6AOSCG0020-29C
November 2012

Configuration Guide

Configuring the AOS Voice Loopback Account

This configuration guide describes the function and configuration of the ADTRAN Operating System (AOS) voice loopback account. The AOS voice loopback account is configured using the AOS command line interface (CLI). This guide contains an overview of the user AOS voice loopback account feature, and describes how to create and configure a voice loopback account, assign a voice loopback account for media loopback, and establish and monitor voice loopback calls.

This guide consists of the following sections:

Overview

Transmission quality for calls and other types of media are monitored on AOS voice products using voice loopback accounts and media loopback accounts. These accounts can be used to make test calls and media sessions to verify the integrity of the media path between an AOS voice product and a specified endpoint.

Voice Loopback Account

A loopback account is a specific type of user account the AOS unit uses to place and receive Realtime Transport Protocol (RTP) loopback calls. Loopback calls allow you to actively troubleshoot voice issues. Loopback calls can originate from the CLI, or the configured loopback extension can automatically answer an incoming call. During the loopback call, you hear a mirrored copy of the transmitted audio path. The loopback call gives you the ability to verify two-way audio, test latency, and judge call quality.

At five second intervals, the voice loopback account sends compound Realtime Transport Control Protocol (RTCP) packets, which include RTCP sender report (RTCP SR) and RTCP receiver report (RTCP RR) packets as well as RTCP extended report (RTCP XR) blocks. These RTCP packets allow the user to monitor packet loss, packet duplication, jitter, and other signal quality metrics.

The RTCP SR contains the Network Time Protocol (NTP) and RTP timestamps and the sender's packet count and octet count. The RTCP RR contains the fraction lost, cumulative number of packets lost, extended highest sequence number received, interarrival jitter, last RTCP SR report timestamp, and delay since last RTCP SR.

NOTE

For more information on the contents of the RTCP SR and RTCP RR, refer to ***RFC 1889- RTP: A Transport Protocol for Real-Time Applications*** *sections 6.3.1 and 6.3.2.*

The RTCP XR contains the loss run-length encoding (RLE), duplicate RLE, statistics summary, and voice over Internet Protocol (VoIP) metrics.

NOTE

For more information on the contents of the RTCP XR report blocks, refer to ***RFC 3611 - RTP Control Protocol Extended Reports (RTCP XR)****.*

Media Loopback

Media loopback is a draft extension to the Session Description Protocol (SDP) that enables media sessions, such as VoIP and Video over Internet Protocol, to be established where the media is looped back to the transmitter. This is typically referred to as active monitoring of services. Like a voice loopback account, media loopback allows the integrity and quality of the media path to be actively monitored. Media loopback is enabled on AOS voice products by designating a voice loopback account for media loopback.

 Copyright © 2012 ADTRAN, Inc.

Hardware and Software Requirements and Limitations

The AOS voice loopback and AOS media loopback are available on AOS products as outlined in the *AOS Feature Matrix*, available online at https://supportforums.adtran.com.

AOS voice loopback accounts can only be configured from the CLI.

The AOS media loopback feature is only available on AOS products running AOS version A4.03 or later.

The RTCP reporting features are only available on AOS products running AOS version A5.01 or later.

Media loopback calls with **rtp-pkt-loopback** loopback type are only supported on AOS products running AOS version R10.1.0 or later.

SDP attributes of Media loopback calls must conform to version 13 of the Internet Engineering Task Force (IETF) media loopback draft (draft-ietf-mmusic-media-loopback-13). Of the two RTP payload types defined by the draft, only **direct payload format** is supported. This payload type specifies that the incoming RTP headers do not need to be preserved. The AOS unit will reject media loopback calls that lack this attribute.

As of AOS firmware release R10.5.0, the loopback interface supports IPv6 functionality and features. For more information about configuring IPv6 on the loopback interface, refer to the *Configuring IPv6 in AOS* configuration guide, available online at https://supportforums.adtran.com.

Configuring the AOS Voice Unit for Voice Loopback

There are three main steps that must be completed before voice loopback calls can be placed on the unit.

