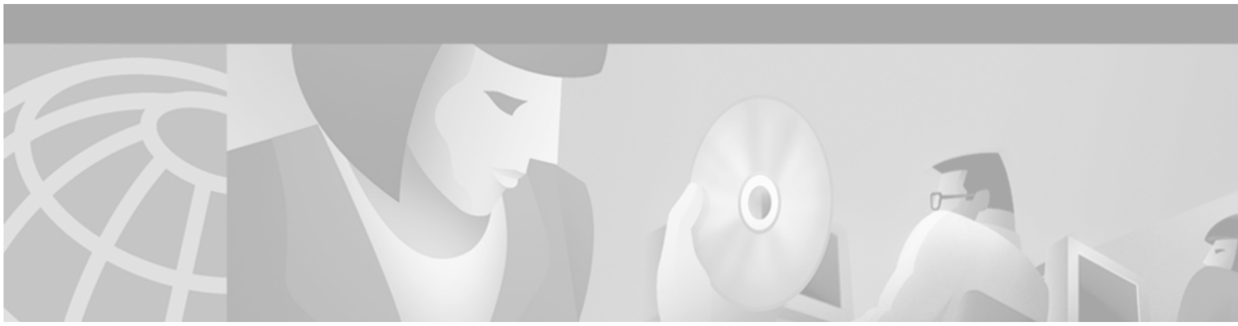

Enhanced Voice Services for Japan for Cisco 800 Series Routers

This feature module describes the Enhanced Voice Services for Japan Cisco IOS features, including INS-NET-64 voice features. It describes the benefits of the new features, supported platforms, configuration, related documents, and provides command-reference information.

This document includes the following sections:

Feature Overview

The Enhanced Voice Services for Japan Cisco IOS features consist of the following voice capabilities for the Cisco 800 series routers:

- Caller ID

  Provides analog Caller ID support for Japanese-language display, Caller ID- equipped, analog telephones. The Cisco 800 series router receives the Caller ID information from the INS-NET-64 switch. The router software prepares the Caller ID with a tone, transmits the Caller ID to plain old telephone service (POTS) port 1 or 2 on the router, and displays the Caller ID on the telephone.

- Call Blocking on Caller ID

  Allows Cisco 800 series routers to reject an incoming voice call, based on local directory number (LDN) Caller IDs. Using the command-line interface (CLI), you can configure blocking for up to ten Caller ID numbers for each LDN.

- Local Call Waiting

  Notifies you with a call-waiting tone of an incoming call while you are already connected to a telephone call. You can place the first call on hold by pressing the on-and-off-hook button (flash), connect to the second call, and then return to the first call after finishing with the second.

The feature uses both B channels of the ISDN line, enabling local call-waiting support on the router. Unlike standard ISDN call waiting, local call waiting does not require a subscription to call waiting from a service provider.

- E Ya Yo

  Conceals the caller ID of the outgoing call from the receiving device. To activate the feature, dial 184 before dialing the number of the receiving device, as specified in the Nippon Telegraph and Telephone (NTT) Communications Corporation user manual. This feature is specific to NTT Communications Corporation switches and is offered free of charge. The router handles this feature as a regular outgoing call and requires no special operation.

- Voice Warp

  On the INS-NET-64 switch, forwards all incoming calls for a terminal device to another device. Voice-warp registration, activation, and deactivation requests are sent to the switch for each LDN. The routers support the registration, activation, and deactivation requests for devices attached to the PHONE 1 or 2 port. The forwarding function itself is performed by the INS-NET-64 switch. This feature can be deactivated after its registration and activation phases.

  During the registration phase of the device, you can:

  - Create a list of forwarding destination numbers and select one as the active destination.
  - Specify whether an announcement is made to the caller, to the forwarding device, or both, when the call is forwarded.
  - Set the no-answer timer parameter from 5 to 60 seconds at 5-second intervals. This setting affects the redirection of calls when the voice-warp feature is activated.

  During the activation phase, you determine whether calls are redirected all the time or only if the receiving device is busy or does not answer within the specified no-answer time period. You can use the telephone keypad dialing sequence as specified in the NTT user manual for any of the operations described above and to hear the Voice Warp registration details for a local device.

- Voice Select Warp

  An enhanced version of the Voice Warp feature. You can create a list of incoming caller IDs that is used in call redirection, either by redirecting incoming calls only from matching caller IDs, or by redirecting all calls except those from matching caller IDs. You can use the telephone keypad dialing sequence as specified in the NTT user manual for any of the Voice Select Warp feature operations and to hear the Voice Warp registration details for a local device.

- Nariwake

  Checks for Caller IDs that you register (using the telephone keypad for each LDN) and presents a distinctive ring to the telephone port receiving the incoming call if a match is detected. The routers provide three different ring cadences that you can set for calls from both registered and unregistered callers.

  The default ring cadence setting is ring 1 for registered callers and ring 0 for unregistered callers. The on-and-off period for normal ringing signals (ring 0) and ringing signals for Nariwake service (ring 1) are defined in the NTT user manual.

  The number of Caller IDs you can register for each LDN at one time is defined by the INS-NET-64 switch, not by the router. You can register this feature with the list of caller IDs for each LDN, cancel the registration for the LDN, or get registration information from the INS-NET-64 switch. You can use the telephone keypad dialing sequence as specified in the NTT Communications Corporation user manual for any of the Nariwake feature operations and to hear the Nariwake registration details for a local device.

- Trouble Call Blocking

  Also described as *nuisance telephone call refusal service* by INS Net. The network rejects all incoming calls to a particular telephone number from a troublesome caller. You do not have to specify the actual telephone number of the caller.

  When activated, the caller hears a standard telephone announcement and a disconnect message. For information about the announcement or message, see the NTT user manual.

  You are not automatically notified of incoming call attempts. However, to confirm call blocking results, you can listen to an announcement listing the number of incoming calls from blocked telephone numbers during the previous two months.

