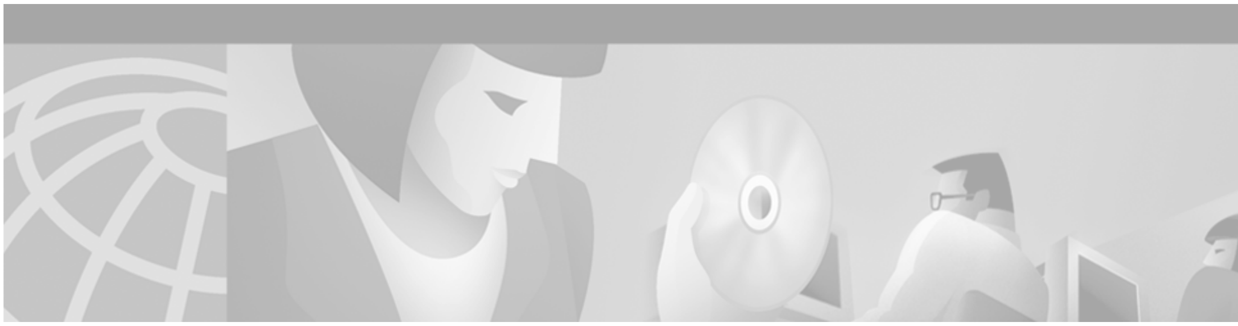

# Configurable PSTN Cause Code to SIP Response Mapping

### Document Update Alert

This document was originally produced for Cisco IOS Release 12.2(11)T. This feature has been updated in subsequent releases, and more recent documentation is available.

**If you are using Cisco IOS Release 12.2(11)T or higher**, refer to the following section in the Configuring SIP Message Components, Session Timers, and Responses chapter of the *Cisco IOS SIP Configuration Guide*, Cisco IOS Voice Configuration Library, Release 12.3:

- *Configurable PSTN Cause Code to SIP Response Mapping*

### Feature History

| Release | Modification |
|---|---|
| 12.2(2)XB | This feature was introduced on the Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco AS5300, Cisco AS5350, and Cisco AS5400 platforms. |
| 12.2(2)XB2 | This feature was introduced on the Cisco AS5850 universal gateway. |
| 12.2(8)T | This feature was integrated into Cisco IOS Release 12.2(8)T. The Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms were not supported in this release. |
| 12.2(11)T | This feature was integrated into Cisco IOS Release 12.2(11)T and support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms. |

This document describes Configurable PSTN Cause Code to SIP Response Mapping in Cisco IOS Release 12.2(11)T. This feature allows customization of the standard RFC 2543 mappings between the Session Initiation Protocol (SIP) and the Public Switched Telephone Network (PSTN) networks.

This document includes the following sections:

# Feature Overview

For calls to be established between a SIP network and a PSTN network, the two networks must be able to interoperate. One aspect of their interoperation is the mapping of PSTN cause codes, which indicate reasons for PSTN call failure or completion, to SIP status codes or events. The opposite is also true: SIP status codes or events are mapped to PSTN cause codes. Event mapping tables found in this document show the standard or default mappings between SIP and PSTN.

However, you may want to customize the SIP user agent software to override the default mappings between the SIP and PSTN networks. The Configurable PSTN Cause Code to SIP Response Mapping feature allows you to configure specific map settings between the PSTN and SIP networks. Thus, any SIP status code can be mapped to any PSTN cause code, or vice versa.

When set, these settings can be stored in the NVRAM and are restored automatically on bootup.

## Default Mappings

The following table lists PSTN cause codes and the corresponding SIP event mappings that are set by default. Any code other than the codes listed are mapped by default to *500 Internal server error*.

