



SIP: CLI for Caller ID When Privacy Exists

The SIP: CLI for Caller ID When Privacy Exists feature adds three command-line interface (CLI) options that make the handling of caller ID information more flexible. Specifically, the SIP: CLI for Caller ID When Privacy Exists feature addresses the following situations:

- Passing along caller ID information when privacy exists
- Handling the Display Name field when no display name exists
- Allowing caller ID information to be passed to ISDN as network-provided

History for the SIP: CLI for Caller ID When Privacy Exists Feature

Release	Modification
12.4(4)T	This feature was introduced.

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Prerequisites for SIP: CLI for Caller ID When Privacy Exists

- Establish a working IP network.
- Configure VoIP.
- Ensure that the gateway has voice functionality configured for SIP.

**Note**

For information about configuring voice functionality, see the [Cisco IOS Voice Configuration Library](#).

Information About SIP: CLI for Caller ID When Privacy Exists

The SIP: CLI for Caller ID When Privacy Exists feature is comprised of three main components, as follows:

- [SIP: Caller ID Removable to Improve Privacy, page 2](#)
- [SIP: Calling Number Substitution for the Display Name When the Display Name is Unavailable, page 3](#)
- [SIP: Calling Number Passing as Network-Provided or User-Provided, page 4](#)

SIP: Caller ID Removable to Improve Privacy

The caller ID information is passed through from the ISDN-to-SIP by copying the number in the Calling Party Number information element (IE) in an ISDN Setup message into the Calling Number field of the SIP Remote-Party-ID and From headers.

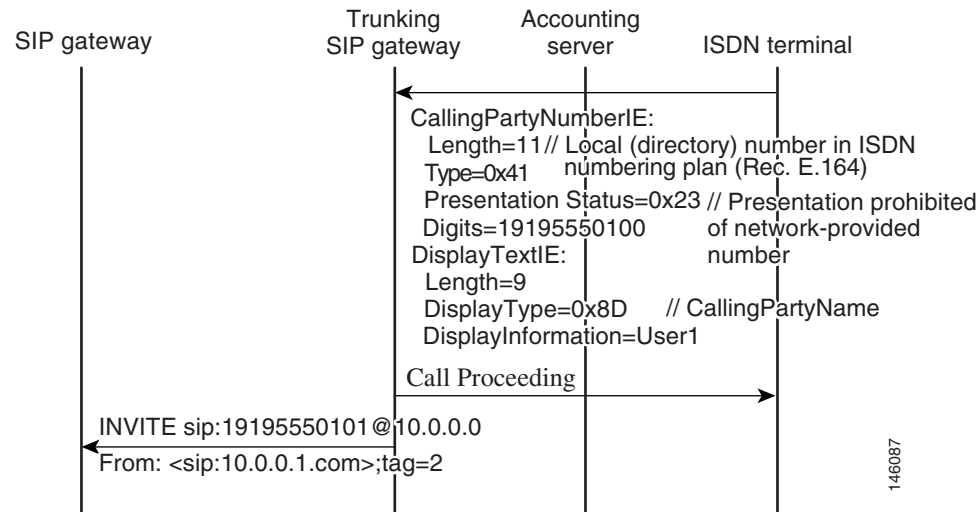
The Calling Name from the ISDN Display IE is copied into the SIP Display Name field in the SIP Remote-Party-ID and From headers. The Calling Party Number IE contains a Presentation Indicator field that is set to presentation allowed, presentation restricted, number not available due to interworking, or reserved. Presentation allowed and presentation restricted are translated into privacy set to off or privacy set to null, respectively, in the SIP Remote-Party-ID header field.

However, for added privacy, the SIP: CLI for Caller ID When Privacy Exists feature introduces CLI to completely remove the Calling Number and Display Name from an outgoing message's From header if presentation is prohibited. This prohibits sending the SIP Remote Party ID header, because the Cisco gateway does not send SIP Remote-Party ID headers without both a Display Name and Calling Number.

**Note**

The SIP: Caller ID Removable to Improve Privacy option is available both globally and at the dial-peer level.

See [Figure 1](#) for call flows and [Table 1](#) and [Table 2](#) for additional presentation mapping.