  The number of callers that you can block is defined by the service provider at the time the service is activated. If you request an additional telephone number to block beyond the defined limit, the oldest number is discarded (unblocked) before the new number is registered.

  To add a new number, you must hang up the telephone, go off-the-hook, and dial the call-blocking telephone keypad sequence within 60 seconds. When the feature is activated, you receive a recorded announcement indicating whether or not the activation is successful.

  The feature can be turned off for either the last added (blocked) number or for all call-blocked numbers. A recorded announcement indicates the changes after they are made.

- I Number

  Supports the use of multiple terminal devices with one subscriber line. The telephone numbers of the subscriber line and router ports are assigned by the service provider. Calls coming into any of the assigned numbers are routed through the same subscriber line to the terminal device attached to the target port.

- POTS Dial

  Supports the POTS dial feature for Japanese telephones. Using a dial application on your workstation, you can dial a telephone number for the POTS port on the router.

  If the telephone is on the hook when you issue the dial command, the router rings the telephone, waits until the telephone is taken off the hook, and then dials the requested number. If the telephone is off the hook when you issue the command, the router dials the requested number.

- POTS Disconnect

  Disconnects a telephone number from the POTS port on the router.

Benefits

- Caller ID—Provide analog Caller ID support for Japanese Caller ID-equipped telephones.
- Call Blocking on Caller ID—Reject incoming voice calls based on LDN Caller IDs.
- Local Call Waiting—Provide call waiting on a local basis for Cisco 800 series routers.
- E Ya Yo—Prevent the Caller ID of an outgoing call from being visible to a receiving device.
- Voice Warp—On a switch, forward registered incoming calls for a terminal device to another terminal device. List more than one forwarding destination number in the switch register and then select one to be the active number. Specify an announcement to be heard on the caller side, the forwarding side, or both when a call is forwarded.
- Voice Select Warp—Create a registration list of Caller IDs, and use it to redirect incoming calls. Choose to ignore the registration list, which causes functionality to be the same as Voice Warp.

- Nariwake—Provide distinctive ring cadences for registered Caller IDs to telephone ports receiving incoming calls.
- Trouble Call Blocking—Refuse nuisance telephone calls.
- I Number—Use one subscriber line for multiple terminal devices.
- POTS Dial—Dial a telephone number on a Cisco 800 series router POTS port by using a dial application on your workstation.
- POTS Disconnect—Disconnect a telephone number from a Cisco 800 series router POTS port.

Restrictions

You must subscribe to the NTT services to use the Enhanced Voice Services for Japan Cisco IOS features. Therefore, except for the Call Blocking on Caller ID feature and Local Call Waiting, support is limited to telephone service inside Japan.

The following limitations also apply:

- Caller ID
  - You must subscribe to Caller ID service before using this feature.
  - In Japan, the analog Caller ID feature supports only Japanese Caller ID telephones.
- Call Blocking on Caller ID
  - You must subscribe to Caller ID service before using this feature.
  - This function is not enabled during setup; it is only enabled if you enter Caller ID numbers for blocking through the CLI.
  - The routers store a maximum of ten Caller ID telephone numbers to block. Cisco 800 series routers do not accept additional Caller ID numbers if ten numbers already exist. In this case, you must remove a number before adding another Caller ID number for blocking.
- Local Call Waiting
  - This feature is not supported if any of the interactive voice response (IVR) features (such as voice warp, voice select warp, and Nariwake) are in use.
  - The call waiting feature is provided locally; therefore each call must have its own separate B channel. Local Call Waiting is not available if data traffic is already on-going or if both B channels are in use, for example, if POTS 1 and POTS 2 are already connected.
  - If an ISDN line already supports Call Waiting before Local Call Waiting is configured on a Cisco 811 or 813 router, the router activates ISDN Call Waiting instead of Local Call Waiting.
- Voice Warp
  - You must subscribe to the Voice Warp service before using this feature.
  - Activating the Voice Warp feature disables support for the Call Waiting feature for both local and network calls.
  - Status information for this feature is delivered over voice only.
  - The routers support this feature for one only LDN. If more than one LDN is configured, only the primary LDN can be used with this feature.
- Voice Select Warp—All Voice Warp limitations apply to the Voice Select Warp feature.

- Nariwake
  - You must subscribe to Nariwake service before using this feature.
  - Activating the Nariwake feature disables support for the Call Waiting feature for both local and network calls.
  - The Cisco 800 series routers support this feature for one LDN only. If more than one LDN is configured, only the primary LDN can be used with this feature.
- Trouble Call Blocking
  - The maximum number of troublesome callers you can block is defined when the service is activated. If you request to block more than the maximum number, the oldest blocked number must be unblocked before the new telephone number can be registered.
  - When multiple NTT services are provided with the troublesome call refusing feature, the features could possibly limit or interact with each other.
  - The Cisco 800 series routers support this feature for one LDN only. If more than one LDN is configured, only the primary LDN can be used with this feature.
- I Number—You must subscribe to the I Number service before using this feature.

Related Documents

Release Notes and Caveats

- *Cross-Platform Release Notes for Cisco IOS Release 12.0*
- *Caveats for Cisco IOS Release 12.0 T*

Cisco 811 and 813 Routers

- *Cisco 811 and Cisco 813 Routers Hardware Installation Guide*
- *Quick Start Guide: Setting Up Cisco 811 and Cisco 813 Routers*

Cisco 800 Series Routers

- *Cisco 800 Series Router Quick Start Guide*
- *Cisco 800 Series Routers Hardware Installation Guide*
- *Cisco 800 Series Routers Software Configuration Guide*

Cisco 805 Routers

- *Cisco 805 Router Hardware Installation Guide*
- *Quick Start Guide – Setting Up the Cisco 805 Router*
- *Cisco 805 Router Software Configuration Guide*

Supported Platforms

- Cisco 811
- Cisco 813

Prerequisites

Before using the Enhanced Voice Services for Japan Cisco IOS features, use the Cisco IOS command **pots country** ***jp*** to configure the router telephone ports to Japanese standards. The following requirements must also be met:

- E Ya Yo—You must subscribe to the NTT Communications Corporation E Ya Yo feature and connect the router to a Japanese INS-NET-64 switch.
- Voice Warp—You must subscribe to the NTT Communications Corporation Voice Warp and Caller ID services and connect the router to a Japanese INS-NET-64 switch.
- Voice Select Warp—You must subscribe to the NTT Communications Corporation Voice Select Warp feature connect the router to a Japanese INS-NET-64 switch.
- Nariwake—You must be subscribed to the NTT Communications Corporation service for distinctive incoming calls.
- Trouble Call Blocking—You must be subscribed to the NTT Communications Corporation service feature for refusing troublesome calls.

