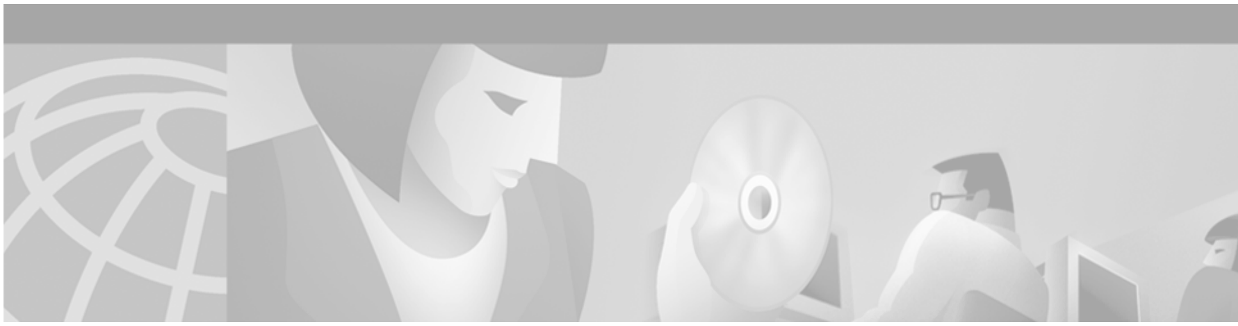


Analog DID for Cisco 2600 and Cisco 3600 Series Routers

This document describes Cisco IOS configuration for Direct Inward Dialing (DID) as supported on Cisco 2600 and Cisco 3600 series modular access routers. This document includes the following sections:

Feature Overview

Direct Inward Dialing (DID) is a service offered by telephone companies that enables callers to dial directly to an extension on a Private Branch Exchange (PBX) without the assistance of an operator or automated call attendant. This service makes use of DID trunks, which forward only the last three to five digits of a phone number to the PBX. If, for example, a company has a PBX with extensions 555-1000 to 555-1999, and a caller dials 555-1234, the local CO would forward 234 to the PBX. The PBX would then ring extension 234. This entire process is transparent to the caller.

When this feature is configured, a voice-enabled Cisco 2600 and Cisco 3600 series router can receive calls from a DID trunk and connect them to the appropriate extensions. The DID state machine is identical to the E&M state machine and uses one of the following signaling types:

Immediate Start

The originating end seizes the line by going off hook and, without waiting for a response, it begins to outpulse digits. The address signaling used with immediate-start signaling consists only of dial-pulsing.

Wink-Start

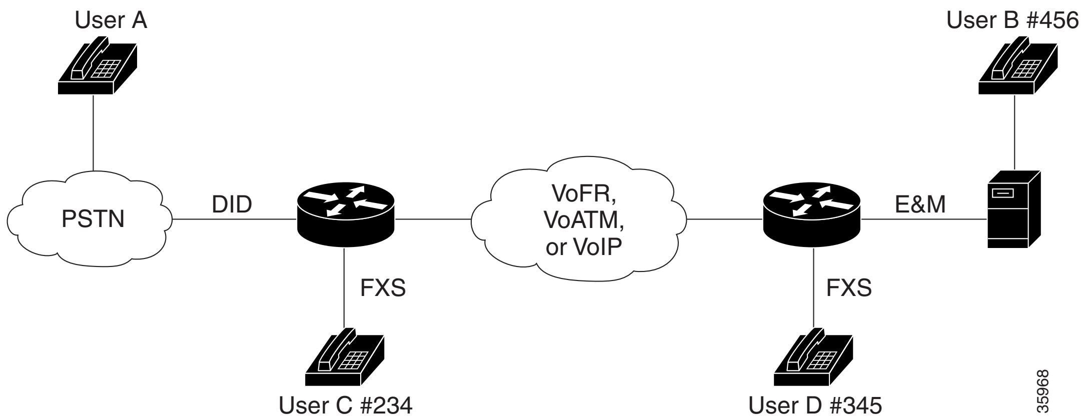
The originating end seizes the line by going off-hook. It waits for acknowledgement from the other end before outpulsing digits. This serves as an integrity check that will identify a malfunctioning trunk and allow the network to send a re-order tone to the calling party.

