



## Configuration Guide

### Switchboard and Dial Plan

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This configuration guide explains the definitions and the functionality of the switchboard versus the dial plan in the ADTRAN voice products. A thorough knowledge of the differences in the roles and setup of these two tasks is essential to successful implementation of voice applications in ADTRAN voice products.

ADTRAN voice products combine data and voice into a single platform controlled by the ADTRAN Operating System (AOS). For detailed information regarding specific command syntax, refer to the *AOS Command Reference Guide* available on the *AOS Documentation CD* shipped with your AOS unit or online at [www.adtran.com](http://www.adtran.com).

This guide consists of the following sections:

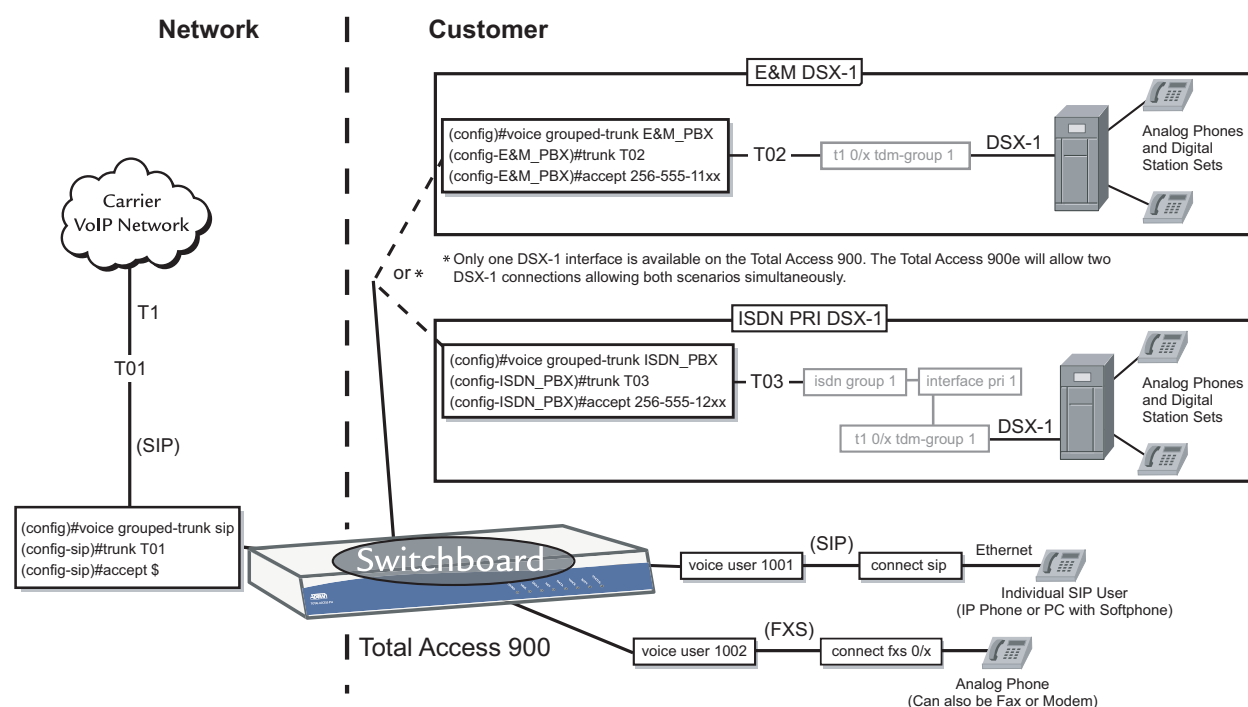
- *Understanding the Switchboard* on page 2
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## Understanding the Switchboard

The switchboard in AOS voice products ensures that inbound and outbound phone calls are routed to the proper interface. Inbound/outbound accept and reject numbers are programmed for voice trunks and referenced by the switchboard whenever a call enters or leaves the unit.

For every trunk (ISDN, FXO, SIP, E&M) connected to an AOS unit that carries voice calls, a voice trunk must be configured. Each voice trunk is then placed into a trunk group. Within the trunk group, accept and reject numbers used by the switchboard to make call routing decisions are collectively assigned to the included voice trunks.

Individual voice users can also be created. These users can be SIP phone users treated as IP endpoints, or they can be analog users assigned to a specific FXS interface. In both cases, a unique phone number is programmed for each user and this number is referenced by the switchboard.