Configuring Enhanced Voice Services for Japan Cisco IOS Features

Many of the Enhanced Voice Services for Japan Cisco IOS features were developed for other Cisco routers before they were ported to Cisco 800 series routers. In some cases, CLI commands were created or modified to allow the features to run on Cisco 800 series routers. The following sections provide step instructions for configuring only those features that require new or changed Cisco IOS commands specifically created or modified to run on Cisco 800 series routers.

The Local Call Waiting feature is enabled by a single command in global configuration mode; see the section "pots call-waiting" for command syntax. The features POTS Dial and POTS Disconnect are also single commands in Exec mode; see the sections "test pots dial" and "test pots disconnect" for command examples.

The features E Ya Yo, Voice Warp, Voice Select Warp, and Trouble Call Blocking require no new or changed Cisco IOS commands to run on Cisco 800 series routers. To configure these features, see the *Cisco 800 Series Routers Software Configuration Guide*.

Caller ID

The following procedure provides step instructions for configuring the Caller ID feature:

	Command	Purpose
Step 1	`router(config)#` **`pots country jp`**	Configure the router telephone ports to Japanese standards.
Step 2	`router(config)#` **`dial-peer voice`** ***`number`*** **`pots`**	Enter dial-peer configuration mode, and select the POTS port.
Step 3	`router(config-dial-peer)#` **`caller-id`**	Enable the Caller ID feature.

Call Blocking on Caller ID

The following procedure provides step instructions for configuring the Call Blocking on Caller ID feature:

	Command	Purpose
Step 1	`router(config)#` **`dial-peer voice`** ***`number`*** **`pots`**	Enter dial-peer configuration mode, and select the POTS port.
Step 2	`router(config-dial-peer)#` **`block-caller`** ***`number`***	Block the Caller ID: *number*. For example, block incoming calls from the telephone number 408-555-1234.

Nariwake

The following procedure provides step instructions for configuring the Nariwake feature:

	Command	Purpose
Step 1	`router(config)#` **`pots country jp`**	Configure the router telephone ports to Japanese standards.
Step 2	`router(config)#` **`dial-peer voice`** ***`number`*** **`pots`**	Enter dial-peer configuration mode, and select the POTS port.
Step 3	`router(config-dial-peer)#` **`registered-caller ring`** ***`cadence`***	Configure the Nariwake service registered caller ring cadence. For example, you could set the ring cadence for registered callers to 2.
Step 4	`router(config-dial-peer)#` **`destination-pattern not-provided`**	Optional. If you also subscribe to I Number and Dial-In services, configure a dial-peer.

I Number

The following procedure provides step instructions for configuring the I Number feature:

	Command	Purpose
Step 1	`router(config)#` **`int bri`** ***`number`***	Enter the basic rate interface number, such as **bri0**.
Step 2	`router(config-if)#` **`isdn i-number`** ***`n1 ldn1`***	Configure the first router POTS port to use a single subscriber line, such as 5551234.
Step 3	`router(config-if)#` **`isdn i-number`** ***`n2 ldn2`***	Configure the second router POTS port to use a single subscriber line, such as 5556789.
Step 4	`router(config-if)#` **`exit`**	Exit basic rate interface configuration.
Step 5	`router(config)#` **`dial-peer voice`** ***`number`*** **`pots`**	Enter dial-peer configuration mode, and select the POTS port.
Step 6	`router(config-dial-peer)#` **`destination-pattern 5551234`**	Set the first dial-peer destination pattern to the corresponding LDN, such as 5551234.
Step 7	`router(config-dial-peer)#` **`exit`**	Exit **destination-pattern** configuration.
Step 8	`router(config)#` **`dial-peer voice`** ***`number`*** **`pots`**	Enter dial-peer configuration mode, and select the POTS port.
Step 9	`router(config-dial-peer)#` **`destination-pattern`** ***`number`***	Set the second dial-peer destination pattern to the corresponding LDN.
Step 10	`router(config-dial-peer)#` **`exit`**	Exit **destination-pattern** configuration.

Configuration Examples

See the "Examples" headings in the sections "Command Reference" and "Debug Commands" for commands samples.

Command Reference

This section documents new or changed commands for the Enhanced Voice Services for Japan Cisco IOS features. All other commands used with these commands are documented in the Cisco IOS Release 12.1 configuration and command-reference publications.

- **caller-id**
- **block-caller**
- **isdn i-number**
- **pots call-waiting**
- **registered-caller ring**
- **show pots csm**
- **test pots dial**
- **test pots disconnect**

caller-id

To enable Caller ID, use the dial-peer configuration **caller-id** command.

caller-id

no caller-id

The **no** form of the **caller-id** command disables Caller ID.

Syntax Description

This command contains no arguments or keywords.

Defaults

Caller ID is disabled.

Command Modes

Dial-peer configuration.

Command History

Release	Modification
Cisco IOS Release 12.1.(2)XF	The command **caller-id** was introduced on the Cisco 800 series routers.

Usage Guidelines

This command is available on Cisco 800 series routers that have POTS ports. The command is effective only if you subscribe to Caller ID service. If you enable Caller ID on a router without subscribing to the Caller ID service, Caller ID information does not appear on the telephone display.