Delay Dial

The originating end seizes the line and waits 200 ms to see if the far end is on-hook. If so, the originating end then outpulses digits. If the far end is off-hook, the originating end waits until the far end is on-hook before outpulsing digits.

Figure 1 shows a hypothetical topology where a user connected to the PSTN (Caller A) dials various numbers and is connected to the appropriate extensions on a PBX.

Figure 1 DID Support for Cisco 2600 and Cisco 3600 Series Routers



Number dialed by User A	Number received by router	Extension receiving call
555-1234	234	User C
555-1345	345	User D
555-1456	456	User B
555-1678	678	No dial-peer match found; fast busy tone is played

Benefits

The DID feature makes it seem that all extensions on a PBX have direct lines to the PSTN. This is accomplished without the expense associated with connecting each extension to the PSTN. In addition to saving the cost of an operator, DID gives callers the feeling that they are calling an individual, rather than a large company.

Restrictions

Direct Inward Dialing can be configured with the following restrictions:

- Dial tone is not present on DID voice ports.
- Outgoing calls are not allowed on DID voice ports. If an outgoing call is attempted, the caller will get a fast busy signal.

Related Documents

For information on installing and configuring Cisco 2600 and Cisco 3600 routers, refer to the following online documents:

- Cisco 2600 Series Routers
- Cisco 3600 Series Routers

For more information about voice configuration, see the following Cisco IOS Release 12.1 guides:

- *Cisco IOS Multiservice Applications Configuration Guide*
- *Cisco IOS Multiservice Applications Command Reference*

The following configuration guides describe the configuration of IP, Frame Relay, and ATM:

- For more information about configuring IP, see the *Cisco IOS IP and IP Routing Configuration Guide.*
- For more information about configuring Frame Relay, see "Configuring Frame Relay" in the Cisco IOS Release 12.1 *Wide-Area Networking Configuration Guide*.
- For more information about configuring ATM, see "Configuring ATM" in the Cisco IOS Release 12.1 *Wide-Area Networking Configuration Guide*.

The following online feature documentation describes new voice-port features that are available in Cisco IOS Release 12.1(2)T and available with Direct Inward Dialing:

- *Voice Port Enhancements in Cisco 2600 and 3600 Series Routers and MC3810 Series Concentrators*
- *Voice Port Testing Enhancements in Cisco 2600 and 3600 Series Routers and MC3810 Series Concentrators*

Supported Platforms

- Cisco 2610
- Cisco 2611
- Cisco 2612
- Cisco 2613
- Cisco 2620
- Cisco 2621
- Cisco 3620
- Cisco 3640
- Cisco 3661
- Cisco 3662

Supported Standards, MIBs, and RFCs

Standards

- EIT/TIA-464B Requirements for Private Branch Exchange (PBX) Switching Equipment

MIBs

None.

RFCs

None.

Prerequisites

The following hardware, software, and basic configurations are required to support Direct Inward Dialing:

- Install the required Cisco IOS release.
- Obtain DID service from your service provider.
- Establish a working network.
  - For more information about configuring IP, see the *Cisco IOS IP and IP Routing Configuration Guide*.
  - For more information about configuring Frame Relay, see "Configuring Frame Relay" in the *Wide-Area Networking Configuration Guide*.
  - For more information about configuring ATM, see "Configuring ATM" in the *Cisco IOS Wide-Area Networking Configuration Guide*.
- Complete your company's dial plan.
- Establish a working telephony network based on your company's dial plan:
  - For information about setting up a voice network, see *Cisco IOS Multiservice Applications Configuration Guide.*
  - For more information about configuring VoATM on Cisco 3600 series routers, see *Voice over ATM on Cisco 3600 Series Routers.*
  - For more information about configuring VoFR on Cisco 2600 and 3600 series routers, see *Voice over Frame Relay Using FRF.11 and FRF.12 Configuration Updates.*
- Install the VIC-2DID cards. For more information about VIC-2DID cards, see *Update to Cisco WAN Interface Cards Hardware Installation Guide.*
- Install at least one other network module or WAN interface card to provide the connection to the LAN or WAN.

