI each.

T.38 packets sent after T.38 INVITE even when the ADTRAN unit is not configured for T.38
If a reINVITE is received from a gateway attempting to set up T38, and the user/trunk associated with the call is not configured for T38, it is possible for the ADTRAN unit to send T38 packets even though the request is properly rejected with a *488 Not Acceptable Here* message.

RFC 2833 failure when padding is used
The ADTRAN unit will not properly process RFC 2833 DTMF packets if padding is used to increase the size of the RTP packet.

T.38 failure with multiple page fax against Sonus GSX gateways
Multiple page faxes may fail when using T38 against Sonus GSX gateways.

Problem with processor utilization during VPN tunnel renegotiation
VPN tunnel renegotiation can cause the packet routing processor queue to spike to 100%. This spike in processor utilization could cause momentary voice quality issues.

200 OK sent with T.38 RTP port set to zero
The ADTRAN unit may respond to a T38 request from a Nortel CS2K with the T38 RTP port set to zero. The ADTRAN unit will immediately send a BYE to disconnect the call.

RFC 2833 events are not detected/processed
DTMF digits received by the ADTRAN unit using RTP payload type 127 may not be detected properly.

Only one emergency call can be processed at a time
If an emergency call is pending, all other emergency calls will be held until the initial call is processed or torn down.

T.38 fax failure with Interactive Intelligence IP PBX
T38 PRI to SIP fax calls initiated by an Interactive Intelligence PBX over a PRI trunk may fail.

show interface t1 0/1 performance total command does not actually show the total for the performance
In some cases, the s**how interface t1 0/1 performance total** command does not actually show the total for the performance intervals. The total value is viewable in the GUI.

ifCounterDiscontinuityTime should be updated when clear counters command is issued
When performing a **clear counters** on an ADTRAN unit, the ifCounterDiscontinuityTime is not updated. This could cause incorrect data to be recorded when monitoring via SNMP.

T.38 fax failure against Audiocodes gateway
T38 calls against Lucent LCS or Audiocodes gateways may fail to negotiate properly.

No banner displayed before user and password prompt when connecting via SSH v1
If an AAA authentication banner is configured, users logging in using SSH v1 will not see the banner when prompted for a login. This same configuration for Telnet or SSH v2 users works correctly.

Voice quality statistics jitter buffer average is greater than the maximum

In rare instances, the output of the **show voice quality-stats** command could show that the average size of the jitter buffer is higher than the maximum value.

ADSL interface configuration constantly changes

The interop-flag and phy-flag change constantly on the ADSL interface, resulting in the running configuration always being different than the startup configuration. This could cause issues if the ADTRAN unit is configured to report back to N-command MSP.

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.

DTMF tone degradation with G.729 codec

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%.

WEB GUI page incorrectly shows Time Server that is installed

With an NTP Server configured, the Summary Web Page will only show the status of the SNTP Server as not configured.

Using a range for a dial-plan entry does not work

Using a range for a dial-plan entry does not work properly

Packets dropped from SHDSL ATM NIM2

NV 6310: Unit will drop about 1 out of every 15K packets from the SHDSL to Ethernet direction.

Higher speed line rates do not work on SHDSL NIM2

Throughput on SHDSL interfaces in the NV 6310 / 6330 are lower than expected in speeds above 5696 Kbps.

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 if both sides of the SIP dialog 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.

3-way call feature not working properly in all cases

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

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

After a PPP link is lost and then recovers, it is possible that an endpoint will not hear dial tone after going off hook or that the user may prematurely hear a busy signal. If calls were attempted

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

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

PRI goes out of service when attempting ISDN to ground start trunk calls on FXO 0/1

PRI to ground start trunk calls do not work on the TA900e when the PRI is on T1 0/3 and the ground start trunk is on FXO 0/1. The PRI will go out of service when this type of call is attempted on these ports. This is not an issue on the TA900. This also works on the TA900e if the PRI is on T1 0/4 or if the ground start trunk is on any other port except FXO 0/1.

Calls between FXS and FXO port on the Quad FXO/FXS card fail

Calls between FXS and FXO port on the Quad FXS/FXO modules will fail.