Step 1: Accessing the CLI on page 3

Step 2: Configuring a Loopback Account on page 4

Step 3: Creating a Dial Plan on page 8

Step 1: Accessing the CLI

All AOS products can be accessed using a computer with VT100 terminal emulation software. AOS products can also be accessed through the unit's Ethernet interface using a Telnet client.

To establish a connection to the NetVanta router CONSOLE port, you need the following items:

- PC with VT100 terminal emulation software
- Straight-through serial cable with a DB-9 (male) connector on one end and the appropriate interface for your terminal or PC communication port on the other end

> NOTE
>
> *You can find VT100 terminals on most PCs by navigating to **Start** > **Programs** > **Accessories** > **Communications** > **HyperTerminal** > **HyperTerminal**. When you have opened a HyperTerminal session, enter the settings described in Step 4 below.*

1. Connect the DB-9 (male) connector of your serial cable to the **CONSOLE** port on the front or rear panel of the unit.

 Copyright © 2012 ADTRAN, Inc.

2. Connect the other end of the serial cable to the PC with VT100 terminal emulation software.

> NOTE *Many PCs do not come with a standard serial port. A universal serial bus (USB) to serial adapter can be used instead. The drivers for the USB to serial adapter must be installed according to the manufacturer's instructions. If the USB to serial adapter is not properly installed on your PC, you will not be able to communicate with the AOS unit and you should seek support from the USB to serial adapter manufacturer.*

3. Provide power to the unit as appropriate. Refer to your unit's hardware installation guide for more details.

4. Once the unit is powered up, open a session in a VT100 terminal emulation program on the computer using the following settings: 9600 baud (or bps), 8 data bits, no parity bits, 1 stop bit, and no flow control. Press **<Enter>** to activate the AOS CLI.

5. Enter **enable** at the > prompt and enter the Enable mode password when prompted. The default password is **password**.

You can also access the CLI from a Telnet client. In order to do this, you must know the IP address of the AOS device. If you do not know the unit's IP address, you must use the **CONSOLE** port to access the CLI. To access the CLI using a Telnet client, follow these steps:

1. Connect the AOS device's **ETH 0/1** port to you computer's Ethernet interface using an Ethernet cross-over cable.

2. Open a Telnet client on your computer. You can access the Telnet client by navigating to **Start > Run** and entering **telnet 10.10.10.1** (Windows® XP). If you are running Windows Vista, you will need to turn on the Telnet client before you access it. To do this, navigate to **Start > Control Panel > Programs and Features > Turn Windows features on or off > Telnet Client Option** and select **OK**. **10.10.10.1** is the unit's default IP address. If you have changed your unit's IP address, you will need to enter that address.

3. Enter **enable** at the > prompt and enter the enable password when prompted. The default password is **password**.

Step 2: Configuring a Loopback Account

Before configuring an AOS voice loopback account, you must configure a Session Initiation Protocol (SIP) trunk through which the voice loopback account is registered. For more information on configuring a SIP trunk, refer to the *Configuring SIP Trunking and Networking for the NetVanta 7000 Series* and *Voice Traffic over SIP Trunks* configuration guides, available online at https://supportforums.adtran.com.

Once you have configured the SIP trunk, you can configure a loopback account. This is a specific type of user account that the AOS voice unit will use to place and receive the actual loopback call. A loopback extension has basically the same configurable call quality options and follows the same call routing rules as a standard user account. Loopback accounts also require the same configuration as any other user account in order to properly interact with the SIP server. At a minimum, you must configure the number of simultaneous loopback calls allowed on the system and the SIP registration options.

 Copyright © 2012 ADTRAN, Inc.

To configure the minimum requirements for a loopback account, follow these steps:

1. From the Global Configuration mode prompt, create a loopback account and enter the Voice Loopback Account Configuration mode using the following command:

   (config)#**voice loopback** <*extension*>

   The <*extension*> variable specifies the desired extension for the voice loopback account.

