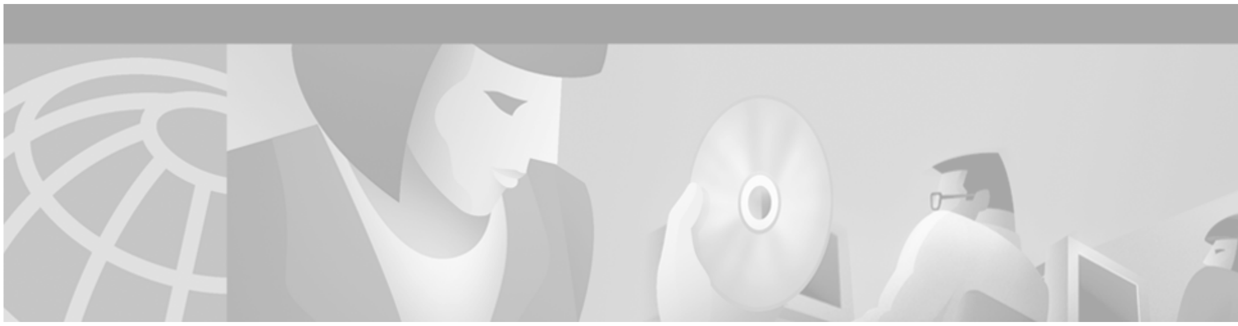

Caller ID on Cisco 2600 and 3600 Series Routers and Cisco MC3810 Multiservice Concentrators

This appendix describes Cisco IOS configuration for caller ID as supported on the Cisco MC3810 multiservice concentrator and on Cisco 2600 and 3600 series routers. It includes the following sections:

To identify the hardware platform or software image information associated with a feature in this appendix, use the Feature Navigator on Cisco.com to search for information about the feature or refer to the software release notes for a specific release. For more information, see the “Identifying Supported Platforms” section in the “Using Cisco IOS Software” chapter.

Caller ID Overview

Caller ID (sometimes called *CLID* or *ICLID* for incoming call line identification) is an analog service offered by a central office (CO), which supplies calling party information to subscribers. Typically, the calling party number, and sometimes the name, appears on a station (also called *extension*) device such as a PC telephony software application screen or the display on a telephone. Type 1 caller ID provides the calling party information while the call is ringing, and Type 2 caller ID provides the additional convenience of calling number display while the recipient is on another call. In this release, Cisco provides only Type 1 caller ID support.

The caller ID feature supports the sending of calling party information from foreign exchange station (FXS) loop-start and ground-start ports into a caller ID equipped telephone device. The FXS port emulates the extension interface of a private-branch exchange (PBX) or the subscriber interface for a CO switch.

The caller ID feature supports receiving calling-party information at foreign exchange office (FXO) loop-start and ground-start ports. The FXO port emulates a connection to a telephone and allows connection to a PBX extension interface or (where regulations permit) a CO subscriber line.

The following are benefits of using caller ID:

- Enterprises—Caller ID is invaluable for increasing efficiency through its use in computer telephony integration (CTI) applications, where for example, calling party information can be used to retrieve client information from a database when a customer call is received.

- Service Provider—In traditional telephony, caller ID is a standard service that service provider customers expect. With the Cisco support for caller ID, service providers can offer the feature for packet-switched Voice over IP (VoIP), Voice over Frame Relay (VoFR), and Voice over ATM (VoATM) services.

Calling Name and Number

Figure 130 shows a hypothetical topology where users, indicated by telephone icons, receive different types of caller-ID support depending upon whether the caller-ID information from the caller passes through an FXO or FXS port before reaching the party who receives the call.