Configuration Tasks

After verifying that your router meets the requirements listed in the "Prerequisites" section on page 5, see the following sections for configuration tasks for DID. Each task in the list is identified as either optional or required:

- Configuring Voice Ports to Support Direct Inward Dialing (Required)
- Verifying Direct Inward Dialing Voice-Port Configuration (Optional)

Configuring Voice Ports to Support Direct Inward Dialing

Follow these steps in order to configure voice ports for Direct Inward Dialing. Not all commands required to configure voice ports appear here. Use the reference information in the "Related Documents" section on page 4 to find out more about voice-port configuration.

Direct Inward Dialing Configuration Steps

	Command	Purpose
Step 1	`Router(config)#` **`configure terminal`**	Enter global configuration mode.
Step 2	`Router(config)#` **`voice-port`** `slot/subunit/port`	Enter voice-port configuration mode on a Cisco 2600 or Cisco 3600 series router.
Step 3	`Router (config-voiceport)#` **`signal did`** `{` **`immediate`** `|` **`wink-start`** `|` **`delay-dial`** `}`	This command enables Direct Inward Dialing on the voice port. • **immediate** if the voice port must use the immediate start protocol • **wink-start** if the voice port must use the wink start protocol • **delay-dial** if the voice port must use the delay start protocol
Step 4	`Router(config-voiceport)#` **`timing wait-wink`** *`milliseconds`*	(Optional for wink-start ports only) This command sets the maximum time to wait for wink signalling after an outgoing seizure is sent.
Step 5	`Router(config-voiceport)#` **`timing wink-wait`** *`milliseconds`*	(Optional for wink-start ports only) This command sets the maximum time to wait before sending wink signal after an incoming seizure is detected.
Step 6	`Router(config-voiceport)#` **`timing wink-duration`** *`milliseconds`*	(Optional for wink-start ports only) This command sets the duration of a wink-start signal.
Step 7	`Router(config-voiceport)#` **`timing delay-duration`** *`milliseconds`*	(Optional for delay-dial ports only) This command sets the duration of the delay signal.
Step 8	`Router(config-voiceport)#` **`timing delay-start`** *`milliseconds`*	(Optional for delay-dial ports only) This command sets the delay interval after an incoming seizure is detected.

Verifying Direct Inward Dialing Voice-Port Configuration

To verify voice-port configuration, enter the **show voice port** command. You can specify a voice port or view the status of all configured voice ports. In the following example, the specified Cisco 2600 FXS port is configured for DID.

```
Router# show voice port 1/0/0

Foreign Exchange Station with Direct Inward Dialing (FXS-DID) 1/0/0 Slot is 1, Sub-unit
is 0, Port is 0
 Type of VoicePort is DID-IN
 Operation State is DORMANT
 Administrative State is UP
 No Interface Down Failure
 Description is not set
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Music On Hold Threshold is Set to -38 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancel Coverage is set to 8 ms
 Playout-delay Mode is set to default
 Playout-delay Nominal is set to 60 ms
 Playout-delay Maximum is set to 200 ms
 Connection Mode is normal
 Connection Number is not set
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Ringing Time Out is set to 180 s
 Companding Type is u-law
 Region Tone is set for US

 Analog Info Follows:
 Currently processing none
 Maintenance Mode Set to None (not in mtc mode)
 Number of signaling protocol errors are 0
 Impedance is set to 600r Ohm
 Wait Release Time Out is 30 s
 Station name None, Station number None

 Voice card specific Info Follows:
 Signal Type is wink-start
 Dial Type is dtmf
 In Seizure is inactive
 Out Seizure is inactive
 Digit Duration Timing is set to 100 ms
 InterDigit Duration Timing is set to 100 ms
 Pulse Rate Timing is set to 10 pulses/second
 InterDigit Pulse Duration Timing is set to 750 ms
 Clear Wait Duration Timing is set to 400 ms
 Wink Wait Duration Timing is set to 200 ms
 Wait Wink Duration Timing is set to 550 ms
 Wink Duration Timing is set to 200 ms
 Delay Start Timing is set to 300 ms
 Delay Duration Timing is set to 2000 ms
 Dial Pulse Min. Delay is set to 140 ms
 Percent Break of Pulse is 60 percent
 Auto Cut-through is disabled
 Dialout Delay for immediate start is 300 ms
```

