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 A4.03.00
October 15th, 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 A4.03.00 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 A4.03.00

- **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 A4.03.00.

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

A4.03.00 includes bug fixes up to and including A2.07.00. This section highlights major bug fixes in AOS version A4.03.00.

auto-link not backing up files

Issue Detail

- Products without compact flash would not report any files to be backed up, and thus would not perform backups to n-Command MSP. This issue has been addressed.

Proxy Core Dump

Issue Detail

- A reboot would occur if a SIP REFER message were sent that included an orbit parameter.

ISDN number templates don't apply to called party number

Issue Detail

- If an INVITE were received that did not have a phone-context specified in the Request-URI, ISDN number templates would not be processed properly for the called party. This issue has been addressed.

SIP to ETSI PRI calls not matching ISDN number templates

Issue Detail

- ISDN number templates would not match called party numbers with a leading one or zero. This issue has been addressed.

Reboot when turning off "debug sip stack messages"

Issue Detail

- Long headers in a SIP message could result in a reboot when turning off "debug sip stack messages". This issue has been addressed.

Reboot when removing VRF

Issue Detail

- If a configured VRF were removed, the ADTRAN would reboot. This issue has been addressed.

Calling Party Name not preserved when call is transferred from one SIP trunk to another

Issue Detail

- Calling Party Name was not preserved when a call was transferred from one SIP trunk to another. This issue has been addressed.

503 error on Trunk Groups page when selecting a trunk group

Issue Detail

- In some cases, clicking on a trunk group on the Trunk Groups page in the GUI could cause a 503 error. This issue has been addressed.

NV 6310: T1 is the only option when connecting a PRI to an E1

Issue Detail

- When the ADTRAN was configured for an E1 country, "connect t1" was still shown when configuring a PRI interface. This issue has been addressed.

NV 6310: PRI interface does not allow "connect" statement without switch-type defined

Issue Detail

- A PRI interface will not allow a "connect" statement without a switch-type defined. The NV 6310 did not have a switch type defined by default. A change has been made so that the switch type of NI2 (when configured for T1) or ETSI (when configured for E1) will be used by default when attempting to connect the PRI interface to a TDM group. This issue has been addressed.

Lost packets being reported incorrectly for outbound calls

Issue Detail

- During periods of silence, the counters used to calculate the "show media-gateway channel" statistics would continue to look for packets that were not present, even though packets were not expected. These packets were reported as lost. This issue has been addressed.

Lost packets on show voice quality-stats doesn't match up with show media-gateway channel stats

Issue Detail

- The output of "show voice quality-stats", "show media-gateway session 0/x.y", and "show media-gateway channel 0/x.y" did not show the same values for various statistics. The statistics have been updated to count packets in the same way across all 3 commands. This issue has been addressed.

AAA: SSH does not permit multiple login attempts against local auth DB

Issue Detail

- With AAA configured, a user attempting to log in over SSH would not be allowed multiple attempts if the unit was configured to authenticate against the local database. SSH will now prompt multiple times, up to the configured limit, when incorrect credentials are entered. This issue has been addressed.

IAD sends unknown as Call-ID in PUBLISH message when using VQM reporter

Issue Detail

- Previously the Call-ID in the PUBLISH sent by the VQM reporter was listed as "unknown". The ADTRAN will now do the following, depending on the call type. For a SIP to TDM call (or vice-versa) we now display the SIP Call-ID in the VQM report. For a SIP to SIP call, we will now show both Call-ID's in the VQM report separated by a '/'. This issue has been addressed.

Negotiation of MRRU to a higher value breaks OSPF

Issue Detail

- Changes were made in A4.01 (merged from 17.05.02) that would allow an MLPPP interface to up-negotiate the MRRU value advertised by the remote peer. These changes introduced a new issue with OSPF. Since the IP MTU is not configurable in AOS and the L2 PPP MTU requested by the remote peer can now be large than 1520, there could be a mismatch between the MTU values advertised in OSPF. The MTU the ADTRAN uses in the OSPF database description packet was based on the IP MTU in the Adtran (which is not necessarily the same as the interface MTU). The value advertised by the peer is most likely tied to its L2 PPP interface. This issue has been addressed.

MGCP: RTP sent even after mode changes to recvonly or inactive

Issue Detail

- In MGCP, if the ADTRAN received a ConnectionMode parameter that indicated a recvonly or inactive state, the unit would improperly keep sending RTP. This issue has been addressed.