Figure 130 Caller ID and ANI Support

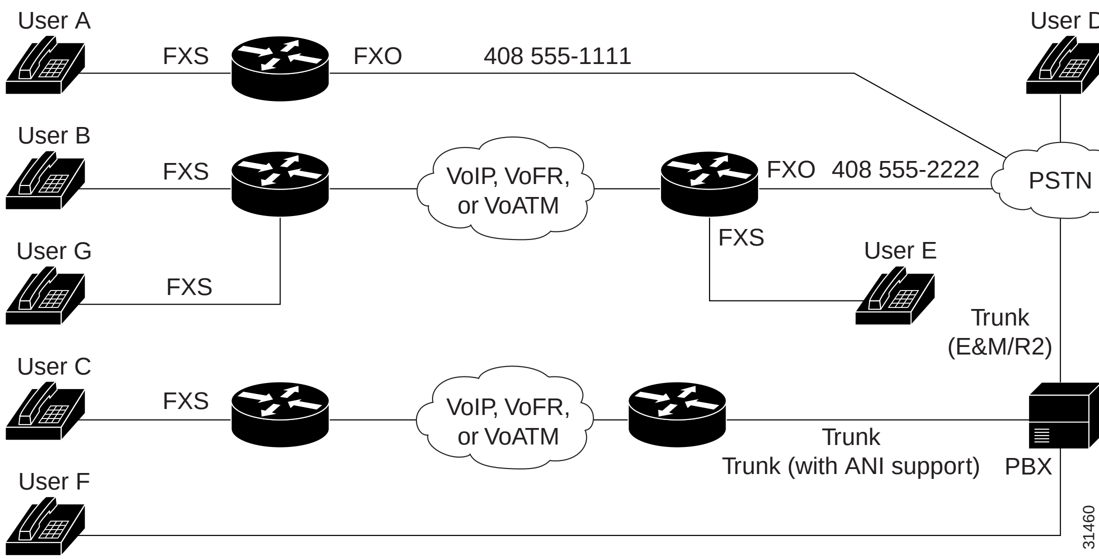

Table 56 shows how caller-ID information is received by the users in Figure 130 as follows:

- When an outbound caller-ID call is carried by a trunk with E&M or R2 signaling, the recipient sees only the ANI calling number of the caller.
- When caller-ID information is sent from an originating FXS station by way of the PSTN, the recipient sees only the identification of the FXO port through which the call is routed.
- When caller-ID information is sent from an originating station over a trunk with E&M or R2 signaling and through the PSTN, the recipient sees only the trunk identification because the ANI information is not preserved by the PSTN.

Table 56 Caller-ID Information Received

User Originating Call	User Receiving Call	Caller-ID Information Received
A	D	D receives the caller ID of the PSTN subscriber line only (408 555-1111).
D	A	A receives the calling number and name of D, provided that the PSTN subscriber line (408 555-1111) is enabled for Caller ID.
D	B	B receives the calling number and name of D, provided that the PSTN subscriber line (408 555-1111) is enabled for Caller ID.
B	D	D receives the caller ID of the PSTN subscriber line only (408 555-2222).
B	E	E receives the Calling Number and Name string of B.
B	G	G receives the Calling Number and Name string of B.
E	B	B receives the Calling Number and Name string of E.
F	C	C receives the Calling Number of F.
C	F	Calling Number of C.
D	C	C Receives Calling Number of D through ANI.
C	D	Calling Number of C goes through ANI to the PSTN. However, the PSTN displays only the trunk ID, so D sees only this information.
C	F	The information that F receives depends on the PBX features available.

Call Time Display

When caller-ID information is sent, the local time set on the router is transmitted with the station name and number. If a call received on an FXO port is terminated on an FXS port, the calling time received on the FXO port is replaced by the local time while transmitting caller ID to the FXS port. This is also true for calls received from the network. The router should be configured to retrieve network time at boot up from an NTP server in order to maintain the correct local time setting.

For more information about voice configuration, refer to the following:

- *Cisco IOS IP Routing Configuration Guide*
- *Cisco IOS Wide-Area Networking Configuration Guide*

The following online feature documentation and installation guides describe the configuration and installation of hardware components:

- For information about installing Cisco MC3810 multiservice concentrators, see *Cisco MC3810 Multiservice Concentrator Hardware Installation* at the following location: http://www.cisco.com/univercd/cc/td/doc/product/access/multicon/3810hwig/index.htm
- For information about installing Cisco 2600 series routers, see the documents listed at the following location: http://www.cisco.com/univercd/cc/td/doc/product/access/acs_mod/cis2600/index.htm
- For information about installing Cisco 3600 series routers, see the documents listed at the following location: http://www.cisco.com/univercd/cc/td/doc/product/access/acs_mod/cis3600/index.htm