***Table 1 Default PSTN Cause Code to SIP Event Mappings***

| PSTN Cause Code | Description | SIP Event |
|---|---|---|
| 1 | Unallocated number | 404 Not found |
| 2 | No route to specified transit network | 404 Not found |
| 3 | No route to destination | 404 Not found |
| 17 | User busy | 486 Busy here |
| 18 | No user response | 480 Temporarily unavailable |
| 19 | No answer from the user | |
| 20 | Subscriber absent | |
| 21 | Call rejected | 403 Forbidden |
| 22 | Number changed | 410 Gone |
| 26 | Non-selected user clearing | 404 Not found |
| 27 | Destination out of order | 404 Not found |
| 28 | Address incomplete | 484 Address incomplete |
| 29 | Facility rejected | 501 Not implemented |
| 31 | Normal, unspecified | 404 Not found |
| 34 | No circuit available | 503 Service unavailable |
| 38 | Network out of order | 503 Service unavailable |
| 41 | Temporary failure | 503 Service unavailable |
| 42 | Switching equipment congestion | 503 Service unavailable |
| 47 | Resource unavailable | 503 Service unavailable |
| 55 | Incoming class barred within Closed User Group (CUG) | 403 Forbidden |
| 57 | Bearer capability not authorized | 403 Forbidden |

*Table 1 Default PSTN Cause Code to SIP Event Mappings (continued)*

| PSTN Cause Code | Description | SIP Event |
|---|---|---|
| 58 | Bearer capability not presently available | 501 Not implemented |
| 65 | Bearer capability not implemented | 501 Not implemented |
| 79 | Service or option not implemented | 501 Not implemented |
| 87 | User not member of Closed User Group (CUG) | 503 Service Unavailable |
| 88 | Incompatible destination | 400 Bad request |
| 95 | Invalid message | 400 Bad request |
| 102 | Recover on Expires timeout | 408 Request timeout |
| 111 | Protocol error | 400 Bad request |
| **Any code other than those listed above:** | | **500 Internal server error** |

The following table lists the SIP events and the corresponding PSTN cause codes mappings that are set by default.

*Table 2 Default SIP Event to PSTN Cause Code Mapping*

| SIP Event | PSTN Cause Code | Description |
|---|---|---|
| 400 Bad request | 127 | Interworking, unspecified |
| 401 Unauthorized | 57 | Bearer capability not authorized |
| 402 Payment required | 21 | Call rejected |
| 403 Forbidden | 57 | Bearer capability not authorized |
| 404 Not found | 1 | Unallocated number |
| 405 Method not allowed | 127 | Interworking, unspecified |
| 406 Not acceptable | | |
| 407 Proxy authentication required | 21 | Call rejected |
| 408 Request timeout | 102 | Recover on Expires timeout |
| 409 Conflict | 41 | Temporary failure |
| 410 Gone | 1 | Unallocated number |
| 411 Length required | 127 | Interworking, unspecified |
| 413 Request entity too long | | |
| 414 Request URI (URL) too long | | |
| 415 Unsupported media type | 79 | Service or option not implemented |
| 420 Bad extension | 127 | Interworking, unspecified |
| 480 Temporarily unavailable | 18 | No user response |
| 481 Call leg does not exist | 127 | Interworking, unspecified |
| 482 Loop detected | | |
| 483 Too many hops | | |
| 484 Address incomplete | 28 | Address incomplete |

*Table 2 Default SIP Event to PSTN Cause Code Mapping (continued)*

| SIP Event | PSTN Cause Code | Description |
|---|---|---|
| 485 Address ambiguous | 1 | Unallocated number |
| 486 Busy here | 17 | User busy |
| 487 Request cancelled | 127 | Interworking, unspecified |
| 488 Not acceptable here | 127 | Interworking, unspecified |
| 500 Internal server error | 41 | Temporary failure |
| 501 Not implemented | 79 | Service or option not implemented |
| 502 Bad gateway | 38 | Network out of order |
| 503 Service unavailable | 63 | Service or option unavailable |
| 504 Gateway timeout | 102 | Recover on Expires timeout |
| 505 Version not implemented | 127 | Interworking, unspecified |
| 580 Precondition Failed | 47 | Resource unavailable, unspecified |
| 600 Busy everywhere | 17 | User busy |
| 603 Decline | 21 | Call rejected |
| 604 Does not exist anywhere | 1 | Unallocated number |
| 606 Not acceptable | 58 | Bearer capability not presently available |

## Benefits

The Configurable PSTN Cause Code to SIP Response Mapping feature offers control and flexibility. By using command-line interface commands, you can easily customize the default or standard mappings that are currently available between PSTN and SIP networks. This allows for flexibility when setting up deployment sites.