The configuration of Caller ID must match the device connected to the POTS port. That is, if a telephone supports the Caller ID feature, use the command **caller-id** to enable the feature or if the telephone does not support the Caller ID feature, use the command default or disable the Caller ID feature. Odd ringing behavior might occur if the Caller ID feature is disabled when it is a supported telephone feature or enabled when it is not a supported telephone feature.

Examples

The following example enables a router to use the Caller ID feature.

```
router(config)# dial-peer voice 1 pots
router(config-dial-peer)# caller-id
router(config-dial-peer)#
```

Related Commands

Command	Description
block-caller *number*	Configure Call Blocking on Caller ID.
debug pots csm csm	Activate events from which an application can determine and display the status and progress of calls to and from POTS ports.
isdn i-number	Configure several terminal devices to use one subscriber line.
pots call-waiting	Enable local call waiting on a router.
registered-caller ring	Configure the Nariwake service registered caller ring cadence.

block-caller

To configure Call Blocking on Caller ID, use the dial-peer configuration **block-caller** command.

block-caller *number*

no block-caller *number*

The **no** form of the **block-caller** command disables Call Blocking on Caller ID.

Syntax Description

Command Elements	Description
number	telephone number to block. You can use a period (.) as a digit wildcard. For example, the command **block-caller 5.51234** blocks all numbers beginning with the digit 5, followed by any digit, and then sequentially followed by the digits 5, 1, 2, 3, and 4.

Defaults

Call blocking is disabled; the router does not block any calls for any LDNs based on Caller ID numbers.

Command Modes

Dial-peer configuration.

Command History

Release	Modification
Cisco IOS Release 12.1.(2)XF	The command **block-caller** was introduced on the Cisco 800 series routers.

Usage Guidelines

This command is available on Cisco 800 series routers that have POTS ports. For each dial-peer, you can enter up to ten Caller ID numbers to block. The routers do not accept additional Caller ID numbers if ten numbers already exist. In this case, a number must be removed before another Caller ID number can be added for blocking.

If you do not specify the block-caller command for a local directory, all voice calls to that local directory are accepted. If you specify the block-caller command for a local directory, the router verifies that the incoming calling-party number does not match any Caller ID numbers in that local directory before processing or accepting the voice call. Each specified Caller ID number and incoming calling-party number is compared from right to left, up to the number of digits in the specified Caller ID number or incoming calling-party number, whichever has less digits.

This command is effective only if you subscribe to Caller ID service. If you enable call blocking on Caller ID without subscribing to the Caller ID service, the routers do not perform the verification process on calling-party numbers and do not block any calls.

Examples

The following example configures a router to block calls from a caller whose Caller ID number is 408-555-1234.

```
router(config)# dial-peer voice 1 pots
router(config-dial-peer)# block-caller 4085551234
router(config-dial-peer)#
```

Related Commands

Command	Description
caller-id	Identify incoming calls with Caller ID.
debug pots csm csm	Activate events from which an application can determine and display the status and progress of calls to and from POTS ports.
isdn i-number	Configure several terminal devices to use one subscriber line.
pots call-waiting	Enable local call waiting on a router.
registered-caller ring	Configure the Nariwake service registered caller ring cadence.

isdn i-number

To configure several terminal devices to use one subscriber line, use the global configuration **isdn i-number** command.

isdn i-number *n ldn*

Syntax Description

Command Elements	Description
n	Subscriber line 1, 2 or 3, as specified in the NTT specification.
ldn	LDN assigned to the router POTS port.

Defaults

The default is each terminal device uses one subscriber line.

Command Modes

Interface BRI0 configuration.

Command History

Release	Modification
Cisco IOS Release 12.1.(2)XF	The command **isdn i-number** was introduced on the Cisco 800 series routers.

Usage Guidelines

Enter the command **interface bri** before entering the command **isdn i-number**.

Examples

The following example shows screen output for two LDNs configured under interface BRI0:

```
router(config)# interface bri0
router(config-if)# isdn i-number 1 5551234
router(config-if)# isdn i-number 2 5556789
router(config-if)# exit
router(config)# dial-peer voice 1 pots
router(config-dial-peer)# destination-pattern 5551234
router(config-dial-peer)# exit
router(config)# dial-peer voice 2 pots
router(config-dial-peer)# destination-pattern 5556789
router(config-dial-peer)# exit
router(config)#
```

Related Commands

Command	Description
interface bri	Configure a BRI interface and enter interface configuration mode.

pots call-waiting

To enable the Local Call Waiting feature on a Cisco 800 series router, use the global configuration **pots call-waiting** command.

pots call-waiting [**local** | **remote**]

no pots call-waiting [**local** | **remote**]

Syntax Description

Command Elements	Description
local	Enable call waiting on a local basis for the routers.
remote	Rely on the network provider service instead of the router to hold calls.

Defaults

The call waiting default is **remote** if the Call Waiting feature is not configured. In that case, the call holding pattern follows the settings of the service provider rather than those of the router.

Command Modes

Global configuration.

Command History

Release	Modification
Cisco IOS Release 12.1.(2)XF	The command **pots call-waiting** was introduced on the Cisco 800 series routers.

Usage Guidelines

To display the call waiting setting, use the command **show run** or **show pots status**. The ISDN call waiting service is used if it is available on the ISDN line connected to the router even if local call waiting is configured on the router. That is, if the ISDN line supports call waiting, the local call waiting configuration on the router is ignored.

Examples

The following example enables local call waiting on a router:

```
router(config)# pots call-waiting local
router(config)#
```

Related Commands

Command	Description
call waiting	Configure Call Waiting for a specific dial-peer.

registered-caller ring

To configure the Nariwake service registered caller ring cadence, use the dial-peer configuration **registered-caller ring** command.

registered-caller ring *cadence*

Syntax Description

Command Elements	Description
cadence	A value of 0, 1, or 2. The default ring cadence for registered callers is 1 and for unregistered callers is 0. The on and off periods of ring 0 (normal ringing signals) and ring 1 (ringing signals for the Nariwake service) are defined in the NTT user manual.