All voice trunks and individual voice users are connected to the switchboard inside the AOS unit. Since the switchboard is where call routing decisions take place, it is important to ensure direct inward dial numbers (DIDs) have been properly configured for trunk groups and individual voice users.



*Correct placement of incoming accept/reject numbers and wildcards on voice trunks is critical for successful voice implementation through AOS voice products. It is highly recommended that you double check placement of these values in your configuration to ensure they have been properly configured in a trunk group and not accidentally placed in the dial plan configuration.*

## Configuring Accept/Reject Numbers for Voice Trunks



Complete basic router configuration prior to beginning the following example. Enable Ethernet interfaces and assign all IP addresses. For SIP trunks, set the media gateway and configure SIP parameters (other than those pertaining to the switchboard). For step-by-step instructions on how to configure SIP trunks, refer to **Voice Traffic over SIP Trunks** configuration guide. Also make sure to enable DSX-1 and FXS physical interfaces. For assistance configuring physical interfaces, refer to the **PRI Signaling for DSX-1 and FXS User** configuration guides. All configuration guides are available on the AOS Documentation CD shipped with your unit or online at [www.adtran.com](http://www.adtran.com).

Follow these steps to configure accept/reject numbers and wildcards for voice trunks. These numbers are referenced by the switchboard to make call routing decisions.

### Step 1

Create a voice trunk for each connection type. All voice trunks are created using the following command.

```
(config)#voice trunk <trunk ID> type [analog | isdn | sip | t1-rbs]
```

<trunk ID> Specifies a two-digit identifier to assign to this voice trunk in the format <Txx> (e.g., T01).



This configuration guide assumes that SIP trunks have already been created. For existing SIP trunks, re-enter the trunk configuration mode and proceed to **Step 3** on page 4.

### Step 2

Associate the analog FXO, ISDN PRI, or T1-RBS voice trunk with a physical interface. This command varies depending on the type of signaling the trunk is being configured for.

#### Analog FXO

```
(config-Txx)#connect fxo <slot/port>
```

<slot/port> Specifies the slot/port of the FXO trunk. The integral FXO interface available on any Total Access 900e is numbered 0/0. The 8 FXO ports on the Total Access 924/924e with 16 FXS and 8 FXO ports are numbered 0/1 to 0/8.



This option is currently available via the integral FXO interface on any Total Access 900e product. FXO interfaces are also available on the Total Access 924/924e with 16 FXS and 8 FXO interfaces.

### ISDN PRI

(config-Txx)#**connect isdn-group** <number>

<number> Specifies the ISDN group number. Range is 1 to 255. The number you place here references a previously-created ISDN group. For more information about configuring ISDN groups, refer to the *PRI Signaling for DSX-1* configuration guide available on the AOS Documentation CD shipped with your unit or online at [www.adtran.com](http://www.adtran.com).

### T1-RBS

(config-Txx)#**connect t1** <slot/port> **tdm-group** <number>

<slot/port> Specifies the slot/port of the T1-RBS trunk. Slots are numbered 0 to 0 (slot 0 is the integrated System Controller slot), and ports are numbered 1 to n.

<number> Specifies the TDM group number. Range is 1 to 255. The number you place here references a previously-created TDM group.

## Step 3

By default, calls can not be routed between two voice trunks. Therefore, if you create more than one voice trunk and would like calls to be routed between them, issue the following command within the voice trunk configuration on each trunk:

(config-Txx)#**no reject-external**

Exit trunk configuration mode when finished.

(config-Txx)#**exit**

## Step 4

Create a trunk group where you will place all voice trunks to be associated with common accept/reject numbers or wildcards.

(config)#**voice grouped-trunk** <name>

<name> Specifies the name given to this trunk group. It can be helpful to use names that will identify the type of trunks associated with this group (e.g., MAIN, SIP, ISDN, DSX, etc.).

## Step 5

Add appropriate trunk(s) to the newly created trunk group.

(config-trunk group)#**trunk** <trunk ID>

<trunk ID> Identifies existing voice trunks to add to the trunk group in the format <Txx> (e.g., **T01**).

## Step 6

Define accept and reject numbers for this group of trunks. Accept numbers are the numbers you want to assign to the trunk (typically DIDs).