## Related Features and Technologies

- Cisco SIP Proxy Server
- Cisco VoIP

## Related Documents

The following documents contain information related to the Cisco SIP functionality:

- *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2
- *Cisco IOS Voice, Video, and Fax Command Reference*, Release 12.2
- *Cisco IOS IP Configuration Guide,* Release 12.2
- *Cisco IOS IP Command Reference, Volume 1 of 3: Addressing and Services*, Release 12.2
- *Cisco IOS IP Command Reference, Volume 2 of 3: Routing Protocols*, Release 12.2

- *Cisco IOS IP Command Reference, Volume 3 of 3: Multicast*, Release 12.2
- *SIP Gateway Support of RSVP and TEL URL,* Release 12.2(2)XB

# Supported Platforms

- Cisco 2600 series
- Cisco 3600 series
- Cisco AS5300 universal access server
- Cisco AS5350 universal gateway
- Cisco AS5400 universal gateway
- Cisco AS5850 universal gateway
- Cisco 7200 series

*Table 3 Cisco IOS Release and Platform Support for this Feature*

| Platform | 12.2(2)XB | 12.2(2)XB2 | 12.2(8)T | 12.2(11)T |
|---|---|---|---|---|
| Cisco 2600 series | X | X | X | X |
| Cisco 3600 series | X | X | X | X |
| Cisco 7200 series | X | X | X | X |
| Cisco AS5300 universal access server | X | X | Not supported | X |
| Cisco AS5350 universal gateway | X | X | Not supported | X |
| Cisco AS5400 universal gateway | X | X | Not supported | X |
| Cisco AS5850 universal gateway | Not supported | X | Not supported | X |

### Determining Platform Support Through Cisco Feature Navigator

Cisco IOS software is packaged in feature sets that support specific platforms. To get updated information regarding platform support for this feature, access Cisco Feature Navigator. Cisco Feature Navigator dynamically updates the list of supported platforms as new platform support is added for the feature.

Cisco Feature Navigator is a web-based tool that enables you to quickly determine which Cisco IOS software images support a specific set of features and which features are supported in a specific Cisco IOS image. You can search by feature or release. Under the release section, you can compare releases side by side to display both the features unique to each software release and the features in common.

To access Cisco Feature Navigator, you must have an account on Cisco.com. If you have forgotten or lost your account information, send a blank e-mail to cco-locksmith@cisco.com. An automatic check will verify that your e-mail address is registered with Cisco.com. If the check is successful, account details with a new random password will be e-mailed to you. Qualified users can establish an account on Cisco.com by following the directions at http://www.cisco.com/register.

Cisco Feature Navigator is updated regularly when major Cisco IOS software releases and technology releases occur. For the most current information, go to the Cisco Feature Navigator home page at the following URL:

http://www.cisco.com/go/fn

### Availability of Cisco IOS Software Images

Platform support for particular Cisco IOS software releases is dependent on the availability of the software images for those platforms. Software images for some platforms may be deferred, delayed, or changed without prior notice. For updated information about platform support and availability of software images for each Cisco IOS software release, refer to the online release notes or, if supported, Cisco Feature Navigator.

**Note** As of Cisco IOS Release 12.2(2)XB, Cisco Feature Navigator does not support features included in this limited-lifetime release.

# Supported Standards, MIBs, and RFCs

### Standards

No new or modified standards are supported by this feature.

### MIBs

- CISCO-SIP-UA-MIB

To obtain lists of supported MIBs by platform and Cisco IOS release, and to download MIB modules, go to the Cisco MIB web site on Cisco.com at the following URL:
http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml.

### RFCs

- RFC 2543, *SIP: Session Initiation Protocol*

# Prerequisites

The following are general prerequisites for SIP deployment:

- Ensure that your Cisco 2600 series, Cisco 3600 series, or Cisco 7200 series router has 16-MB Flash memory and 64-MB DRAM memory, minimum. A Cisco AS5300 must have a minimum of 16-MB Flash memory and 128-MB DRAM memory. A Cisco AS5400 must have a minimum of 32-MB Flash memory and 256-MB DRAM memory.
- Ensure that the gateway has voice functionality that is configurable for SIP.
- Establish a working IP network.

  For more information about configuring IP, refer to:
  *Cisco IOS IP Configuration Guide,* Release 12.2
- Configure VoIP.