Defaults

The default Nariwake service registered caller ring cadence is Ring 1.

Command Modes

Dial-peer configuration.

Command History

Release	Modification
Cisco IOS Release 12.1.(2)XF	The command **registered-caller ring** was introduced on the Cisco 800 series routers.

Usage Guidelines

If your ISDN line is provisioned for the I Number or dial-in services, you must also configure a dial-peer by using the command **destination-pattern not-provided.** Either port 1 or 2 can be configured under this dial-peer. The router then forwards the incoming call to the voice port 1. See the “Examples” section for details.

If more than one dial-peer is configured with **destination-pattern not-provided**, the router uses the first configured dial-peer for the incoming calls. To display the Nariwake ring cadence setting, use the **show run** command.

Examples

The following example sets the ring cadence for registered callers to 2.

```
router(config)# pots country jp
router(config)# dial-peer voice 1 pots
router(config-dial-peer)# registered-caller ring 2
router(config-dial-peer)#
```

Add the **destination-pattern not-provided** command if you also subscribe to the I Number and dial-in services.

```
router(config-dial-peer)# destination-pattern not-provided
router(config-dial-peer)#
```

show pots csm

To show the current state of calls and the most recent event received by the call switching module (CSM) on the Cisco 800 series router, use the Exec **show pots csm** command.

show pots csm *port*

Syntax Description

Command Elements	Description
port	Port number 1 or 2.

Command Modes

Exec.

Command History

Release	Modification
Cisco IOS Release 12.1.(2)XF	The command **show pots csm** was introduced on the Cisco 800 series routers.

Examples

The following is an example of **show pots csm** command output:

```
router# show pots csm 1

POTS PORT: 1

   CSM Finite State Machine:
      Call 0 - State: idle, Call Id: 0x0
               Active: no
               Event: CSM_EVENT_NONE Cause: 0
      Call 1 - State: idle, Call Id: 0x0
               Active: no
               Event: CSM_EVENT_NONE Cause: 0
      Call 2 - State: idle, Call Id: 0x0
               Active: no
               Event: CSM_EVENT_NONE Cause: 0

router#
```

Related Commands

Command	Description
test pots dial	Dial a telephone number for the POTS port on the router by using a dial application on your workstation.
test pots disconnect	Disconnect a telephone call for the POTS port on the router.

test pots dial

To dial a telephone number for the POTS port on the router by using a dial application on your workstation, use the Exec **test pots dial** command.

test pots *port* **dial** *number*[#]

If the telephone is on the hook when you issue the dial command, the router rings the telephone, waits until the telephone is taken off the hook, and then dials the requested number. If the telephone is off the hook and providing a dial tone when you issue the command, the router dials the requested number.

Syntax Description

Command Elements	Description
port	Port number 1 or 2.
number	Telephone number to dial.
#	Turn off dual tone multifrequency (DTMF) detection from the telephone while sending the *enbloc* signal. If you do not include the pound sign character (**#**) to terminate the *number* variable, you can use the telephone keypad to complete the call.

Command Modes

Exec.

Command History

Release	Modification
Cisco IOS Release 12.1.(2)XF	The command **test pots** *port* **dial** was introduced on the Cisco 800 series routers.

Examples

The following POTS dial command dials the telephone number 408-555-1234:

```
router# test pots 1 dial 4085551234#
router#
```

For an example of the command **test pots** *port* **dial** with debug output, see the command-reference section “debug pots csm.”

Related Commands

Command	Description
show pots csm	Show the current state of calls and the most recent event received by the CSM on the router.
test pots disconnect	Disconnect a telephone call for the POTS port on the router.

test pots disconnect

To disconnect a telephone call for the POTS port on the router, use the Exec **test pots disconnect** command.

test pots *port* **disconnect**

Syntax Description

Command Elements	Description
port	Port number 1 or 2.

Command Modes

Exec.

Command History

Release	Modification
Cisco IOS Release 12.1.(2)XF	The command **test pots** *port* **disconnect** was introduced on the Cisco 800 series routers.

Examples

The following POTS disconnect command disconnects a telephone call from POTS port 1:

```
router# test pots 1 disconnect
router#
```

For an example of the command **test pots** *port* **disconnect** command with debug output, see the command-reference section “debug pots csm.”

Related Commands

Command	Description
show pots csm	Show the current state of calls and the most recent event received by the CSM on the router.
test pots dial	Dial a telephone number for the POTS port on the router by using a dial application on your workstation.

Debug Commands

This section documents new **debug** command.

debug pots csm

To activate events from which an application can determine and display the status and progress of calls to and from POTS ports, use the Exec **debug pots csm** command.

debug pots csm

Syntax Description

Command Elements	Description
csm	Call switching module. See the section "Usage Guidelines" for details.

Command Modes

Exec.

Command History

Release	Modification
Cisco IOS Release 12.1.(2)XF	The command **debug pots csm** was introduced on the Cisco 800 series routers.

Usage Guidelines

To see debug messages, enter Configuration mode command **logging console** as follows:

```
router(config) # logging console
router(config) # exit
```

Debug messages are displayed in one of two formats that are relevant to the POTS dial feature:

```
hh:mm:ss: CSM_STATE: CSM_EVENT, call id = ??, port = ?
```

or

```
hh:mm:ss: EVENT_FROM_ISDN:dchan_idb=0x???????, call_id=0x????, ces=? bchan=0x????????,
event=0x?, cause=0x??
```

Each format element is defined as follows:

Command Elements	Description
hh:mm:ss	Timestamp in hours, minutes, and seconds.
CSM_STATE	One of the call CSM states listed in Table 1.
CSM_EVENT	One of the CSM events listed in Table 2.
call id	Hexadecimal value from 0x00 to 0xFF.