(config-trunk group)#**accept** <pattern>

<pattern> Specifies the accept numbers for the trunks. You can enter an exact phone number, or you can use wildcards to help define accepted numbers. The available wildcards for this command are:

**M** = any digit 1 to 8

**X** = any single digit (0 to 9)

**N** = any digit 2 to 9

**\$** = any number of digits of any value

[abc] = any digit contained in the bracketed list



*Do not use dashes, commas, spaces, etc., inside the brackets. Commas are implied between numbers in the brackets.*



*It is important to be as specific as possible when configuring accept numbers for ISDN PRI and T1-RBS voice trunks. Generic wildcards (e.g., \$) placed on more than one trunk can result in misrouted calls. Refer to the section **Accept Number Configuration and Caveats** on page 6 for examples on how to properly configure accept numbers.*

The option to enter reject numbers is also available, although reject numbers are not required for proper switchboard function. Reject numbers allow you to restrict calls from specified numbers to voice trunks.

(config-trunk group)#**reject** <pattern>

<pattern> Specifies the reject numbers for the trunks. You can enter an exact phone number, or you can use wildcards to help define rejected numbers. The available wildcards for this command are:

**M** = any digit 1 to 8

**X** = any single digit (0 to 9)

**N** = any digit 2 to 9

**\$** = any number of digits of any value

[abc] = any digit contained in the bracketed list



*Do not use dashes, commas, spaces, etc., inside the brackets. Commas are implied between numbers in the brackets.*

Exit trunk group configuration mode when finished.

(config-trunk group)#**exit**

## Step 7

Save the configuration to memory.

(config)#**do write memory**

## Accept Number Configuration and Caveats

As you implement accept numbers on voice trunks, consider the following discussion on common mistakes and the proper solutions.

### Be Specific with Accept Numbers for ISDN PRI and/or T1-RBS

Since the SIP voice trunk is typically configured with the \$ wildcard (any number of digits of any value), it is extremely important to be as specific as possible when configuring accept numbers for ISDN PRI and/or T1-RBS voice trunks. AOS voice products always find the best/longest match for a number when deciding to which interface to route a call. The following is an example of an accept number configuration for voice trunks that is too generic and will not execute properly:

```
(config)#voice grouped-trunk SIP
(config)#trunk T01
(config-T01)#accept $
```

```
(config)#voice grouped-trunk DSX
(config)#trunk T02
(config-T02)#accept NXX-NXX-XXXX
(config-T02)#accept NXX-XXXX
```

The accept numbers configured for the DSX trunk group here are so generic that ALL calls (including SIP) will be passed out the DSX port. Remember that AOS voice products find the best/longest match for any number. In this example, all seven- or ten-digit calls match the accept numbers that were specified for the DSX trunk group. It is true that a seven- or ten-digit call also matches the \$ wildcard on the SIP trunk, but the DSX trunk numbers are more specific than \$, while still being generic enough to accept all seven- or ten-digit numbers.



*Accept numbers are being configured here, not dialing patterns. Refer to [Understanding the Dial Plan](#) on page 7 to configure dial pattern numbers.*

### Accurately Represent the Range of DID Numbers

To save time and effort, it is tempting to avoid tedious configuration of DID numbers. However, you must take into consideration that other local businesses or residences might own the numbers you have inadvertently included in the accept statement. For example:

```
(config)#voice grouped-trunk DSX
(config)#trunk T02
(config-T02)#accept 256-555-1XXX
```

The accept statement in this configuration is sufficient if a business actually has 1000 DID numbers assigned to them. However, many businesses do not own that many DID numbers. In those cases, the configuration specified works *most* of the time, but not all of the time. The problem arises when users try to dial the outside phone number of a residence or business whose number falls into the range of DID numbers specified in the accept range on the DSX-1 interface. When this situation occurs, the call never makes it to the outside network. Instead, it is routed via the DSX-1 to the company's own phone system and auto-attendant.