  For more information about configuring VoIP, refer to:
  *Cisco IOS Voice, Video, and Fax Command Reference*, Release 12.2

# Configuration Tasks

See the following sections for configuration tasks for the Configurable PSTN Cause Code to SIP Response Mapping feature. Each task in the list is identified as either required or optional.

- Mapping PSTN Codes to SIP Status Codes (optional)
- Mapping SIP Status Codes to PSTN Cause Codes (optional)

## Mapping PSTN Codes to SIP Status Codes

To configure an incoming PSTN cause code to a SIP status code, complete the following steps beginning in global configuration mode:

| | Command | Purpose |
|---|---|---|
| **Step 1** | `Router(config)#` **`sip-ua`** | Enters the SIP user agent configuration mode. |
| **Step 2** | `Router(config-sip-ua)#` **`set pstn-cause`** *`value`* **`sip-status`** *`value`* | Sets a PSTN cause code to correspond with a SIP status code.<br>**pstn-cause** *value*: Sets the value of the PSTN cause code. Range is 1-127.<br>**sip-status** *value:* Sets the value of the SIP status code. Range is 400-699. |
| **Step 3** | `Router(config-sip-ua)#` **`exit`** | Exits the SIP user agent configuration mode. |

## Mapping SIP Status Codes to PSTN Cause Codes

To configure an incoming SIP status code to a PSTN cause code, complete the following steps beginning in global configuration mode:

| | Command | Purpose |
|---|---|---|
| **Step 1** | `Router(config)#` **`sip-ua`** | Enters the SIP user agent configuration mode. |
| **Step 2** | `Router(config-sip-ua)#` **`set sip-status`** *`value`* **`pstn-cause`** *`value`* | Sets a SIP error status code to a PSTN cause code.<br>**sip-status** *value:* Sets the value of the SIP status code. Range is 400-699.<br>**pstn-cause** *value*: Sets the value of the PSTN cause code. Range is 1-127. |
| **Step 3** | `Router(config-sip-ua)#` **`exit`** | Exits the SIP user agent configuration mode. |

# Verifying PSTN and SIP Code Mapping

## Verifying PSTN Cause Code to SIP Status Code Mapping

The following example shows sample output for the **show sip-ua map** command:

```
Router# show sip-ua map pstn-sip
PSTN-Cause   Configured       Default
             SIP-Status      SIP-Status
1               404             404
2               404             404
3               404             404
4               500             500
5               500             500
6               500             500
7               500             500
8               500             500
9               500             500
15              500             500
16              500             500
17              486             486
18              480             480
19              480             480
20              480             480
21              403             403
22              410             410
26              404             404
27              404             404
28              484             484
29              501             501
30              500             500
31              404             404
34              503             503
35              500             500
36              500             500
37              500             500
38              503             503
39              500             500
40              500             500
41              503             503
42              503             503
43              500             500
44              500             500
45              500             500
46              500             500
47              503             503
49              500             500
50              500             500
53              500             500
55              403             403
57              403             403
58              501             501
62              500             500
63              500             500
65              501             501
66              500             500
69              500             500
70              500             500
79              501             501
81              500             500
82              500             500
83              500             500
```

```
84                500               500
85                500               500
86                500               500
87                503               503
88                400               400
90                500               500
91                500               500
93                500               500
95                400               400
96                500               500
97                500               500
98                500               500
99                500               500
100               500               500
101               500               500
102               408               408
103               500               500
110               500               500
111               400               400
126               500               500
127               500               500
```

## Verifying SIP Cause Code to PSTN Status Code Mapping

The following example shows sample output for the **show sip-ua map** command:

```
Router# show sip-ua map sip-pstn
SIP-Status   Configured       Default
             PSTN-Cause      PSTN-Cause
400             127             127
401             57              57
402             21              21
403             57              57
404             1               1
405             127             127
406             127             127
407             21              21
408             102             102
409             41              41
410             1               1
411             127             127
413             127             127
414             127             127
415             79              79
420             127             127
480             18              18
481             127             127
482             127             127
483             127             127
484             28              28
485             1               1
486             17              17
487             127             127
488             127             127
500             41              41
501             79              79
502             38              38
503             63              63
504             102             102
505             127             127
580             47              47
600             17              17
603             21              21
604             1               1
606             58              58
```