Command Elements	Description
port	Telephone port 1 or 2.
EVENT_FROM_ISDN	A CSM event. Table 2 shows a list of CSM events.
dchan_idb	Internal data structure address.
ces	Connection end point suffix used by ISDN.
bchan	Channel used by the call. A value of 0xFFFFFFFF indicates that a channel is not assigned.
event	A hexadecimal value that is translated into a CSM event. Table 3 shows a list of events and the corresponding CSM events.
cause	A hexadecimal value that is given to call-progressing events. Table 4 shows a list of cause values and definitions.

CSM States

Table 1 shows the values for CSM states.

Table 1 CSM States

CSM State	Description
CSM_IDLE_STATE	Telephone on the hook.
CSM_RINGING	Telephone ringing.
CSM_SETUP	Setup for outgoing call in progress.
CSM_DIALING	Dialing number of outgoing call.
CSM_IVR_DIALING	Interactive voice response (IVR) for Japanese telephone dialing.
CSM_CONNECTING	Waiting for carrier to connect the call.
CSM_CONNECTED	Call connected.
CSM_DISCONNECTING	Waiting for carrier to disconnect the call.
CSM_NEAR_END_DISCONNECTING	Waiting for carrier to disconnect the call.
CSM_HARD_HOLD	Call on hard hold.
CSM_CONSULTATION_HOLD	Call on consultation hold.
CSM_WAIT_FOR_HOLD	Waiting for carrier to put call on hard hold.
CSM_WAIT_FOR_CONSULTATION_HOLD	Waiting for carrier to put call on consultation hold.
CSM_CONFERENCE	Waiting for carrier to complete call conference.
CSM_TRANSFER	Waiting for carrier to transfer call.
CSM_APPLIC_DIALING	Call initiated from IOS CLI.

CSM Events

Table 2 shows the values for CSM events.

Table 2 CSM Events

CSM Events	Description
CSM_EVENT_INTER_DIGIT_TIMEOUT	Time waiting for dial digits has expired.
CSM_EVENT_TIMEOUT	Near or far end disconnect timeout.
CSM_EVENT_ISDN_CALL	Incoming call.
CSM_EVENT_ISDN_CONNECTED	Call connected.
CSM_EVENT_ISDN_DISCONNECT	Far end disconnected.
CSM_EVENT_ISDN_DISCONNECTED	Call disconnected.
CSM_EVENT_ISDN_SETUP	Outgoing call requested.
CSM_EVENT_ISDN_SETUP_ACK	Outgoing call accepted.
CSM_EVENT_ISDN_PROC	Call proceeding and dialing completed.
CSM_EVENT_ISDN_CALL_PROGRESSING	Call being received in band tone.
CSM_EVENT_ISDN_HARD_HOLD	Call on hard hold.
CSM_EVENT_ISDN_HARD_HOLD_REJ	Hold attempt rejected.
CSM_EVENT_ISDN_CHOLD	Call on consultation hold.
CSM_EVENT_ISDN_CHOLD_REJ	Consultation hold attempt rejected.
CSM_EVENT_ISDN_RETRIEVED	Call retrieved.
CSM_EVENT_ISDN_RETRIEVE_REJ	Call retrieval attempt rejected.
CSM_EVENT_ISDN_TRANSFERRED	Call transferred.
CSM_EVENT_ISDN_TRANSFER_REJ	Call transfer attempt rejected.
CSM_EVENT_ISDN_CONFERENCE	Call conference started.
CSM_EVENT_ISDN_CONFERENCE_REJ	Call conference attempt rejected.
CSM_EVENT_ISDN_IF_DOWN	ISDN interface down.
CSM_EVENT_ISDN_INFORMATION	ISDN information element received (used by NTT IVR application).
CSM_EVENT_VDEV_OFFHOOK	Telephone off the hook.
CSM_EVENT_VDEV_ONHOOK	Telephone on the hook.
CSM_EVENT_VDEV_FLASHHOOK	Telephone hook switch has flashed.
CSM_EVENT_VDEV_DIGIT	DTMF digit has been detected.
CSM_EVENT_VDEV_APPLICATION_CALL	Call initiated from IOS CLI.

Events

Table 3 shows the values for events that are translated into CSM events.

Table 3 Event Values

Hexadecimal Value	Event	CSM Event
0x0	DEV_IDLE	CSM_EVENT_ISDN_DISCONNECTED
0x1	DEV_INCALL	CSM_EVENT_ISDN_CALL
0x2	DEV_SETUP_ACK	CSM_EVENT_ISDN_SETUP_ACK
0x3	DEV_CALL_PROC	CSM_EVENT_ISDN_PROC
0x4	DEV_CONNECTED	CSM_EVENT_ISDN_CONNECTED
0x5	DEV_CALL_PROGRESSING	CSM_EVENT_ISDN_CALL_PROGRESSING
0x6	DEV_HOLD_ACK	CSM_EVENT_ISDN_HARD_HOLD
0x7	DEV_HOLD_REJECT	CSM_EVENT_ISDN_HARD_HOLD_REJ
0x8	DEV_CHOLD_ACK	CSM_EVENT_ISDN_CHOLD
0x9	DEV_CHOLD_REJECT	CSM_EVENT_ISDN_CHOLD_REJ
0xa	DEV_RETRIEVE_ACK	CSM_EVENT_ISDN_RETRIEVED
0xb	DEV_RETRIEVE_REJECT	CSM_EVENT_ISDN_RETRIEVE_REJ
0xc	DEV_CONFR_ACK	CSM_EVENT_ISDN_CONFERENCE
0xd	DEV_CONFR_REJECT	CSM_EVENT_ISDN_CONFERENCE_REJ
0xe	DEV_TRANS_ACK	CSM_EVENT_ISDN_TRANSFERRED
0xf	DEV_TRANS_REJECT	CSM_EVENT_ISDN_TRANSFER_REJ

Cause

This table shows cause values that are assigned only to call-progressing events.