Command Reference

This section documents new or modified commands. All other commands used with this feature are documented in the Cisco IOS Release 12.1 command reference publications and in Cisco IOS Release 12.1 feature modules.

- **signal did**

signal did

To enable Direct Inward Dialing on a voice port, use the **signal did** voice-port configuration command. To disable the command's effect, use the **no** form of this command.

signal did {**immediate-start** | **wink-start** | **delay-start**}

no signal did

Syntax Description

immediate-start	Use this setting if the DID voice port must use immediate-start signaling.
wink-start	Use this setting if the DID voice port must use wink-start signaling.
delay-start	Use this setting if the DID voice port must use delay-dial signaling.

Defaults

The default value is immediate-start

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(5)XM	This command was introduced for Cisco 2600 and Cisco 3600 series routers.

Usage Guidelines

The **signal did** command enables a voice port to receive calls from a DID trunk.

This command applies to Cisco 2600 and Cisco 3600 series routers.

Examples

The following example configures a voice port on a Cisco 2600 or 3600 router where DID information is received:

```
voice-port 1/0/1
  signal did immediate-start
```

Glossary

ATM—Asynchronous Transfer Mode. International standard for cell relay in which multiple service types (such as voice, video, or data) are conveyed in fixed-length (53-byte) cells. Fixed-length cells allow cell processing to occur in hardware, thereby reducing transit delays. ATM is designed to take advantage of high-speed transmission media such as E3, SONET, and T3.

CAS—channel-associated signaling. Trunk signaling (for example, in a T1 line) in which control signals, such as those for synchronizing and bounding frames, are carried in the same channel along with voice and data signals.

CCS—common channel signaling. Trunk signaling (for example, using Primary Rate Interface) in which a control channel carries signaling for separate voice and data channels.

CO—central office. Local telephone company office to which all local loops in a given area connect and in which circuit switching of subscriber lines occurs.

codec—Coder-decoder. Device that typically uses pulse code modulation to transform analog signals into a digital bit stream and digital signals back into analog.

CTI—Computer telephony integration.

DTMF—Dual-tone multifrequency. Use of two simultaneous voice-band tones for dialing (such as touch tone).

DSP—digital signal processor.

DID—Direct Inward Dialing.

E&M—rEceive and transMit, or Ear and Mouth. Type of signaling originally developed for analog two-state voltage telephony using the ear and mouth leads; in digital telephony, uses two bits.

ETSI—European Telecommunication Standards Institute.

FXO—Foreign Exchange Office. A voice interface emulating a PBX trunk line to a switch or telephone equipment to a PBX extension interface.

FXS—Foreign Exchange Station. A voice interface for connecting telephone equipment, emulates the extension interface of a PBX or the subscriber interface for a switch.

IETF—Internet Engineering Task Force

ISDN—Integrated Services Digital Network. Communication protocol, offered by telephone companies, that permits telephone networks to carry data, voice, and other source traffic.

POTS—plain old telephone service

PSTN—Public Switched Telephone Network. General term referring to the variety of telephone networks and services in place worldwide.

SNMP—Simple Network Management Protocol. Network management protocol used almost exclusively in TCP/IP networks. SNMP provides a means to monitor and control network devices, and to manage configurations, statistics collection, performance, and security.

T1—Digital WAN carrier facility. T1 transmits DS 1-formatted data at 1.544 Mbps through the telephone switching network, using alternate mark inversion or B8ZS coding.

T1 trunk—Digital WAN carrier facility. See T1.

TDM—time-division multiplexing

Trunk—Physical and logical connection between two switches across which network traffic travels. A backbone is composed of a number of trunks.