The solution is to use ranges in brackets. The following example shows how to configure accept numbers for a DID range of 256-555-(1260 to 1500):

```
(config)#voice grouped-trunk DSX
(config-DSX)#accept 256-555-12[6789]X      (covers 1260 to 1299)
(config-DSX)#accept 256-555-1[34]XX      (covers 1300 to 1499)
(config-DSX)#accept 256-555-1500
```



*Do not use dashes, commas, spaces, etc., inside the brackets. Commas are implied between numbers in the brackets.*

## Configuring Individual Voice Users

Individual voice users can either be configured for SIP or directly assigned to physical FXS interfaces.

```
(config)#voice user <extension>
      <extension> Specifies voice user's extension.
```

- Configure a voice user for SIP:

```
(config-voice user)#connect sip
```

- Connect a voice user to a physical FXS interface:

```
(config-voice user)#connect fxs <slot/port>
```

*<slot/port> Specifies the slot/port of the FXS interface. FXS interfaces are numbered 0/1 to 0/n, where n is the number of FXS ports for this unit.*

Exit voice user and terminal configuration mode and save the configuration to memory.

```
(config-voice user)#exit
(config)#exit
#write memory
```

## Understanding the Dial Plan

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.).

A named-digit-timeout (NDT) can be created and associated with a number-complete template to indicate the amount of time to wait for additional digits to be dialed before routing the call. Setting the NDT to **default**, uses the system interdigit timeout. Setting the NDT to **none** will immediately route the call after a template match. If no NDT timeout value is specified, the system will assign a value of 0.

## Configuring the Dial Plan

It is possible for AOS voice units to successfully route calls through the unit without any dial plan entries configured. However, dial plan entries may be needed in certain setups for proper execution of inbound and outbound calls.



*Complete basic router configuration prior to beginning the following examples. Enable Ethernet interfaces and assign all IP addresses. For SIP trunks, set the media gateway and configure detailed SIP parameters (other than those pertaining to the switchboard). For step-by-step instructions on how to configure SIP trunks, refer to the **Voice Traffic over SIP Trunk** configuration guide. Also make sure to enable DSX-1 and FXS physical interfaces. For assistance configuring physical interfaces, refer to the **PRI Signaling for DSX-1 and FXS User** configuration guides. Finally, accept/reject numbers should be configured on voice trunks as outlined in **Understanding the Switchboard** on page 2.*

*All configuration guides are available on the AOS Documentation CD shipped with your unit or online at [www.adtran.com](http://www.adtran.com)*

Use the **voice dial-plan** command to create a dial plan entry.

```
(config)#voice dial-plan <pattern ID> <group> <pattern> [default | none | <NDT name>]
```

<pattern ID> Specifies the identification number to assign to this dial plan. Range is 1 to 255.

<group> 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**.

<pattern> 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:

**M** = any digit 1 to 8

**X** = any single digit (0 to 9)

**N** = any digit 2 to 9

[*abc*] = any digit contained in the bracketed list

**default** Optional. Sets the named-digit-timeout to the default value. The default value is set with the **voice timeouts interdigit** command.

**none** Optional. Indicates that no named-digit-timeout is associated with this dial plan entry.

<NDT name> Optional. 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.

The following are a few examples of how to use the dial plan command to create an entry. Please be aware that it is not recommended to copy the commands verbatim into your dial plan configuration. Rather, these examples are provided to illustrate use of wildcards when entering patterns.

```
(config)#voice dial-plan 1 always-permitted 9-911
```

```
(config)#voice dial-plan 2 local 9-NXX-XXXX
```



Brackets are used to define a range of numbers. This example defines the range 2000 to 6999.

```
(config)#voice dial-plan 3 extensions [23456]XXX
```



*Do not use dashes, commas, spaces, etc., inside the brackets. Commas are implied between numbers in the brackets.*

Use the **voice timeouts named-digit-timeout** command to create a named-digit-timeout and assign it a value.

```
(config)#voice timeouts named-digit-timeout <name> <value>
```

<name> Specifies a name for the named-digit-timeout.

<value> Indicates the value in seconds for the named-digit-timeout.

Creating a named-digit-timeout does not require a value. If a value is not specified, the system assigns the value of 0 seconds.



*When removing a named-digit-timeout and its value, if the NDT is assigned to a dial plan entry, then the deletion is not allowed. The dial plan entry must be removed first and added back into the system without the NDT association.*

## Dial Plan Caveats

As you create functional dial plan entries, consider the following examples of dial plan configurations that **do not** work and possible solutions.



*Users can bypass the number-complete timeout by typing # after completing a dial string. This tone forces the end-of-number decision to be executed immediately.*

### Example 1

```
(config)#voice dial-plan 1 local NXX-XXXX
```

```
(config)#voice dial-plan 2 long-distance NXX-NXX-XXXX
```

In this example, all numbers dialed (whether they are seven-digit local or ten-digit long distance) always match the first entry only. The reason for this match is because the local template is too similar to the beginning of the long distance template. Always matching the first entry in this case causes a problem when users attempt to dial a long distance number. Even though ten digits are needed to complete a long distance call, AOS voice units stop listening for digits after the seventh one is dialed.

