ADTRAN The Network Access Company

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

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

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

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

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

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

Release Notes
AOS IAD Products

AOS Release A1.02.00
April 14, 2008

Contents

Purpose and Supported Platforms

AOS Voice Products release A1 is a major system release that adds many new features and addresses customer issues that have been uncovered in previous code releases.

Release A1.02.00 is Generally Available code, meaning that it has been subjected to both Design Verification and Product Qualification testing. Results obtained during this testing have been evaluated and the code has been determined to be ready for General Availability. Caveats discovered during testing but not addressed in this build are listed in Appendix A.

A listing of available documents for this release appears in Appendix B. Configuration guides, white papers, data sheets, and other documentation may be found on Adtran's knowledge base, http://kb.adtran.com.

The contents of these release notes will focus on the Netvanta 6355/1355 and the Total Access 900/900e series platforms. Netvanta 7100 release notes are available on the Adtran knowledge base:

http://kb.adtran.com/article.asp?article=2346&p=2

Supported Platforms for A1.02.00

- **NetVanta 7100** – Converged IP Office in a Box
- **NetVanta 6355/1355** – Multiservice Access Gateway
- **TA 900 Series** – VoIP Multiservice Access Gateway, single T1 interface
- **TA900e Series** – VoIP Multiservice Access Gateway, multi-T1 interface

Summary of New Features

This section highlights the major features, commands, and behavioral changes for AOS A1.02.00. For a list of related documents, please see Appendix B.

Additions for TA 900/900e series in A1.02

MGCP – Media Gateway Control Protocol

MGCP is a newly added VoIP call control method available for 2nd generation TA 900s. FXS ports are the only available endpoints for MGCP. No T1 CAS or T1 PRI support will be provided in this release. Media will still be transported over RTP, as it currently is with our SIP implementation. SIP and MGCP can coexist on the same IAD at the same time (i.e. SIP used for PRI delivery while MGCP used for FXS delivery).

3-Way Conferencing

Local 3-way conferencing is now supported in the 2nd gen TA 900 series. 3-way conferencing is only supported in 2nd generation TA 900s. The only local conferencing participants currently supported are FXS voice users. The IAD is limited to only 3 separate 3-way conferences at a time.

Additions for All Voice Products in A1:

VQM – Voice Quality Monitoring

Ability to make VoIP quality measurements on RTP media flows terminated or passed through the IAD. Results will estimate MOS scores for particular calls as well as jitter buffer performance.

Loopback Accounts

When the loopback account call is connected, the RTP audio is looped back. This will provide an easy method to verify proper operation and configuration during install, and can be used with VQM to troubleshoot network issues. This feature includes the ability to initiate SIP calls via the CLI.

AWCP - ADTRAN Wireless Control Protocol

AWCP is a Layer 2 Control and Management protocol that enables the platform to act as a Wireless Access Controller for the Netvanta 150 Access Point. Up to eight Netvanta 150 Access Points can be configured and managed by the AC.

Top Talkers

The NetFlow flows that are generating the heaviest system traffic are known as the "top talkers." The NetFlow MIB and Top Talkers feature can be used for security monitoring or accounting purposes for top talkers, and matching and identifying key users of the network.

Full PRI support

All 23 B channels of a PRI can be used simultaneously. The number DSP resources are increasing from 16 to 24 channels on the RoHS-E1 units.

Top visited websites

The top websites feature is designed to report top websites requested by users to system administrators. This feature is intended to be used in conjunction with the *ip urlfilter* command so that customers without a Websense server will have a simple URL filtering package.

Additional Features

- VRF aware DHCP server
- VRF aware firewall
- Portal-List(s) to assign username access to different portals (system applications such as http, telnet, ssh, ftp, and console). Previously, a username for ftp could also telnet or ssh into the unit.

Summary of Bug Fixes

This section highlights major bug fixes in AOS version A1.02.00.

Multiple T.38 calls will fail instead of failing over to G.711

Issue Detail

- If more then one T.38 call is placed through the IAD at one time, the second and subsequent fax calls will fail without failing over to G.711. Note: Only one T.38 call is supported at a time.

Corrective Action

- Made change to active queue counter to correctly maintain the number of T.38 calls. The correct flag is set to failover to g711 if more then one T.38 call is active.

Missed digits when dialing from certain analog phones

Issue Detail

- Digit detection in the IAD can be problematic with older analog phone stations.

Corrective Action

- Changes to the DTMF detection libraries in the DSP will increase the reliability of digit detection in A1.02.

SRV records do to not update change of priorities

Issue Detail

- If the configured DNS server changes the SRV record priorities, the IAD will not properly update the new priorities in the DNS table.