Figure 1 Call Flow for Blocking Caller ID Information When Privacy Exists**Table 1** Presentation to Privacy Mapping with CLI Disabled

Presentation Indicator	From Remote Party ID (RPID)
Presentation Allowed	From: "User1" <sip:19195550100@10.0.0.0>;tag=1 Remote-Party-ID: "User1" <sip:19195550100@10.0.0.0>;party=calling;privacy=off
Presentation Prohibited	From: "User1" <sip:19195550100@10.0.0.0>;tag=1 Remote-Party-ID: "User1"<sip:19195550100@10.0.0.0>;party=calling;privacy=full

Table 2 Presentation to Privacy Mapping with CLI Enabled

Presentation Indicator	From RPID
Presentation Allowed	From: "User1" <sip:19195550100@10.0.0.0>;tag=1 Remote-Party-ID: "User1"<sip:19195550100@10.0.0.0>;party=calling;privacy=off
Presentation Prohibited	From: <sip:10.0.0.0>;tag=1 Remote Party ID not sent

SIP: Calling Number Substitution for the Display Name When the Display Name is Unavailable

When the Display information element (IE) in a PSTN-to-SIP call is not available with a Setup message, the Cisco gateway leaves the Display Name field in the SIP Remote-Party-ID and From headers blank.

When presentation is allowed, the SIP: CLI for Caller ID When Privacy Exists feature enables the substitution of the Calling Number for the missing Display Name in the SIP Remote-Party-ID and From headers. Upon receipt of a Setup message where a name to follow is indicated, the Calling Number is not copied into the Display Name.

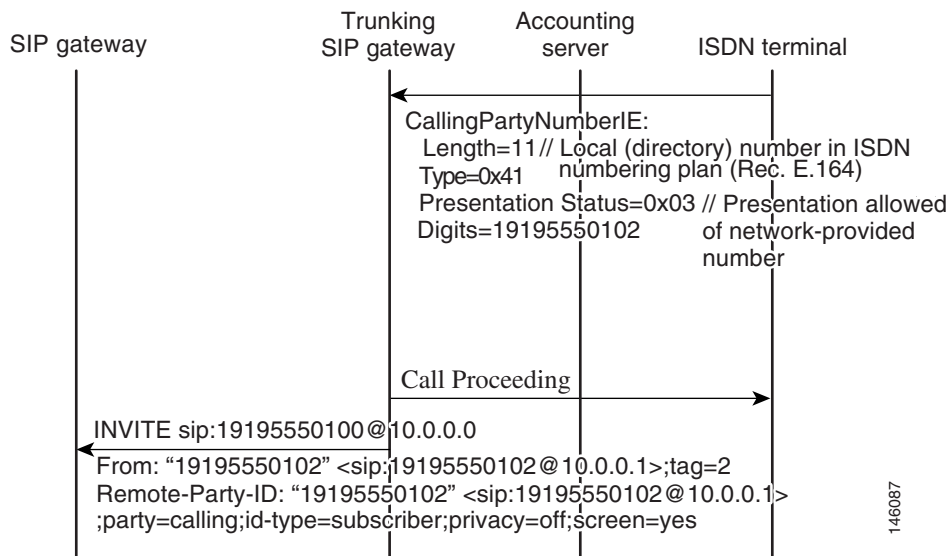
Also, the SIP Extensions for Caller Identity and Privacy on SIP gateway feature added the ability to hardcode calling name and number in the SIP Remote-Party-ID and From headers. The SIP Extensions for Caller Identify and Privacy feature settings take precedence over the SIP: CLI for Caller ID When Privacy Exists feature settings.

**Note**

The SIP: Calling Number Substitutions for the Display Name When the Display Name is Unavailable option is available both globally and at the dial-peer level.

See [Figure 2](#) for the call flow where the Calling Number is substituted for the Display Number.

Figure 2 *Call Flow for Substituting the Calling Number for the Display Name When the Display Name is Unavailable*



SIP: Calling Number Passing as Network-Provided or User-Provided

ISDN numbers can be passed along as network-provided or user-provided in an ISDN Calling Party information element (IE) Screening Indicator field. The Cisco gateway automatically sets the Screening Indicator to user-provided in SIP-to-ISDN calls.

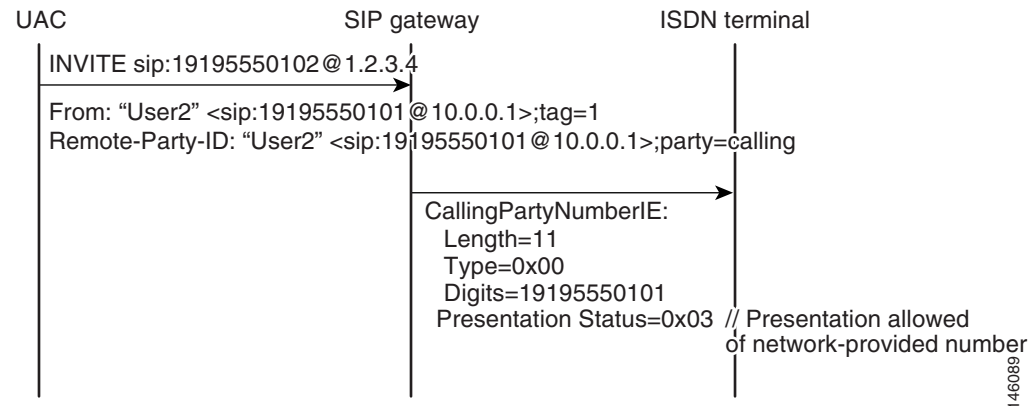
The SIP: CLI for Caller ID When Privacy Exists feature allows toggling between user-provided and network-provided ISDN numbers for the screening indicator. Therefore, after bits 1 and 2 are set to reflect network-provided, any existing screening information is lost. However, presentation information in bits 6 and 7 is preserved.