2. From the Voice Loopback Account Configuration mode, specify the number of simultaneous loopback calls allowed on the system using the following command:

   (config-LB-extension)#**appearances** <*value*>

   The <*value*> variable specifies the number of simultaneous loopback calls allowed on the system. The valid range is **1** to **5**.

3. From the Voice Loopback Account Configuration mode, configure the SIP registration options for the loopback account using the following command:

   (config-LB-extension)#**sip-identity** <*station*> <*Txx*> **register [auth-name** <*username*>**]**
   [password <*password*>**]**

   The <*station*> variable specifies the SIP station of the voice loopback account to use with the SIP trunk.

   The <*Txx*> variable specifies the SIP trunk through which to register the user to the SIP server. The trunk is specified in the format Txx (for example, **T01**).

   The **register** parameter registers the loopback account to the SIP server.

   The optional **auth-name** <*username*> parameter sets the user name that will be required as authentication for registration to the SIP server.

   The optional **password** <*password*> parameter sets the password that will be required as authentication for registration to the SIP server.

The following example shows the minimal settings for a voice loopback account:

(config)#**voice loopback 1001**
(config-LB-1001)#**appearances 5**
(config-LB-1001)#**sip-identity 1001 T01 register auth-name 1001 password password**

Copyright © 2012 ADTRAN, Inc.

Optional Voice Loopback Account Settings

Additional options can be configured from the Voice Loopback Account Configuration mode to change how the voice loopback account operates. If you are using voice loopback to monitor call quality or troubleshoot voice issues, it is recommended that the voice loopback account use the same configuration settings as other voice users on the unit. *Table 1* below lists additional voice loopback account commands and explains their functions.

Table 1. Optional Voice Loopback Account Settings

Command	Explanation
alc	The **alc** command enables auto level control (ALC). ALC reduces RTP received signals that are out of specification to the predefined levels. It is not necessary to enable ALC on those networks that guarantee signal levels to be within specification. Use the **no** form of this command to disable this feature.
caller-id [name *<name>* **|** **number** *<number>***]**	The **caller-id** command specifies a name or number to display as the caller ID information. By default, the caller ID will display the extension number used when creating the loopback account. If no name or alternate number is specified, the loopback extension number will be displayed. Use the **no** form of this command to return to the default value. The **name** *<name>* parameter specifies the name of the loopback account. The **number** *<number>* parameter specifies the number of the loopback account.
codec-group *<name>*	The **codec-group** command specifies the coder-decoder (CODEC) list to be used by this account. Use the **no** form of this command to remove the CODEC list from the account. The *<name>* variable specifies the CODEC list to be used for this account.
echo-cancellation	The **echo-cancellation** command is used to improve voice quality for packetized-based voice calls, such as VoIP or Media Gateway Control Protocol (MGCP). Use the **no** form of this command to disable this feature. By default, **echo-cancellation** is enabled.
nls	The **nls** command enables the non-linear suppression (NLS) option for the user. This option sets the echo canceller to reduce acoustic echo. Use the **no** form of this command to disable this feature.
num-rings *<value>*	The **num-rings** command specifies the number of rings before the loopback account answers (when called). By default, **num-rings** is set to **0**. Use the **no** form of this command to return to the default value. The *<value>* variable specifies the number of rings before answering. Specify **0** through **9** rings. Entering **0** specifies answering the call immediately.
plc	The **plc** command enables packet loss concealment (PLC). PLC is used to prevent choppy connections by replacing a lost packet with another voice packet in the data stream. By default, PLC is enabled. Use the **no** form of this command to disable this feature.

 Copyright © 2012 ADTRAN, Inc.