# Troubleshooting Tips

Use the **debug ccsip all** command to enable all SIP debugging capabilities, or use one of the following SIP debug commands:

- **debug ccsip calls**
- **debug ccsip error**
- **debug ccsip events**
- **debug ccsip messages**
- **debug ccsip states**

# Configuration Examples

This section shows the two commands that change the standard mappings between the SIP and PSTN networks. The commands **set sip-status** and **set pstn-cause** are highlighed in the following configuration.

```
Router# show running config
Building configuration...

Current configuration : 1564 bytes
!
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname 3660-1
!
!
clock timezone GMT 0
voice-card 1
!
ip subnet-zero
!
!
ip domain-name sip.com
ip name-server 10.10.1.8
!
isdn switch-type primary-5ess
!
!
voice service voip
 sip
!
!
!
!
no voice hpi capture buffer
no voice hpi capture destination
!
fax interface-type fax-mail
mta receive maximum-recipients 0
!
controller T1 1/0
 framing esf
 linecode b8zs
 ds0-group 0 timeslots 1-24 type e&m-wink-start
 ds0 busyout 2-24
!
controller T1 1/1
 framing sf
 linecode ami
!
!
!
!
interface FastEthernet0/0
 no ip address
 shutdown
 duplex auto
 speed auto
!
```

```
interface FastEthernet0/1
 ip address 10.10.1.3 255.255.255.0
 duplex auto
 speed auto
 ip rsvp bandwidth 75000 75000
!
ip classless
ip route 0.0.0.0 0.0.0.0 FastEthernet0/1
ip http server
ip pim bidir-enable
!
!
!
!
call rsvp-sync
!
voice-port 1/0:0
 output attenuation 3
!
voice-port 2/0/0
!
voice-port 2/0/1
!
voice-port 2/1/0
!
voice-port 2/1/1
!
!
mgcp profile default
!
dial-peer cor custom
!
!
!
dial-peer voice 3640110 voip
 application session
 incoming called-number 3640110
 destination-pattern 3640110
 rtp payload-type nte 102
 session protocol sipv2
 session target ipv4:10.10.1.4
 dtmf-relay rtp-nte
 codec g711ulaw
!
dial-peer voice 3660110 pots
 application session
 destination-pattern 3660110
 port 2/0/0
!
sip-ua
 set sip-status 486 pstn-cause 34
 set pstn-cause 17 sip-status 503
 no oli
!
!
line con 0
 exec-timeout 0 0
line aux 0
line vty 0 4
 login
!
!
end
```

# Command Reference

This section documents new commands. All other commands used with this feature are documented in the Cisco IOS Release 12.2 command reference publications.

### New Commands

- **set pstn-cause**
- **set sip-status**
- **show sip-ua map**

# set pstn-cause

To map an incoming PSTN cause code to a SIP error status code, use the **set pstn-cause** command in SIP user agent configuration mode. To achieve default capabilities, use the **no** form of this command.

**set pstn-cause** *value* **sip-status** *value*

**no set pstn-cause**

## Syntax Description

| | |
|---|---|
| **pstn-cause** *value* | Sets the value of the PSTN cause code. The valid range is from 1 to 127. |
| **sip-status** *value* | Sets the value of the SIP status code that is to correspond with the PSTN cause code. The valid range of the SIP status code is from 400 to 699. |

## Defaults

The default mappings defined in the following table are used:

***Table 4 Default PSTN Cause Code to SIP Event Mappings***

| PSTN Cause Code | Description | SIP Event |
|---|---|---|
| 1 | Unallocated number | 404 Not found |
| 2 | No route to specified transit network | 404 Not found |
| 3 | No route to destination | 404 Not found |
| 17 | User busy | 486 Busy here |
| 18 | No user responding | 480 Temporarily unavailable |
| 19 | No answer from the user | |
| 20 | Subscriber absent | |
| 21 | Call rejected | 403 Forbidden |
| 22 | Number changed | 410 Gone |
| 26 | Non-selected user clearing | 404 Not found |
| 27 | Destination out of order | 404 Not found |
| 28 | Address incomplete | 484 Address incomplete |
| 29 | Facility rejected | 501 Not implemented |
| 31 | Normal, unspecified | 404 Not found |
| 34 | No circuit available | 503 Service unavailable |
| 38 | Network out of order | 503 Service unavailable |
| 41 | Temporary failure | 503 Service unavailable |
| 42 | Switching equipment congestion | 503 Service unavailable |
| 47 | Resource unavailable | 503 Service unavailable |
| 55 | Incoming class barred within the Closed User Group (CUG) | 403 Forbidden |
| 57 | Bearer capability not authorized | 403 Forbidden |