GUI: media-gateway set to loopback doesn't display specific interface

Issue Detail

- The web GUI would not display the specific loopback interface used when applied to a physical interface as the media-gateway address. This issue has been addressed.

MGCP: TA 900 rejects SDP when originator field includes a "-"

Issue Detail

- If the date and time on the ADTRAN was on or after July 25 2010, the SDP session ID for MGCP calls would begin with a negative integer. This could be problematic when calling from ADTRAN unit to ADTRAN as the receiving ADTRAN would not process SDP with negative integers in the session ID field. This would result in one way audio when calling from ADTRAN to ADTRAN. This issue has been addressed.

Timing related DSP reboot

Issue Detail

- Changes were made to DSP code that should address certain intermittent DSP reboots. This issue has been addressed.

Reboot during Security Test - SIP Protocol

Issue Detail

- This reboot occurred while parsing a malformed SIP packet during a security test. The SIP parser determined the packet was invalid and logged an error message that specified the full contents of the malformed packet. The reboot occurred during the construction of this error message. This issue has been addressed.

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

Issue Detail

- Calls would fail between a private extension on one phone and an SLA/BLA line on a second phone when both phones were behind transparent proxy in survivability mode. When the private extension dialed the SLA extension, the SLA phone could answer the call but it would be disconnected within a few seconds. This issue has been addressed.

NTP master command does not restore correctly

Issue Detail

- The CLI command 'ntp master' was not restored after a reboot. This issue has been addressed.

2B-Channel transfer failure

Issue Detail

- During a 2B channel transfer, the ADTRAN did not have the ability to handle receiving a 180 w/ SDP. This will cause a 2B channel transfer to fail.

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

Issue Detail

- Unresolved DNS queries generated by the IAD for VoIP Name Resolution did 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 were resolved.

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 A4.03.00 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 (A4.03.00) 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 A4.03.00

Upgrade Instructions for NV 6310/6330

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 A4.03.00 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 A4.03.00 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-A4-03-00.biz or EFMDSLA-A4-03-00.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 A4.03.00 application code by entering the following command:

```
Router(config)#boot system flash NV6300A-A4-03-00-E.biz
verify
```

The router should then be rebooted with the following command:

```
Router#reload
```

After rebooting, the Adtran will be running AOS version A4.03.00.

To verify that the NIM successfully loaded A4.03.00, 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 A4.03.00

The following is a list of errata that still exist in A4.03.00

Call processing related reboot
The most likely scenario in which this reboot could occur is if a call were modified at the same time as it was being torn down. The timing window in which this could occur is very small, so the likelihood of seeing this reboot is low.

FXS codecs won't initialize after a reboot
If the ADTRAN receives an inbound call immediately after bootup, it is possible that the analog codec associated with the FXS port receiving the call will not initialize properly. The result is that the 4 FXS ports associated with the analog codec would be in the "down" state until the ADTRAN is rebooted.

‘debug mgcp stack messages’ truncated from CLI
When using "debug mgcp stack messages", MGCP messages longer than 1,024 characters are truncated from the CLI output. This issue is purely cosmetic.

Trunk manager reboot
If a PRI goes down while a call is in a pre-connect state (i.e. ALERTING), it is possible for a reboot to occur.

Possible reboot during call setup and teardown
If a call is initiated and then terminated almost simultaneously, the ADTRAN may experience a reboot. The timing window in which this could occur is very small, so the likelihood of seeing this reboot is low.

‘no-alg’ config parameter not showing up in config
The "no-alg" parameter was not applied to an Access Control Policy entry even though the command was accepted.

Voice quality stats show invalid characters
When a call duration is extremely short (less than a couple of seconds), the delay in "show voice quality-stats" could show "-(".

With IRB configured, eth 0/1 is always associated with bridge group
When configuring IRB on the IPBG, eth 0/1 was always included as a part of the bridge group. Bridging and IRB are not currently supported on IPBGs.

Reboot while manually generating an exception report
Manually generating an exception report on a unit with active calls could result in a reboot.

T.38 sent after T.38 invite even when the ADTRAN is not configured for T.38
If a re-INVITE is received from a gateway attempting to setup T38 and the user/trunk associated with the call is not configured for T38, it is possible for the ADTRAN to send T38 packets even though the request is properly rejected with a "488 Not Acceptable Here".

Reboot after modifying ISDN trunk resource selection
In rare cases, changing the resource selection on a PRI trunk could cause a reboot.

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

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

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

Nortel 3-way conferencing with conferencing-uri fails
Three-way conferencing against Nortel CS2K with 'conferencing-uri' fails. Using local conferencing allows three-way calls to work properly.