Table 4 Cause Values

Hexadecimal Value	Cause Definitions
0x01	UNASSIGNED_NUMBER
0x02	NO_ROUTE
0x03	NO_ROUTE_DEST
0x04	NO_PREFIX
0x06	CHANNEL_UNACCEPTABLE
0x07	CALL_AWARDED
0x08	CALL_PROC_OR_ERROR
0x09	PREFIX_DIALED_ERROR
0x0a	PREFIX_NOT_DIALED
0x0b	EXCESSIVE_DIGITS
0x0d	SERVICE_DENIED

Table 4 Cause Values (continued)

Hexadecimal Value	Cause Definitions
0x10	NORMAL_CLEARING
0x11	USER_BUSY
0x12	NO_USER_RESPONDING
0x13	NO_USER_ANSWER
0x15	CALL_REJECTED
0x16	NUMBER_CHANGED
0x1a	NON_SELECTED_CLEARING
0x1b	DEST_OUT_OF_ORDER
0x1c	INVALID_NUMBER_FORMAT
0x1d	FACILITY_REJECTED
0x1e	RESP_TO_STAT_ENQ
0x1f	UNSPECIFIED_CAUSE
0x22	NO_CIRCUIT_AVAILABLE
0x26	NETWORK_OUT_OF_ORDER
0x29	TEMPORARY_FAILURE
0x2a	NETWORK_CONGESTION
0x2b	ACCESS_INFO_DISCARDED
0x2c	REQ_CHANNEL_NOT_AVAIL
0x2d	PRE_EMPTED
0x2f	RESOURCES_UNAVAILABLE
0x32	FACILITY_NOT_SUBSCRIBED
0x33	BEARER_CAP_INCOMPAT
0x34	OUTGOING_CALL_BARRED
0x36	INCOMING_CALL_BARRED
0x39	BEARER_CAP_NOT_AUTH
0x3a	BEAR_CAP_NOT_AVAIL
0x3b	CALL_RESTRICTION
0x3c	REJECTED_TERMINAL
0x3e	SERVICE_NOT_ALLOWED
0x3f	SERVICE_NOT_AVAIL
0x41	CAP_NOT_IMPLEMENTED
0x42	CHAN_NOT_IMPLEMENTED
0x45	FACILITY_NOT_IMPLEMENT
0x46	BEARER_CAP_RESTRICTED
0x4f	SERV_OPT_NOT_IMPLEMENT
0x51	INVALID_CALL_REF

Table 4 Cause Values (continued)

Hexadecimal Value	Cause Definitions
0x52	CHAN_DOES_NOT_EXIST
0x53	SUSPENDED_CALL_EXISTS
0x54	NO_CALL_SUSPENDED
0x55	CALL_ID_IN_USE
0x56	CALL_ID_CLEARED
0x58	INCOMPATIBLE_DEST
0x5a	SEGMENTATION_ERROR
0x5b	INVALID_TRANSIT_NETWORK
0x5c	CS_PARAMETER_NOT_VALID
0x5f	INVALID_MSG_UNSPEC
0x60	MANDATORY_IE_MISSING
0x61	NONEXISTENT_MSG
0x62	WRONG_MESSAGE
0x63	BAD_INFO_ELEM
0x64	INVALID_ELEM_CONTENTS
0x65	WRONG_MSG_FOR_STATE
0x66	TIMER_EXPIRY
0x67	MANDATORY_IE_LEN_ERR
0x6f	PROTOCOL_ERROR
0x7f	INTERWORKING_UNSPEC

Examples

This section provides debug output examples for three call scenarios, displaying the sequence of events that occur during a POTS dial call or POTS disconnect call.

- Call Scenario 1

  In this example call scenario, port 1 is on the hook, the application dial is set to call 4085552221, and the far-end successfully connects.

```
router# debug pots csm
router# test pots 1 dial 4085552221#
router#
```

  The following screen output shows an event indicating that port 1 is being used by the dial application:

```
01:58:27: CSM_PROC_IDLE: CSM_EVENT_VDEV_APPLICATION_CALL, call id = 0x0, port = 1
```

The following screen output shows events indicating that the CSM is receiving the application digits of the number to dial:

```
01:58:27: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
01:58:27: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
01:58:27: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
01:58:27: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
01:58:27: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
01:58:27: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
01:58:27: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
01:58:27: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
01:58:27: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
01:58:27: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
```

The following screen output shows that the telephone connected to port 1 is off the hook:

```
01:58:39: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_OFFHOOK, call id = 0x0, port = 1
```

The following screen output shows a call-proceeding event pair indicating that the router ISDN software has sent the dialed digits to the ISDN switch:

```
01:58:40: EVENT_FROM_ISDN:dchan_idb=0x280AF38, call_id=0x8004, ces=0x1 bchan=0x0,
event=0x3, cause=0x0
01:58:40: CSM_PROC_ENBLOC_DIALING: CSM_EVENT_ISDN_PROC, call id =
0x8004, port = 1
```

The following screen output shows the call-progressing event pair indicating that the telephone at the far end is ringing:

```
01:58:40: EVENT_FROM_ISDN:dchan_idb=0x280AF38, call_id=0x8004, ces=0x1
bchan=0xFFFFFFFF, event=0x5, cause=0x0
01:58:40: CSM_PROC_ENBLOC_DIALING: CSM_EVENT_ISDN_CALL_PROGRESSING, call id = 0x8004,
port = 1
```

The following screen output shows a call-connecting event pair indicating that the telephone at the far end has answered:

```
01:58:48: EVENT_FROM_ISDN:dchan_idb=0x280AF38, call_id=0x8004, ces=0x1
bchan=0xFFFFFFFF, event=0x4, cause=0x0
01:58:48: CSM_PROC_CONNECTING: CSM_EVENT_ISDN_CONNECTED, call id = 0x8004, port = 1
```