*Table 4 Default PSTN Cause Code to SIP Event Mappings (continued)*

| PSTN Cause Code | Description | SIP Event |
|---|---|---|
| 58 | Bearer capability not presently available | 501 Not implemented |
| 65 | Bearer capability not implemented | 501 Not implemented |
| 79 | Service or option not implemented | 501 Not implemented |
| 87 | User not member of the Closed User Group (CUG) | 503 Service Unavailable |
| 88 | Incompatible destination | 400 Bad request |
| 95 | Invalid message | 400 Bad request |
| 102 | Recover on Expires timeout | 408 Request timeout |
| 111 | Protocol error | 400 Bad request |
| **Any code other than those listed above:** | | **500 Internal server error** |

## Command Modes

SIP user agent configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(2)XB | This command was introduced. |
| 12.2(2)XB2 | This command was implemented on the Cisco AS5850 platform. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T. The Cisco AS5300, Cisco AS5350, Cisco AS5850, and Cisco AS5400 platforms were not supported in this release. |
| 12.2(11)T | This command was integrated into Cisco IOS Release 12.2(11)T and support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms. |

## Usage Guidelines

A PSTN cause code can be mapped only to one SIP status code at a time.

## Examples

The following example shows how to map a SIP status code to correspond with a PSTN cause code:

```
Router(config)# sip-ua
Router(config-sip-ua)# set pstn-cause 111 sip-status 400
Router(config-sip-ua)# exit
```

## Related Commands

| Command | Description |
|---|---|
| **set sip-status** | Sets an incoming SIP error status code to a PSTN release cause code. |

# set sip-status

To map an incoming SIP error status code to a PSTN cause code, use the **set sip-status** command in SIP user agent configuration mode. To achieve default capabilities, use the **no** form of this command.

**set sip-status** *value* **pstn-cause** *value*

**no set sip-status**

## Syntax Description

| | |
|---|---|
| **sip-status** *value* | Sets the value of the SIP status code. The valid range is from 400 to 699. |
| **pstn-cause** *value* | Sets the value of the PSTN cause code that is to correspond with the SIP status code. The valid range of the PSTN cause code is from 1 to 127. |

## Defaults

The default mappings defined in the following table are used:

*Table 5 Default SIP Event to PSTN Cause Code Mapping*

| SIP Event | PSTN Cause Code | Description |
|---|---|---|
| 400 Bad request | 127 | Interworking, unspecified |
| 401 Unauthorized | 57 | Bearer capability not authorized |
| 402 Payment required | 21 | Call rejected |
| 403 Forbidden | 57 | Bearer capability not authorized |
| 404 Not found | 1 | Unallocated number |
| 405 Method not allowed | 127 | Interworking, unspecified |
| 406 Not acceptable | | |
| 407 Proxy authentication required | 21 | Call rejected |
| 408 Request timeout | 102 | Recover on Expires timeout |
| 409 Conflict | 41 | Temporary failure |
| 410 Gone | 1 | Unallocated number |
| 411 Length required | 127 | Interworking, unspecified |
| 413 Request entity too long | | |
| 414 Request URI (URL) too long | | |
| 415 Unsupported media type | 79 | Service or option not available |
| 420 Bad extension | 127 | Interworking, unspecified |
| 480 Temporarily unavailable | 18 | No user response |
| 481 Call leg does not exist | 127 | Interworking, unspecified |
| 482 Loop detected | | |
| 483 Too many hops | | |
| 484 Address incomplete | 28 | Address incomplete |
| 485 Address ambiguous | 1 | Unallocated number |