Table 1. Optional Voice Loopback Account Settings *(Continued)*

Command	Explanation
rtp delay-mode [adaptive | fixed]	The **rtp delay-mode** command configures the RTP jitter buffer packet delay mode settings. RTP is used to prevent static on voice connections by enhancing the quality of the packet delivery. By default, the RTP delay mode is set to **adaptive**. This allows minimal latency by adjusting the average packet delay based on the conditions of the network. Use the **no** form of this command to return to the default setting. The **adaptive** parameter configures the RTP jitter buffer packet delay to adjust during a call based on network conditions. The **fixed** parameter configures the RTP jitter buffer packet delay to remain constant.
rtp dtmf-relay [inband | nte <*value*>**]**	The **rtp dtmf-relay** command configures the method by which RTP dual tone multi-frequency (DTMF) events are relayed. The dial digits can be sent inband or out-of-band (OOB) of the voice stream. By default, the **rtp dtmf-relay** is set for **nte 101**. Use the **no** form of this command to return to the default value. The **inband** parameter specifies that RTP DTMF events be relayed inband in the RTP stream. The **nte** <*value*> parameter specifies that RTP DTMF event value be relayed OOB using named telephone events (NTEs). Enter an NTE value between **96** and **127**.
rtp frame-packetization <*value*>	The **rtp frame-packetization** command configures the RTP frame packetization time in milliseconds for individual trunks and users. By default, the **rtp frame-packetization** time is set to **20** milliseconds on all trunks and users. Use the **no** form of this command to return to the default value. The <*value*> variable configures the RTP frame packetization time value in milliseconds. Select from **10**, **20**, or **30** milliseconds.
rtp packet-delay [fax <*value*> **| maximum** <*value*> **| nominal** <*value*>**]**	The **rtp packet-delay** command configures the maximum RTP packet delays. This command is used to set the allowable limits of latency on the network. By default, the RTP packet delays are **fax 300**, **maximum 100**, and **nominal 50**. Use the **no** form of this command to return to the default value. The **fax** <*value*> parameter sets the fax delay time value in milliseconds. Range is **0** to **500** milliseconds. The **maximum** <*value*> parameter sets the maximum delay time value in increments of 10 milliseconds. Range is **40** to **320** milliseconds. The **nominal** <*value*> parameter sets the nominal delay time value in increments of 10 milliseconds. Range is **10** to **240** milliseconds.
rtp qos dscp <*value*>	The **rtp qos dscp** command to configure the RTP differentiated services code point (DSCP) value. By default, the RTP quality of service (QoS) parameter for DSCP is **46**. Use the **no** form of this command to return to the default value. The <*value*> variable configures the RTP value for DSCP. The DSCP values are **0** to **63**.

Copyright © 2012 ADTRAN, Inc.

Table 1. Optional Voice Loopback Account Settings *(Continued)*

Command	Explanation
shutdown	The **shutdown** command disables the loopback account and prevents future calls into the account. By default, no loopback accounts are configured. Use the **no** form of this command to reactivate the loopback account.
sip-identity *<station>* *<Txx>* **[register | register auth-name** *<username>* **password** *<password>***]**	The **sip-identity** command configures the SIP registration options for the user. Use the **no** form of this command to disable the setting. The *<station>* variable specifies the station to be used for SIP trunk (e.g., **station extension**). The *<Txx>* variable specifies the SIP trunk through which to register the server. The trunk is specified in the format Txx (e.g., **T01**). The **register** parameter registers the user to the server. The optional **auth-name** *<username>* parameter sets the user name that will be required as authentication for registration to the SIP server. The optional **password** *<password>* parameter sets the password that will be required as authentication for registration to the SIP server.

Step 3: Creating a Dial Plan

A dial plan must also be configured for the unit to accept and route calls made from the loopback account. The dial plan notifies the AOS voice unit when to stop collecting the digits being dialed and begin forwarding a phone call. Programmed number patterns and types govern the telephone numbers allowed by AOS voice products for inbound and outbound calls. Number-complete templates can be created and stored in the dial plan. The AOS voice unit listens for digits and looks for a match against the number-complete templates in the dial plan. As soon as the digits dialed by the user match a pattern in the dial plan, the call is routed by the switchboard. If the digits dialed do not match any of the number-complete templates, the call is eventually routed by the switchboard after a timeout period expires. In addition to number patterns, call types are defined in the dial plan, allowing the system to recognize dialed numbers as a particular type of call (local, long distance, toll free, etc.). For more information on how to configure your unit's dial plan, refer to the *Configuring the Switchboard and Dial Plan in AOS* configuration guide, available online at https://supportforums.adtran.com.