The following screen output shows a call-progressing event pair indicating that the telephone at the far end has hung up and that the calling telephone is receiving an in-band tone from the ISDN switch:

```
01:58:55: EVENT_FROM_ISDN:dchan_idb=0x280AF38, call_id=0x8004, ces=0x1
bchan=0xFFFFFFFF, event=0x5, cause=0x10
01:58:55: CSM_PROC_CONNECTED: CSM_EVENT_ISDN_CALL_PROGRESSING, call id = 0x8004, port
= 1
```

The following screen output shows that the telephone connected to port 1 has hung up:

```
01:58:57: CSM_PROC_CONNECTED: CSM_EVENT_VDEV_ONHOOK, call id = 0x8004, port = 1
```

The following screen output shows an event pair indicating that the call has been terminated:

```
01:58:57: EVENT_FROM_ISDN:dchan_idb=0x280AF38, call_id=0x8004, ces=0x1
bchan=0xFFFFFFFF, event=0x0, cause=0x0
01:58:57: CSM_PROC_NEAR_END_DISCONNECT: CSM_EVENT_ISDN_DISCONNECTED, call id =
0x8004, port = 1
813_local#
```

- Call Scenario 2

In this example scenario, port 1 is on the hook, the application dial is set to call 4085552221, and the destination number is busy.

```
router# debug pots csm
router# test pots 1 dial 4085552221#
router#
```

The following screen output shows that port 1 is used by the dial application:

```
01:59:42: CSM_PROC_IDLE: CSM_EVENT_VDEV_APPLICATION_CALL, call id = 0x0, port = 1
```

The following screen output shows the events indicating that the CSM is receiving the application digits of the number to call:

```
01:59:42: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
01:59:42: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
01:59:42: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
01:59:42: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
01:59:42: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
01:59:42: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
01:59:42: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
01:59:42: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
01:59:42: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
01:59:42: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
```

The following screen output shows an event indicating that the telephone connected to port 1 is off the hook:

```
01:59:52: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_OFFHOOK, call id = 0x0, port = 1
```

The following screen output shows a call-proceeding event pair indicating that the telephone at the far end is busy:

```
01:59:52: EVENT_FROM_ISDN:dchan_idb=0x280AF38, call_id=0x8005, ces=0x1 bchan=0x0,
event=0x3, cause=0x11
01:59:52: CSM_PROC_ENBLOC_DIALING: CSM_EVENT_ISDN_PROC, call id = 0x8005, port = 1
```

The following screen output shows a call-progressing event pair indicating that the calling telephone is receiving an in-band busy tone from the ISDN switch:

```
01:59:58: EVENT_FROM_ISDN:dchan_idb=0x280AF38, call_id=0x8005, ces=0x1
bchan=0xFFFFFFFF, event=0x5, cause=0x0
01:59:58: CSM_PROC_ENBLOC_DIALING: CSM_EVENT_ISDN_CALL_PROGRESSING, call id = 0x8005,
port = 1
```

The following screen output shows an event indicating that the calling telephone has hung up:

```
02:00:05: CSM_PROC_ENBLOC_DIALING: CSM_EVENT_VDEV_ONHOOK, call id = 0x8005, port = 1
```

The following screen output shows an event pair indicating that the call has been terminated:

```
02:00:05: EVENT_FROM_ISDN:dchan_idb=0x280AF38, call_id=0x8005, ces=0x1
bchan=0xFFFFFFFF, event=0x0, cause=0x0
02:00:05: CSM_PROC_NEAR_END_DISCONNECT: CSM_EVENT_ISDN_DISCONNECTED, call id =
0x8005, port = 1
```

- Call Scenario 3

  In this example call scenario, port 1 is on the hook, the application dial is set to call 408-666-1112, the far end successfully connects, and the command **test pots disconnect** terminates the call.

```
router# debug pots csm
router# test pots 1 dial 4086661112
router#
```

The following screen output follows the same sequence of events as shown in Call Scenario 1:

```
1d03h: CSM_PROC_IDLE: CSM_EVENT_VDEV_APPLICATION_CALL, call id = 0x0, port = 1
1d03h: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
1d03h: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
1d03h: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
1d03h: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
1d03h: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
1d03h: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
1d03h: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
1d03h: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
1d03h: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1
1d03h: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_DIGIT, call id = 0x0, port = 1

1d03h: CSM_PROC_APPLIC_DIALING: CSM_EVENT_VDEV_OFFHOOK, call id = 0x0, port = 1

1d03h: EVENT_FROM_ISDN:dchan_idb=0x2821F38, call_id=0x8039, ces=0x1
   bchan=0x0, event=0x3, cause=0x0
1d03h: CSM_PROC_ENBLOC_DIALING: CSM_EVENT_ISDN_PROC, call id = 0x8039, port = 1

1d03h: EVENT_FROM_ISDN:dchan_idb=0x2821F38, call_id=0x8039, ces=0x1
   bchan=0xFFFFFFFF, event=0x5, cause=0x0

1d03h: CSM_PROC_ENBLOC_DIALING: CSM_EVENT_ISDN_CALL_PROGRESSING, call id = 0x8039,
    port = 1

router# test pots 1 disconnect
```

The test pots disconnect command disconnects the call before you physically have to put the telephone back on the hook.

```
1d03h: CSM_PROC_CONNECTING: CSM_EVENT_VDEV_APPLICATION_HANGUP_CALL, call id = 0x8039,
     port = 1
1d03h: EVENT_FROM_ISDN:dchan_idb=0x2821F38, call_id=0x8039, ces=0x1
   bchan=0xFFFFFFFF, event=0x0, cause=0x0

1d03h: CSM_PROC_DISCONNECTING: CSM_EVENT_ISDN_DISCONNECTED, call id = 0x8039,
    port = 1
1d03h: CSM_PROC_DISCONNECTING: CSM_EVENT_TIMEOUT, call id = 0x8039, port = 1
```