*Table 5 Default SIP Event to PSTN Cause Code Mapping (continued)*

| SIP Event | PSTN Cause Code | Description |
|---|---|---|
| 486 Busy here | 17 | User busy |
| 487 Request cancelled | 127 | Interworking, unspecified |
| 488 Not acceptable here | 127 | Interworking, unspecified |
| 500 Internal server error | 41 | Temporary failure |
| 501 Not implemented | 79 | Service or option not implemented |
| 502 Bad gateway | 38 | Network out of order |
| 503 Service unavailable | 63 | Service or option unavailable |
| 504 Gateway timeout | 102 | Recover on Expires timeout |
| 505 Version not implemented | 127 | Interworking, unspecified |
| 580 Precondition Failed | 47 | Resource unavailable, unspecified |
| 600 Busy everywhere | 17 | User busy |
| 603 Decline | 21 | Call rejected |
| 604 Does not exist anywhere | 1 | Unallocated number |
| 606 Not acceptable | 58 | Bearer capability not presently available |

## Command Modes

SIP user agent configuration

## Command History

| Release | Modification |
|---|---|
| 12.2(2)XB | This command was introduced. |
| 12.2(2)XB2 | This command was implemented on the Cisco AS5850 platform. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T. The Cisco AS5300, Cisco AS5350, Cisco AS5850, and Cisco AS5400 platforms were not supported in this release. |
| 12.2(11)T | This command was integrated into Cisco IOS Release 12.2(11)T and support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms. |

## Usage Guidelines

A SIP status code can be mapped to many PSTN cause codes. For example, 503 can be mapped to 34, 38, and 58.

## Examples

The following example shows how to map a PSTN cause code to correspond with a SIP status code:

```
Router(config)# sip-ua
Router(config-sip-ua)# set sip-status 400 pstn-cause 16
```

## Related Commands

| Command | Description |
|---|---|
| **set pstn-cause** | Sets an incoming PSTN cause code to a SIP error status code. |

# show sip-ua map

To display the mapping table showing PSTN cause codes and their corresponding SIP error status codes or the mapping table showing SIP-to-PSTN codes, use the **show sip-ua map** command in privileged EXEC mode.

**show sip-ua map** {**pstn-sip** | **sip-pstn**}

## Syntax Description

| | |
|---|---|
| **pstn-sip** | Displays PSTN cause code to SIP status code mapping table. |
| **sip-pstn** | Displays SIP status code to PSTN cause code mapping table. |

## Defaults

No default behavior or values.

## Command Modes

Privileged EXEC

## Command History

| Release | Modification |
|---|---|
| 12.2(2)XB | This command was introduced. |
| 12.2(2)XB2 | This command was implemented on the Cisco AS5850 platform. |
| 12.2(8)T | This command was integrated into Cisco IOS Release 12.2(8)T. The Cisco AS5300, Cisco AS5350, Cisco AS5850, and Cisco AS5400 platforms were not supported in this release. |
| 12.2(11)T | This command was integrated into Cisco IOS Release 12.2(11)T and support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms. |

## Examples

The following is a sample output from the **show sip-ua map** command:

```
Router# show sip-ua map pstn-sip
PSTN-Cause   Configured       Default
             SIP-Status      SIP-Status
1               404             404
2               404             404
3               404             404
4               500             500
5               500             500
6               500             500
7               500             500
8               500             500
9               500             500
15              500             500
16              500             500
17              486             486
18              480             480
19              480             480
20              480             480
21              403             403
22              410             410
```

```
26              404             404
27              404             404
28              484             484
29              501             501
30              500             500
31              404             404
34              503             503
35              500             500
36              500             500
37              500             500
38              503             503
39              500             500
40              500             500
41              503             503
42              503             503
43              500             500
44              500             500
45              500             500
46              500             500
47              503             503
49              500             500
50              500             500
53              500             500
55              403             403
57              403             403
58              501             501
62              500             500
63              500             500
65              501             501
66              500             500
69              500             500
70              500             500
79              501             501
81              500             500
82              500             500
83              500             500
84              500             500
85              500             500
86              500             500
87              503             503
88              400             400
90              500             500
91              500             500
93              500             500
95              400             400
96              500             500
97              500             500
98              500             500
99              500             500
100             500             500
101             500             500
102             408             408
103             500             500
110             500             500
111             400             400
126             500             500
127             500             500
```