Corrective Action

- Fixed bug in the DNS table where priority was used to compare record equality. This would result in duplicate entries being inserted into the table when they should have been replaced.

Password encryption does not work with sip-identity on a ring group

Issue Detail

- If password authentication is configured for the sip-identity of a ring group, the password will say it is encrypted in the running config, but the password itself will still show as un-hashed

Corrective Action

- Changes made to check for encryption on ring group configuration.

DSP reboot during T.38 fax call

Issue Detail

- During the setup of a T.38 fax, it is possible that the IAD will reboot. This is due to a memory corruption error in the DSP.

Corrective Action

- Changes made to DSP resource allocation to resolve the reboot.

A-record addresses will purge from DNS table if TTL is lower then the associated SRV record

Issue Detail

- If an SRV record has a TTL longer than that of the A records that it points to, the A record address will expire in the host table and not be re-queried until we receive a SIP message from the SIP server. Due to the amount of time it takes to re-query and resolve an SRV record, as well as respond to the SIP message, the first inbound call after the A-record times out will fail. All subsequent calls will work until the A-record times out again.

Corrective Action

- Modified DNS to support internal derived record registration.

Reboot when receiving 2 RTP streams with the same SSRC value w/ VQM enabled

Issue Detail

- With VQM enabled, the IAD will reboot if we receive 2 RTP streams with the same synchronization source identifier.

Corrective Action

- If two streams are using the same SSRC, the IAD creates two VQM mon handles, storing them with the SSRC. The second map insert will fail, causing the reboot. Made change to handle multiple SSRCs

"ip rtp symmetric-filter" not maintained after reboot

Issue Detail

- "Ip rtp symmetric filter" will not load into the running-config after a reboot, even if the config is saved in the startup-config.

Corrective Action

- Made appropriate changes to maintain after a reboot.

Reboot when receiving an SDP with no media field

Issue Detail

- If the IAD received an SDP with no "m" field, the IAD would reboot.

Corrective Action

- The IAD was trying to parse the pTime attribute from the media field, even if the media field wasn't present. This caused the IAD to reboot. It now checks for "m" before trying to parse the pTime.

Reboot when trying to run a T.38 and G.711 w/ PLC call simultaneously

Issue Detail

- If a G.711 w/PLC call is placed first, followed by a T.38 fax call, the DSP will lockup causing a reboot.

Corrective Action

- G.711 PLC calls were not being tracked by the controller (T.38 cannot run when G.711 PLC is active), which led to the deactivation of a channel the DSP never started, locking the DSP. The controller will now prevent a T.38 channel start attempt when G.711 PLC is active.

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 A1.02.00 from the ADTRAN support 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 (A1.02.00) 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 A1.02.00.

Appendix A – Errata for A1.02.00

The following is a list of errata that still exists in A1.02.00.

Cannot manage IADs running A1 in N-Command

The IADs show up as “Unmanaged SNMP Devices” in N-Command when running any version of A1. This is a problem with how N-command parses the system description field in the IAD. Upgrading IADs to A1 does work without any loss of IAD configuration, but the management functionality is lost. This is a problem with N-command code and is not related IAD code. This will be addressed in the next version of N-command firmware.

“ip unnumbered” removed from interface config after upgrade to A1

Without first configuring an interface with an IP address, the unnumbered command won’t be parsed correctly from the config. This is a problem upon bootup or when configuring a fresh interface. If the IAD is upgraded from a previous release to A1, it will lose the unnumbered config option on the PPP interface, preventing PPP from properly negotiating. This will be addressed in A1.03.

“ip sip/rtp qos dscp” doesn’t verify value

The DSCP value assigned for qos should only be in the range of 0-63. If any value is entered outside that range, there won’t be an error message letting the user know that they entered an incorrect value. This will be addressed in A1.03

NV1355/6355 only - T-1 NIM loses LBO setting after a reboot

If the LBO setting on the T-1 NIM is changed from the default of 0dB, the change will not be maintained after a reboot. This will be addressed in A1.03

Caller-id name override doesn’t work for inbound calls on a trunk

Caller-id number override for voice trunks works properly; however, the CID name field cannot be manipulated. This previously worked in version 12. This will be addressed in A2.01.

INVITE not properly changed after 302 ‘moved temporarily’

If a 302 is used to change the destination port for subsequent calls to a specific user, the URI generated by the IAD contains the requested port number, but the destination port for the SIP packet still contains the original port number. This will be addressed in A2.01.

URI matching for SIP PUD needs to be tweaked to handle certain SIP endpoints

If a SIP endpoint sends a REGISTER through the proxy containing an rinstance but does not send the ristnance in the INVITE, the PUD lookup in the proxy will think that the two contact headers don’t match. This violates RFC 3261. This will be addressed in A2.01.