Caller ID Prerequisites Tasks

The following hardware, software, and basic configurations are required to support caller ID:

- Cisco IOS Release 12.1(3)T software.
- Caller ID service from your service provider.
- A working network. For more information, see the following publications:
  - Cisco IOS *Cisco IOS IP Routing Configuration Guide*
  - Cisco IOS *Wide-Area Networking Configuration Guide*
- Your company's dial plan.
- A working telephony network based on your company's dial plan:
- If applicable to your network, install a 2-channel analog plain old telephone service (POTS) FXS voice interface card (VIC) in a Cisco 2600 series chassis slot or Cisco 2600 or 3600 network module.
- If applicable to your network, install one of the following Cisco MC3810 multiservice concentrator FXO network modules:
  - MC3810-APM-FXO (generic); caller ID is supported in versions v04.xx and later of this APM.
  - MC3810-FXO-PR2 (Pacific Rim 2); caller ID is supported in versions v02.xx and later of this APM.
  - MC3810-FXO-PR3 (Pacific Rim 3); caller ID is supported in versions v02.xx and later of this APM.
  - MC3810-FXO-UK (UK); caller ID is supported in versions v03.xx and later of this APM.
  - MC3810-FXO-GER (Germany); caller ID is supported in versions v03.xx and later of this APM.
- For a Cisco MC3810 multiservice concentrator, install an HCM as follows:
  - An HCM2 to supply 4 or 8 voice or fax channels at high or medium codec complexity.
  - An HCM6 to supply 12 or 24 voice or fax channels at high or medium codec complexity.
- For information about installing Cisco MC3810 multiservice concentrator HCMs, refer to *Cisco MC3810 Multiservice Concentrator Hardware Installation* at the following url: http://www.cisco.com/univercd/cc/td/doc/product/access/multicon/3810hwig/index.htm

  Note The Cisco MC3810 multiservice concentrator voice-compression module does not support caller ID. Install an HCM instead.

- One other network module or WAN interface card to provide the connection to the LAN or WAN.

Note Specific hardware is required to provide full support for the Caller ID features. To determine support for these features in your configuration, review the appropriate hardware documentation and data sheets. This information is available on Cisco.com.

Caller ID Configuration Task List

Voice-port configuration is the only special configuration required to support caller ID. To configure your voice network fully, use the reference information in the section "Caller ID Prerequisites Tasks" section on page 814 to perform the following tasks on your routers:

- Configure your IP, ATM, or Frame Relay network to support real-time voice traffic.
- Configure voice cards for codec settings.
- Configure voice dial peers. Each dial peer defines the characteristics associated with a call leg.

The remainder of this section describes the steps required to configure caller ID on FXS and FXO voice ports.

Configuring Voice Ports to Support Caller ID

To configure voice ports to support caller ID, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)# voice-port slot/port` or	Enters voice-port configuration mode on a Cisco MC3810 multiservice concentrator. The *slot* number for analog voice ports on the Cisco MC3810 multiservice concentrator is always 1. There is no port 0 for voice ports.
	`Router(config)# voice-port slot-number/subunit-number/port`	Enters voice-port configuration mode on a Cisco 2600 or 3600 series router.
Step 2	`Router(config-voiceport)# connection {plar | tie-line | plar-opx} digits {trunk digits [answer-mode]}`	Specifies the voice-port connection type and the destination telephone number. The keywords and arguments are as follows: • **plar**—Specifies private line automatic ringdown. • **tie-line**—Specifies a tie-line connection to a PBX. • **plar-opx**—Specifies a PLAR off-premises extension (the local voice port provides a local response before the remote voice port receives an answer). • **trunk**—Specifies a straight tie-line connection to a PBX. • **answer-mode**—Indicates whether a trunk connection is specified. The router should not attempt to initiate a trunk connection, but should wait for an incoming call before establishing the trunk. • *digits*—Specifies the destination telephone number.