Use the **voice dial-plan** command from the Global Configuration mode prompt to create a dial plan entry.

(config)#**voice dial-plan** *<pattern id>* *<group>* *<pattern>* [**default** | **none** | *<ndt name>*]

The *<pattern id>* variable specifies the identification number to assign to this dial plan. Range is **1** to **255**.

The *<group>* variable describes the type of call the dial plan entry will represent. Available choices are: **900-number**, **always-permitted**, **extensions**, **internal-operator**, **international**, **local**, **long-distance**, **operator-assisted**, **specify-carrier**, **toll-free**, **user1**, **user2**, **user3**.

 Copyright © 2012 ADTRAN, Inc.

The <*pattern*> variable specifies the dialing pattern that will represent this dial plan entry. You can enter an exact phone number, or you can use wildcards to help define rejected numbers. The available wildcards for this command are:

0 - 9	Match the exact digit(s) only
X	Match any single digit 0 through 9
N	Match any single digit 2 through 9
M	Match any single digit 1 through 8
$	Match any number string dialed
[]	Match any digit in the list within the brackets (for example, [1,4,6])
,()	Formatting characters that are ignored but allowed
-	Use within brackets to specify a range, otherwise ignored

The following are example template entries using wildcards:

1) NXX-XXXX	Match any 7-digit number beginning with 2 through 9
2) 1-NXX-NXX-XXXX	Match any number with a leading 1, then 2 through 9, then any 2 digits, then 2 through 9, then any 6 digits
3) 555-XXXX	Match any 7-digit number beginning with 555
4) XXXX$	Match any number with at least 5 digits
5) [7,8]$	Match any number beginning with 7 or 8
6) 1234	Match exactly 1234

Some template number rules:

1. All brackets must be closed with no nesting of brackets and no wildcards within the brackets.

2. All brackets can hold digits and commas, for example: [1239]; [1,2,3,9]. Commas are implied between numbers within brackets and are ignored.

3. Brackets can contain a range of numbers using a hyphen, for example: [1-39]; [1-3,9].

4. The $ wildcard is only allowed at the end of the template, for example: 91256$; XXXX$.

The optional **default** parameter sets the named-digit-timeout to the default value. The default value is set with the **voice timeouts interdigit** command.

The optional **none** parameter indicates that no named-digit-timeout is associated with this dial plan entry.

The optional <*ndt name*> variable specifies the named-digit-timeout to associate with this dial plan entry. The named-digit-timeout is assigned a timeout value with the **voice timeouts named-digit-timeout** command.

 Copyright © 2012 ADTRAN, Inc.

The following is an example of how to use the dial plan command to create an entry. Please be aware that copying these commands verbatim into your dial plan configuration is not recommended. Rather, this example is provided to illustrate the use of wildcards when entering patterns.

(config)#**voice dial-plan 0 always-permitted 911**
(config)#**voice dial-plan 1 local NXX-XXXX**
(config)#**voice dial-plan 2 long-distance 1-NXX-NXX-XXXX**
(config)#**voice dial-plan 3 toll-free 1-800-NXX-XXXX**
(config)#**voice dial-plan 4 toll-free 1-888-NXX-XXXX**
(config)#**voice dial-plan 5 toll-free 1-877-NXX-XXXX**
(config)#**voice dial-plan 6 toll-free 1-866-NXX-XXXX**
(config)#**voice dial-plan 7 international 011-$**

Enabling Media Loopback on Voice Loopback Accounts

Media loopback enables media sessions, such as VoIP and Video over Internet Protocol, to be established where the media is looped back to the transmitter (typically referred to as active monitoring of services). Once a voice loopback account is designated for media loopback, it is registered with the switchboard under the media loopback alias. Only one voice loopback account can be used for media loopback.

Media loopback calls are automatically answered by the unit after a specified number of rings (refer to *num-rings <value> on page 6*) if the SDP offer conforms to version 13 of the IETF media loopback draft. Only media loopback calls with **direct payload format** payload type are supported. The AOS unit will reject media loopback calls that lack this attribute.