### Solutions

- Place a **1** in the long distance dial plan entry. Instruct users to dial a **1** prior to placing long distance phone calls.

```
(config)#voice dial-plan 1 local NXX-XXXX
```

```
(config)#voice dial-plan 2 long-distance 1-NXX-NXX-XXXX
```

Or

- Create a named-digit-timeout (**short1**) with a value (**5**) and associate it with the **local** dial plan entry.

```
(config)#voice timeouts named-digit-timeout short1 5
(config)#voice dial-plan 1 local NXX-XXXX short1
(config)#voice dial-plan 2 local NXX-NXX-XXXX
```

Or

- Remove the seven-digit dial plan entry, leaving only the ten-digit dial plan entry in the configuration. When users dial a ten-digit long distance call, the number will match the ten-digit dial plan entry and be passed to the switchboard immediately (assuming the user dialed the number with less than four seconds between each digit). When users dial a seven-digit local call, AOS voice units pass the call after the number-complete timeout expires.

```
(config)#voice dial-plan local NXX-NXX-XXXX
```



*This solution works because of the number-complete timeout. The default value for this timeout is four seconds. Users have four seconds to complete the number they are dialing before the call is passed to the switchboard. When seven digits are dialed in this solution, AOS voice units wait for three more digits to match the ten-digit number template that has been specified. However, after four seconds have passed, the number is passed to the switchboard anyway.*

*Users may have to wait a few seconds after local dialing is complete before the call passes through the AOS voice unit. Although most users will not even notice the delay, you can manually adjust the timeout to last between 1 and 16 seconds using the following command:*

```
(config)#voice timeouts interdigit <1-16>
```

## Example 2

```
(config)#voice dial-plan 1 local NXX-XXXX
(config)#voice dial-plan 2 long-distance 1-NXX-NXX-XXXX
(config)#voice dial-plan 3 extensions XXXX
```

In this example, the local and long distance plans have been defined in a way that would execute properly if they were the only two entries configured. However, the third dial plan entry for in-house extensions specifies a four-digit template that any number pattern will initially match (except those that begin with **1**). Therefore, all seven-digit local numbers will match **voice dial-plan 3** and be passed to the switchboard after only the fourth digit has been dialed. The result is the inability to successfully reach local phone numbers.

## Solutions

- Remove the four-digit extension dial plan entry, leaving the seven-digit local and ten-digit long distance entries. When users dial a seven-digit local number, it matches the local dial plan entry and immediately passes to the switchboard (assuming it took the user less than four seconds to enter the number). Likewise, a ten-digit long distance number matches the long distance dial plan entry. When users dial a four-digit extension, AOS voice units pass the call after the number-complete timeout expires (refer to the note in *Example 1* on page 9).

```
(config)#voice dial-plan 1 local NXX-XXXX
(config)#voice dial-plan 2 long-distance 1-NXX-NXX-XXXX
```

Or

- Place the number **9** in the local and long distance dialing templates and change the first digit in the extension dialing template to the wildcard **M**.

```
(config)#voice dial-plan 1 local 9-NXX-XXXX
```

```
(config)#voice dial-plan 2 long-distance 91-NXX-NXX-XXXX
```

```
(config)#voice dial-plan 3 extensions MXXX
```

Instruct users to dial a **9** to reach an outside line before they dial a local or long distance number. In addition, users need to dial a **1** prior to long distance numbers. Most users are already familiar with this common dialing practice.



*The **M** wildcard placed at the beginning of the four-digit extension in **voice dial-plan 3** matches numbers 1 through 8. It is important that extensions do not start with the number **9** so that they will match the extension template in **voice dial-plan 3** and so that they do not interfere with the **9** used to indicate outside local or long distance calls.*