	Command	Purpose
Step 3	`Router(config-voiceport)#` **`voice confirmation-tone`**	Enables the two-beep confirmation tone that a caller hears when picking up the handset, if **connection plar** or **connection plar-opx** is configured.
Step 4	`Router(config-voiceport)#` **`dial-type`** `{`**`dtmf`** `|` **`pulse`**`}`	(For FXO ports only) Selects the appropriate dial type for out-dialing.
Step 5	`Router(config-voiceport)#` **`signal`** `{`**`loop-start`** `|` **`ground-start`**`}`	Selects the appropriate signal type for this interface.
Step 6	`Router(config-voiceport)#` **`codec`** `{`**`g729r8`** `|` **`g729ar8`** `|` **`g726r32`** `|` **`g711alaw`** `|` **`g711ulaw`**`}`	(Cisco MC3810 multiservice concentrator only) Configures the voice-port compression mode. The **g729ar8** value is the default and is recommended. **Note** On Cisco 2600 and 3600 series routers, codec configuration is performed on dial peers. On all supported routers, **codec** command options may vary depending upon the voice card settings.
Step 7	`Router(config-voiceport)#` **`cptone`** *`locale`*	Selects the appropriate voice call progress tone for this interface. Caller ID requires this setting. The caller ID standard (Bellcore/Telcordia, ETSI, ETSI-DTMF) is determined by this command. On the Cisco MC3810 multiservice concentrator, the default setting for *locale* is northamerica. On Cisco 2600 and 3600 series routers, the default setting for *locale* is us. See Table 57 on page 816 for a list of options.
Step 8	`Router(config-voiceport)#` **`ring frequency`** `{`**`25`** `|` **`50`**`}`	(Required on Cisco 2600 and 3600 series routers FXS ports only) Selects the appropriate ring frequency (in hertz) specific to the equipment attached to this voice port.
Step 9	`Router(config-voiceport)#` **`caller-id attenuation`** `attenuation`	(Optional on FXO ports only) Specifies an attenuation other than the default of 14 dB (minus 14 dBm), enter a value of from 0 to 64, in decibels.
Step 10	`Router(config-voiceport)#` **`ring number`** *`number`*	(Required on Cisco 2600 and 3600 series routers FXO ports only) Specifies the maximum number of rings to be detected before answering a call.

The following table lists the options that may be used for the *locale* variable with the **cptone** command.

Table 57 cptone Command Entries for the Cisco 2600 and 3600 Series

Command Option	Country	Command Option	Country
ar	Argentina	lu	Luxembourg
au	Australia	my	Malaysia
at	Austria	mx	Mexico
be	Belgium	nl	Netherlands

Table 57 cptone Command Entries for the Cisco 2600 and 3600 Series (continued)

Command Option	Country	Command Option	Country
br	Brazil	nz	New Zealand
ca	Canada	no	Norway
cn	China	pe	Peru
co	Colombia	ph	Philippines
cz	Czech Republic	pl	Poland
dk	Denmark	pt	Portugal
fi	Finland	ru	Russian Federation
fr	France	sg	Singapore
de	Germany	sk	Slovakia
gr	Greece	si	Slovenia
hk	Hong Kong	za	South Africa
hu	Hungary	es	Spain
is	Iceland	se	Sweden
in	India	ch	Switzerland
id	Indonesia	tw	Taiwan
ie	Ireland	th	Thailand
il	Israel	tr	Turkey
it	Italy	gb	Great Britain
jp	Japan	us	United States
kr	Korea Republic	ve	Venezuela

Configuring FXS and FXO Voice Ports to Support Caller ID

To configure caller-ID on FXS and FXO voice ports, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)# caller-id enable`	Enables caller ID. This command applies to FXS voice ports that send caller-ID information and to FXO ports that receive it. By default caller ID is disabled. **Note** If the **station-id** or a **caller-id alerting** command is configured on the voice port, these automatically enable caller ID, and the **caller-id enable** command is not necessary.
Step 2	`Router(config-voiceport)# station-id name name`	Configures the station name on FXS voice ports connected to user telephone sets. This sets the caller-ID information for on-net calls originated by the FXS port. You can also configure the station name on an FXO port of a router for which incoming Caller ID from the PSTN subscriber line is expected. In this case, if no caller-ID information is included on the incoming PSTN call, the call recipient receives the information configured on the FXO port instead. If the PSTN subscriber line does provide caller-ID information, this information is used and the configured station name is ignored. The *name* argument is a character string of 1 to 15 characters identifying the station. **Note** This command applies only to caller-ID calls, not Automatic Number Identification (ANI) calls. ANI supplies calling number identification only.