Use the **media loopback** command from the Voice Loopback Account Configuration mode to designate the voice loopback account to be used for media loopback. For example:

(config)#**voice loopback 5555**
    Configuring Existing Loopback Account “5555”.
(config-LB-5555)#**media-loopback**

If a voice loopback account is already designated for media loopback and you attempt to designate another voice loopback account for media loopback, the following error message will be displayed:

(config)#**voice loopback 1001**
    Configuring Existing Loopback Account “1001”.
(config-LB-1001)#**media-loopback**
%Error: Loopback account 5555 is already designated for media-loopback

In order to change the voice loopback account designated for media loopback, you must use the **no media-loopback** command in the Voice Loopback Account Configuration mode of the current media loopback account or delete the account altogether.

Placing a Voice Loopback Call

Voice loopback calls can originate from the CLI or the configured loopback extension can automatically answer an incoming call made to the extension assigned to the voice loopback account.

 Copyright © 2012 ADTRAN, Inc.

Establishing a Voice Loopback Call

Loopback calls are placed from the CLI using the following command from the Enable mode prompt:

#**voice loopback-call start from** <*number*> **to** <*number*>

The **start from** <*number*> parameter starts a loopback call from the specified extension number. This number should be the number of the voice loopback account.

After issuing the command with the correct loopback extension and a valid number, the called party will receive the loopback call. Upon answering, the called party will hear a copy of the audio that they produced. When the called party hangs up, the call is terminated as if it were a normal call.

Terminating a Voice Loopback Call

In the event that a call is not terminated by the called party, the administrator can terminate the loopback call from the Enable mode in the CLI. The administrator has three options when terminating calls. All three options are shown below.

The following command terminates loopback calls for the specified account:

#**voice loopback-call stop account** <*number*>

The following command terminates all loopback calls:

#**voice loopback-call stop all**

The following command terminates a specific loopback call based on the identification number of the call:

#**voice loopback-call stop id** <*number*>

For all of the above commands, the **stop** parameter terminates active loopback calls. The **account** <*number*> parameter specifies the account on which to terminate voice loopback calls. The **all** parameter specifies all loopback calls for all voice loopback accounts should be terminated. The **id** <*number*> parameter specifies the identification number of the voice loopback call to terminate. The identification number is automatically assigned in ascending order by the unit when a voice loopback call is placed. To determine the identification number of the voice loopback call you would like to terminate, refer to *Monitoring Voice Loopback Calls Using the CLI on page 12*.

 Copyright © 2012 ADTRAN, Inc.

Monitoring Voice Loopback Calls Using the CLI

The **show voice loopback calls** command, issued from the Enable mode, displays information specific to loopback calls. As seen below, the **show voice loopback calls** command displays the ID, extension, CODEC, status, number dialed, and duration:

```
>enable
#show voice loopback calls
ID   Extension   Codec   Status                        Number              Duration
---------------------------------------------------------------------------- (hour:min:sec)
1    1123                <ENDED (invalid number)       -> 8837655               :01
2    1123                Calling                       -> 4001                  :05
3    1123                <ENDED (no appearances)       <- 4001                  :01
4    1123        G729    Connected                     -> 4001                  :07
```

Additionally, the status of the loopback extension can be displayed by using the **show voice extension** command from the Enable mode prompt. The **show voice extension** command displays every configured extension’s status, whether the phone is in use, the do not disturb (DND) option is active, or the extension is currently being forwarded. Loopback extensions cannot be configured with the DND and call forwarding options. Below is sample output of the **show voice extensions** command.

```
>enable
#show voice extension
Account ID  Idle/Ring/Busy    Available    DND    FWD
-------------------------------------------------------

T01           Idle              *           -      -
T06           Idle              *           -      -
T02           Idle              *           -      -
5200          Idle              *           -      -
6000          Idle              *           -      -
6001          Idle              *           -      -
6002          Idle              *           -      -
6003          Idle              *           -      -
T03           Idle              *           -      -
```

 Copyright © 2012 ADTRAN, Inc.