Problems 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.

Request-URI in INVITE generated in response to REFER does not match Refer-To header
If the Refer-To header in a received REFER contains a host that doesn't match the configured SIP server or outbound proxy, the INVITE generated will not be constructed properly, resulting in a failure.

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 sip grammar commands.

re-INVITE w/ Replaces not processed properly when sent immediately following an INVITE
Toshiba SIP PBX interoperability issue: An INVITE w/ Replaces is not processed properly when the original call is in the ReinvitePending state. This results in incomplete calls with the Toshiba.

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 w/ Interactive Intelligence IP PBX
Interactive Intelligence IP PBX interoperability issue: T38 SIP to PRI fax calls initiated by an interactive intelligence PBX over a PRI trunk may fail.

Reboot when Shutting Down PRI Interface
It is possible that the ADTRAN unit will reboot if a PRI interface is shut down while calls are active. The timing window in which this could occur is very small, so the likelihood of seeing this reboot is low.

FTP can not use default AAA list
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”).

‘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.

‘show interface t1 0/1 performance total’ does not actually show the total for the performance intervals
In some cases, the "show interface t1 0/1 performance total" does not actually show the total for the performance intervals. The total value is viewable in the web GUI.

ifCounterDiscontinuityTime should be 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.

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

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.

Reboot with Loopback Plug on Ethernet port
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.

FXO ports 1-8 on TA 900/900eL2 missing impendence values
The FXO ports on the FXO daughter card for the TA 900/900eL2 do not have the options for 600r or 900r impedance values.

Voice Quality-Stats Jitter Buffer Average greater than Max
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.

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

Confirmation tone (g/cf) does not work
When the ADTRAN receives a S:g/cf to play a confirmation tone, no tones are played out the FXS interface.

No banner displayed before user and password prompt when connecting via SSH v1

If a 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 and SSH v2 users works correctly.

T.38 fax failure against Audiocodes and Lucent LCS gateways

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

MGCP codec negotiation / fax issues

MGCP only: With "voice codec-priority offer-sdp" configured, the ADTRAN will ignore the codec list in the local connection options and in the SDP offered by the gateway and will instead prefer the codec list configured on the voice user.

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

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.

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 "shut / no shut" 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.

Incorrect ISDN cause code used for busy trunks

When the ADTRAN receives a SETUP message on a particular B channel that is currently active, it responds with a CALL_PROC and then a DISCONNECT with a cause code of 21 (Call_Rejected). A more accurate response would be to use a cause code of 17 (User_Busy).

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.

RFC 2833 failure when padding is used

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

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

If the "auto-link" command is entered multiple times, the ADTRAN will keep any previous auto-link timers active instead of resetting the timer each time the command is entered. This will result in multiple auto-link updates to an N-command MSP server.

SETUP message with restricted number causes debug display error

If the calling party number is restricted and the calling party name is restricted with a value of "anonymous", the 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.

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

ISDN l2-formatted debug doesn't properly decode SETUP from an Audiocodes gateway
In some cases, "debug isdn l2-formatted" will not properly decode ISDN SETUP messages from Audiocodes gateways.

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.

NV 6310 builds config even when provisioning is invalid
When no subtended-host provisioning is defined on the TA 5000, the NV 6310 still builds some config on receipt of the invalid provisioning.

Fast busy tone is not played when a '503 Service Unavailable' is received
The ADTRAN will issue a forward disconnect to an analog voice user if a "503 Service Unavailable" is received in response to an INVITE for an outbound call. This prevents the user from hearing a fast busy signal.

NV 6310/6330: Issue with calls on Quad FXS/FXO module
Calls between FXS and FXO port on the Quad FXS/FXO modules will fail.

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.

NV 6310/6330: Reboot happened when specifying eth 0/1 for dhcp
This issue seems to be related to the traffic when the box boots up. This issue has never been reproduced after the unit is up and running for more than a few seconds. It has only been reproduced by having the ADTRAN reboot and then during the bootup procedure or immediately after booting up, the ADTRAN may core dump and reboot again. It is not very likely to be seen in the field.

NV 6310/6330: Higher speed SHDSL line rates do not work
Throughput on SHDSL interfaces in the NV 63XX are lower than expected in speeds above 3904 Kbps.

NV 6310/6330: Reboot while doing GUI config
Single occurrence reboot when trying to apply config changes to the SHDSL physical interfaces page. It is not very likely to be seen in the field.

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.

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.

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 TA 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 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 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.