	Command	Purpose
Step 3	`Router(config-voiceport)#` **`station-id number`** *`number`*	Configure the station number on FXS voice ports connected to user telephone sets. This sets the caller-ID information for on-net calls originated by the FXS port. You can also configure the station number on an FXO port of a router for which incoming caller ID from the PSTN subscriber line is expected. In this case, if no caller-ID information is included on the incoming PSTN call, the call recipient receives the information configured on the FXO port instead. If the PSTN subscriber line does provide caller-ID information, this information is used and the configured station name is ignored. If the caller-ID station number is not provided by either the incoming PSTN caller ID or by the station number configuration, the calling number included with the on-net routed call is determined by Cisco IOS software by using a reverse dial-peer search. In this case, the number is obtained by searching for a POTS dial-peer that refers to the voice-port and the destination-pattern number from that dial-peer is used. *number* is a string of 1 to 15 characters identifying the station telephone or extension number.
Step 4	`Router(config-voiceport)#` **`caller-id block`**	(FXS ports only) When this command is configured at the originating end of a call, it requests that the originating calling party information not be displayed at the called party's telephone. **Note** The calling party information is included in the routed on-net call, as this is often required for other purposes, such as billing and call blocking. The request to block display of the calling party information on terminating FXS ports will normally be accepted by Cisco routers, but no guarantee can be made regarding the treatment by other equipment. This command affects all calls sent to an FXO station from the configured FXS station. The central office (CO) may supply a feature code that a user can dial in order to block caller-ID transmission on a call-by-call basis. When a blocked-information call passes through an FXO interface on the way to its destination, the blocking is passed on to the receiving party.

To configure the alerting method, use the following commands beginning in global configuration mode. Configuration of the alerting method is required when the caller ID standard, specified by locale through the **cptone** command, is other than Bellcore/Telcordia (if you do not configure the alerting method, the default c**aller-id alerting ring 1** command is applied). The command that you enter is determined by the Bellcore/Telcordia or ETSI standard that your service provider uses for caller ID. For more information about standards, see the Caller ID Prerequisites Tasks, page 814 section.

	Command	Purpose
Step 1	Router(config)# **voice-port** *slot*/*port*	Enters voice-port configuration mode on a Cisco MC3810 multiservice concentrator. The slot number for analog voice ports on the Cisco MC3810 multiservice concentrator is always 1. There is no port 0 for voice ports.
	or	
	Router(config)# **voice-port** *slot-number*/*subunit-number*/*port*	Enters voice-port configuration mode on a Cisco 2600 or 3600 series router.
Step 2	Router(config-voiceport)# **caller-id alerting ring** {**1** | **2**}	Configure this command on FXO ports where caller ID information is received from a subscriber telephone line. and on FXS voice ports from which caller ID information is transmitted to an attached telephone device. Compatible settings are required on both ends of the telephone line connection or caller ID information may not be displayed. Enter **1** if your telephone line service provider or telephone device specifies it, to provide or expect caller ID information following the first ring at the receiving station. This is the default setting. Enter **2** to provide or expect caller-ID information during the long ring pause following two short rings. This setting is used in Australia and the United Kingdom.
Step 3	Router(config-voiceport)# **caller-id alerting line-reversal**	(FXS ports only) Configure this setting only when the attached telephone device requires line polarity reversal to signal the start of caller-ID information transmission.