*Table 6 show sip-ua map pstn-sip output*

| Field | Description |
|---|---|
| PSTN Cause | Indicates the reasons for PSTN call failure or completion.<br>The PSTN cause code range is from 1-127. |
| Configured SIP Status | Indicates the configured SIP status code or event.<br>The SIP Status code range is from 400-699. |
| Default SIP Status | Indicates the default mapping between and PSTN and SIP network. |

The following example shows the mapped **sip-pstn** settings:

```
Router# show sip-ua map sip-pstn
SIP-Status   Configured        Default
             PSTN-Cause      PSTN-Cause
400             127             127
401             57              57
402             21              21
403             57              57
404             1               1
405             127             127
406             127             127
407             21              21
408             102             102
409             41              41
410             1               1
411             127             127
413             127             127
414             127             127
415             79              79
420             127             127
480             18              18
481             127             127
482             127             127
483             127             127
484             28              28
485             1               1
486             17              17
487             127             127
488             127             127
500             41              41
501             79              79
502             38              38
503             63              63
504             102             102
505             127             127
580             47              47
600             17              17
603             21              21
604             1               1
606             58              58
```

*Table 7 show sip-ua map sip-pstn output*

| Field | Description |
|---|---|
| SIP Status | Indicates the configured SIP status code or event.<br>The SIP status code range is from 400-699. |
| Configured PSTN Cause | Indicates the reasons for PSTN call failure or completion.<br>The PSTN cause code range is from 1-127. |
| Default PSTN Cause | Indicates the default mapping between and SIP and PSTN network. |

## Related Commands

| Command | Description |
|---|---|
| **set sip-status** | Sets an incoming SIP error status code to a PSTN release cause code. |
| **set pstn-cause** | Sets an incoming PSTN release cause code to a SIP error status code. |

# Glossary

**call**—In SIP, a call consists of all participants in a conference invited by a common source. A SIP call is identified by a globally unique call identifier. A point-to-point IP telephony conversation maps into a single SIP call.

**cause code**—Defined by ITU Series Q Recommendation 850. Indicates the reason for PSTN call failure or completion.

**CLI**—command-line interface.

**dial peer**—An addressable call endpoint.

**endpoint—**A terminal or gateway that acts as a source or sink of voice data. An endpoint can call or be called, and it generates or terminates the information stream.

**gateway**—A gateway allows SIP or H.323 terminals to communicate with terminals configured to other protocols by converting protocols. A gateway is the point where a circuit-switched call is encoded and repackaged into IP packets.

**INVITE**—A method that initiates a session. It indicates that a user is invited to participate, provides a session description, indicates the type of media, and provides insight regarding the capabilities of the called and calling parties.

**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. Basic telephone service supplying standard single-line telephones, telephone lines, and access to the PSTN.

**proxy server—**An intermediary program that acts as both a server and a client for the purpose of making requests on behalf of other clients. Requests are serviced internally or by passing them on, possibly after translation, to other servers. A proxy interprets and, if necessary, rewrites a request message before forwarding it.

**PSTN**—Public Switched Telephone Network. PSTN refers to the local telephone company.

**session—**A SIP session is a set of multimedia senders and receivers and the data streams flowing between the senders and receivers. A SIP multimedia conference is an example of a session. The called party can be invited several times by different calls to the same session.

**SIP**—Session Initiation Protocol. An application-layer protocol originally developed by the Multiparty Multimedia Session Control (MMUSIC) working group of the Internet Engineering Task Force (IETF). Their goal was to equip platforms to signal the setup of voice and multimedia calls over IP networks. SIP features are compliant with IETF RFC 2543, published in March 1999.

**SPI**—service provider interface.

**user agent—**A combination of UAS and UAC that initiates and receives calls. See **UAS** and **UAC**.

**UAC—**user agent client. A client application that initiates a SIP request.

**UAS—**user agent server (or user agent). A server application that contacts the user when a SIP request is received, then returns a response on behalf of the user. The response accepts, rejects or redirects the request.

**URL**—Universal Resource Locator. Standard address of any resource on the Internet that is part of the World Wide Web (WWW).

**VoIP—**Voice over IP. The ability to carry normal telephone-style voice over an IP-based Internet with POTS-like functionality, reliability, and voice quality. VoIP is a blanket term that generally refers to the Cisco standards-based approach (for example, H.323) to IP voice traffic.