## Troubleshooting the Dial Plan and Switchboard

There are numerous commands available in AOS voice products to assist with troubleshooting. Debug commands display real-time debug messages on your terminal or telnet screen. The **debug voice switchboard** command allows you to see call setup, routing, and tear down through the switchboard in AOS voice products. The following is example output of the **debug voice switchboard** command:

```
(config)#debug voice switchboard
```

```
1970.01.04 23:24:59 SWITCHBOARD.StateMachine Called the call routine from Idle
1970.01.04 23:24:59 SWITCHBOARD.CCM isMappable:
1970.01.04 23:24:59 SWITCHBOARD.CCM : Call Struct 0x3549810 : Call-ID = 0x00000008
1970.01.04 23:24:59 SWITCHBOARD.CCM : Org Acct = 9026 Dst Acct = T01
1970.01.04 23:24:59 SWITCHBOARD.CCM : Org Port ID = 0x00000001 Dst Port ID = 0x00000000
1970.01.04 23:24:59 SWITCHBOARD.CCM isMappable: Call Connection Type is TDM_TO_RTP
1970.01.04 23:24:59 SWITCHBOARD.CCM isMappable: Reserving RTP Channel 0/2.1
1970.01.04 23:24:59 SWITCHBOARD.CCM isMappable: Creating SDP Offer
1970.01.04 23:24:59 SWITCHBOARD.CCM getBestMatchCodecList: G729 PCMU
1970.01.04 23:24:59 SWITCHBOARD.StateMachine Call sent from 9026 to T01 (9999)
1970.01.04 23:24:59 SWITCHBOARD.StateMachine SB State Change from Idle to Delivering on CallID 8
1970.01.04 23:24:59 SWITCHBOARD.StateMachine Called the deliverResponse routine from Delivering
1970.01.04 23:24:59 SWITCHBOARD.StateMachine DeliverResponse(accept) sent from T01 to 9026
1970.01.04 23:24:59 SWITCHBOARD.StateMachine SB State Change from Delivering to Delivering on CallID 8
1970.01.04 23:24:59 SWITCHBOARD.StateMachine Called the clearCall routine from Delivering
1970.01.04 23:24:59 SWITCHBOARD.StateMachine SB State Change from Delivering to Clearing on CallID 8
1970.01.04 23:24:59 SWITCHBOARD.StateMachine Called the clearResponse routine from Clearing
1970.01.04 23:24:59 SWITCHBOARD.StateMachine SB State Change from Clearing to CallIdlePending on CallID 8
1970.01.04 23:24:59 SWITCHBOARD.CCM release:
1970.01.04 23:24:59 SWITCHBOARD.CCM : Call Struct 0x3549810 : Call-ID = 0x00000008
1970.01.04 23:24:59 SWITCHBOARD.CCM : Org Acct = 9026 Dst Acct = T01
1970.01.04 23:24:59 SWITCHBOARD.CCM : Org Port ID = 0x00000001 Dst Port ID = 0x00000000
1970.01.04 23:24:59 SWITCHBOARD.CCM : SDP Offer = 0x03549c10, ref = 1 (127.0.0.2:10048)
1970.01.04 23:24:59 SWITCHBOARD.CCM : RTP Channel = 0/2.1
1970.01.04 23:24:59 SWITCHBOARD.CCM release: Call Connection Type is TDM_TO_RTP
1970.01.04 23:24:59 SWITCHBOARD.CCM release: Releasing RTP Channel 0/2.1
1970.01.04 23:24:59 SWITCHBOARD.CCM firewallReleaseCall: No action taken, firewall traversal is not enabled
1970.01.04 23:24:59 SWITCHBOARD.StateMachine ClearResponse sent from 9026 to T01
```

One of the lines returned from the **debug voice switchboard** command reveals how the switchboard and dial plan are processing calls:

**1970.01.04 23:24:59 SWITCHBOARD.StateMachine Call sent from 9026 to T01 (9999)**

In this example, the originating endpoint for the call is extension **9026**. The trunk that accepted the call is **T01**, with an accept number of **9999**. The information contained in this line helps diagnose switchboard or dial plan configuration problems by answering questions such as:

- *Is the call being accepted by the correct trunk (**T01** for this example)?*

If the answer is no, accept numbers have not been configured properly on the trunks.

- *Is the destination number (**9999** for this example) the number the originator actually tried to dial?*

If the answer is no, the dial plan is prematurely matching the number against a template and therefore the dial plan configuration needs to be revised.

If, after examining the results of this line against your own configuration, the digit patterns passed match the numbers that were dialed and calls are being routed to the proper trunks and/or individual users, you can safely conclude that the switchboard and dial plan have been configured correctly for the application.

For information on additional **debug** commands that can assist you with troubleshooting, refer to the *AOS Debug Command Reference Guide* provided on the *AOS Documentation CD*.