Improper detection of V.8 ANSAM without phase reversal

In the rare case that a fax/modem device sends a V.8 ANSAM without phase reversals, the IAD will report V.8 ANSAM with phase reversal as soon as the ANSAM is detected. However, when the end of the tone is detected, the IAD then could report V.8 ANSAM (no PR), causing the controller-side behavior to send another INVITE. This will be addressed in A2.01.

Trunk appearance leak on E&M wink trunk w/ dialtone enabled
If dialtone generation is enabled on an E&M wink trunk without saving the change to the startup-config and then rebooting the IAD, a trunk appearance could lockup and by unavailable. Rebooting the IAD will permanently clear up the issue. This will be addressed in A2.01.

aaa authorization command changes terminal length to 0
"aaa authorization exec default group tacacs+ if-authenticated" changes the terminal length to 0. Entering "terminal length <#>" will clear up the problem. This will be addressed in a future release of code.

DNS query initiated by SIP stack causes "ip host" entry to be removed from config
If an IP host entry is configured, and then the SIP stack initiates a DNS query for that FQDN, the ip host entry will be removed from the running config. This will be addressed in a future release of code.

Inbound calls to DID 9111 match emergency-call dial plan
Calls received on a trunk (SIP, PRI, E&M, FXO) with a dialed number of 9111 will match the emergency call dial plan entry. The call will still process correctly; it will just be given an improper priority in the switchboard. This will be addressed in a future release of code.

Problem with GUI system summary report
If 802.1q subinterfaces are configured on an Ethernet port, the GUI system summary will report a problem with the Public security zone not being set correctly. The IAD functions fine, but the GUI has a problem detecting the VLAN sub-interface when checking for the configuration. This will be addressed in a future release of code.

Empty username sent in Authorization header of BYE message
When a BYE originating from the IAD is sent with authentication, the username field is empty in the Authorization header of the BYE message. This will be addressed in a future release of code.

Problems handling 0.0.0.0 SDP in RFC 2543 hold through the SIP proxy
If the proxy receives an INVITE to put a call on hold using RFC 2543, the associated 200OK will cause the IAD to create a firewall association for the non-existant RTP stream using an IP of 0.0.0.0. When this fails, the proxy never forwards the response on to the phone. This will be addressed in a future release of code.

Audio cuts out during call waiting beep on FXO to FXS call
When call waiting is disabled on a voice user and an inbound call hits the FXO trunk (destined for the voice user), the audio cuts out during the period where the call waiting beep would normally be played. This will be addressed in a future release of code.

NV1355/6355 only - Erasing startup-config could result in IAD loading incorrect default configuration
Default config may be restored to the wrong default config after erasing startup-config or issuing a factory-default command. This is due to a change in the way the default config is stored for A1. This will be addressed in a future release of code.

SIP to MGCP Ringback issue

While placing a call from a SIP user to an MGCP endpoint on the same TA 900 with both lines registered, the SIP user will not hear ringback. This is a result of the separation of SIP and MGCP internally in the IAD. 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.

DNS proxy failover failure

If a DNS request is sent through the DNS proxy of the IAD to the primary public DNS server and the DNS server responds with a "destination unreachable", the secondary DNS server won't be queried.

SSH sessions are always listed as authentication in progress

The "show users" command does not list the idle time of any SSH sessions. They are always listed as "authentication in progress" whether or not the remote users have authenticated.

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

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

A resource is reserved for RTP, and if a resource is not available for DTMF generation, that call out the DSX will fail. 23 calls work fine, as long as the 23rd call is not fighting for a resource with another call at that time.

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

The IAD 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.

Appendix B – Related Documents

For configuration guides, installation guides, white papers and more, visit ADTRAN's knowledge base at http://kb.adtran.com.

AOS A1.01 Command Line Reference Guide (13MB file) –
http://kb.adtran.com/article.asp?article=2219&p=2

Voice Quality Monitoring Config Guide -
http://kb.adtran.com/article.asp?article=2262&p=2

[Video]Understanding Voice Quality Monitoring in AOS -
http://kb.adtran.com/article.asp?article=2296&p=2

Integrated Traffic Monitoring Config Guide (Top Talkers Support) -
http://kb.adtran.com/article.asp?article=2157&p=2

Multi-VRF Config Guide –
http://kb.adtran.com/article.asp?article=2156&p=2

URL Filtering/Top Websites Reporting Config Guide -
http://kb.adtran.com/article.asp?article=2158&p=2

AOS Wireless Config Guide –
http://kb.adtran.com/article.asp?article=2078&p=2