	Command	Purpose
Step 4	Router(config-voiceport)# **caller-id alerting dsp-pre-alloc**	(FXO ports, only when caller-ID alerting line-reversal is required) Configure this command on the FXO port when the incoming subscriber telephone line uses line polarity reversal to signal the start of caller-ID information transmission. The Cisco FXO interface cannot detect line-reversal alerting in the on-hook state. For this reason, DSPs must be pre-allocated to serve the Type 1 caller ID information when it arrives. Preallocating the DSPs enables the DSP to continuously monitor for the arrival of caller-ID information.
Step 5	Router(config-voiceport)# **caller-id alerting pre-ring**	(FXS ports only) Configure this setting only when the attached telephone device requires the pre-ring (immediate ring) method to signal the start of a caller ID information. The command activates a 250-ms pre-ring.

Verifying Caller ID on Voice Ports 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 MC3810 multiservice concentrator FXS port is configured with a Bellcore/Telcordia standard (**cptone** value is **northamerica**), a station name, and a station number. The **caller-id alerting ring** setting is 1.

```
Router> show voice port 1/1
 FXS 1/1 Slot is 1, Port is 1
 Type of VoicePort is FXS
 Operation State is UP
 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
 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
 Coder Type is g729ar8
 Voice Activity Detection is disabled
 Nominal Playout Delay is 80 milliseconds
 Maximum Playout Delay is 160 milliseconds
 Region Tone is set for US

 Analog Info Follows:
 Currently processing Voice
 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
 Analog interface A-D gain offset = -3.0 dB
 Analog interface D-A gain offset = -3.0 dB
 FXS idle voltage set to low

 Caller ID Info Follows:
 Standard BELLCORE
 Station name A. Person, Station number 4085551111
 Caller ID presentation unblocked
 Output attenuation is set to 14 dB
 Caller ID is transmitted after 1 rings

 Voice card specific Info Follows:
 Signal Type is loopStart
 Ring Frequency is 20 Hz
 Hook Status is Off Hook
 Ring Active Status is inactive
 Ring Ground Status is inactive
 Tip Ground Status is active
 Digit Duration Timing is set to 100 ms
 InterDigit Duration Timing is set to 100 ms
 Ring Cadence is defined by CPTone Selection
 Ring Cadence are [20 40] * 100 msec
 InterDigit Pulse Duration Timing is set to 500 ms
```

Troubleshooting Tips

If you have caller-ID problems on telephones connected to FXS ports, the following tips may be helpful:

- Try a different brand of phone to confirm that the problem is not caused by a malfunctioning or incompatible caller ID telephone.
- Ensure that the **cptone** command is set correctly to reflect your locale.
- If the call time display is incorrect, check the router clock setting. An NTP network time server is recommended for accurate display of the local time.
- If expected information is not displayed, use the **show call history** command to make sure that the information that the router received during the call setup is complete.
- The line voltage available on FXS voice ports of the Cisco MC3810 multiservice concentrator and Cisco 2600 and 3600 series routers is –24V. Some phones, particularly those manufactured by Bell South, do not recognize –24V caller-ID signaling. On a Cisco MC3810 multiservice concentrator, use the **idle-voltage high** voice-port configuration command to boost the voltage on an FXS port.

If you have caller-ID display problems on FXO ports, the following tips may be helpful:

- Disconnect the router from the phone line and attach a caller-ID equipped telephone to verify that the CO is sending caller-ID information:
  - Listen and watch to see when the caller-ID information is displayed: before the first ring, after the first ring, or after the second ring?
  - Make sure that the router configuration matches the timing of the display. If the phone is answered during the first ring, does this cause the phone not to display the caller-ID information? If so, the CO may be sending the caller-ID information after the first ring, requiring a change to a caller-ID alerting setting. Make sure the router is not configured to answer the call on the FXO before the Caller ID-information is received. If needed, increase the number of rings required before answering.
- Use the **show call history** command to check the information received by the caller ID receiver.

The following **debug** commands may be useful for analyzing problems:

- **debug vpm signal**
- **debug vtsp dsp**
- **debug vtsp session**

Note Specific hardware is required to provide full support for the Caller ID features. To determine support for these features in your configuration, review the appropriate hardware documentation and data sheets. This information is available on Cisco.com.