**Note**

The SIP: Calling Number Passing as Network-Provided or User-Provided option is available both globally and at the dial-peer level.

See [Figure 3](#) for the call flow when the calling number is passed along as network-provided.

Figure 3 *Call Flow for Passing Through the Calling Number as Network-Provided*



How to Configure SIP: CLI for Caller ID When Privacy Exists

This section contains the following procedures:

- [Configuring SIP: Blocking Caller ID Information Globally When Privacy Exists, page 5](#) (optional)
- [Configuring Dial-Peer Level SIP: Blocking of Caller ID Information When Privacy Exists, page 7](#) (optional)
- [Configuring Globally the SIP: Calling Number for Display Name Substitution When Display Name Is Unavailable, page 7](#) (optional)
- [Configuring Dial-Peer-Level SIP: Substitution of the Calling Number for Display Name When the Display Name Is Unavailable, page 8](#) (optional)
- [Configuring Globally the SIP: Pass-Through of the Passing Calling Number as Network-Provided, page 9](#) (optional)
- [Configuring at the Dial-Peer Level the SIP: Pass-Through of Passing the Calling Number as Network-Provided, page 10](#) (optional)
- [Configuring Globally the SIP: Pass-Through of the Passing Calling Number as User-Provided, page 11](#) (optional)
- [Configuring at the Dial-Peer Level the SIP: Pass-Through of Passing the Calling Number as User-Provided, page 12](#) (optional)

Configuring SIP: Blocking Caller ID Information Globally When Privacy Exists

The Call-ID information is private information. In ISDN there is a private setting that can be set to protect this information. However, whenever SIP gets the Call-ID information, it does not hide the private information, rather, it just sets a field to reflect that it is private and not to display it on a Call-ID display. But, the data is still viewable in the SIP message requests. This option allows the Cisco gateway to delete the Call-ID information from the SIP message requests so it cannot be read on the network.

Upon receiving an ISDN Setup message with the calling-party information element, the Cisco gateway translates the presentation indicator to set privacy to full for restricted presentation or to set privacy to off for unrestricted presentation in the Remote-Party-ID header field. The SIP: CLI for Caller ID When Privacy Exists feature introduces a CLI switch that either allows stripping the Calling Number and Display Name from the From and Remote-Party-ID fields in the SIP message requests or passes on the information. However, in cases of unrestricted presentation, the gateway passes the caller ID information, regardless of the CLI setting.

The global commands to strip the Calling Name and Calling Number from the Remote-Party-ID and From headers are as follows:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **clid strip pi-restrict all**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice-service-VoIP configuration mode.
Step 4	clid strip pi-restrict all Example: Router(config-voip-serv)# clid strip pi-restrict all	Enters block call ID information when privacy exists in global configuration mode.
Step 5	exit Example: Router(config-voip-serv)# exit	Exits the current mode.

Configuring Dial-Peer Level SIP: Blocking of Caller ID Information When Privacy Exists

The dial-peer specific command to strip the Calling Number from the Remote-Party-ID and From headers is as follows:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *dial-peer-number* **voip**
4. **clid strip pi-restrict all**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice <i>dial-peer-number</i> voip Example: Router(config)# dial-peer voice 100 voip	Enters dial-peer configuration mode.
Step 4	clid strip pi-restrict all Example: Router(config-dial-peer)# clid strip pi-restrict all	Enters block call ID information when privacy exists in dial-peer configuration mode.
Step 5	exit Example: Router# exit	Exits the current mode.

Configuring Globally the SIP: Calling Number for Display Name Substitution When Display Name Is Unavailable

When this is enabled, if there is no Display Name field but there is a number, it copies the number into the Display Name field, so the number is displayed on the recipient's Call-ID display.

The Cisco gateway omits the Display Name field if no display information is received. This feature also introduces a CLI switch that allows the Calling Number to be copied into the Display Name field, as long as presentation is not prohibited.

The steps for substituting the Calling Number for the Display Name when it is unavailable in the Remote-Party-ID and From headers are as follows:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **clid substitute name**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice-service-VoIP configuration mode.
Step 4	clid substitute name Example: Router(config-voip-serv)# clid substitute name	Substitutes the calling number for the display name when the display name is unavailable in the global configuration mode.
Step 5	exit Example: Router(config-voip-serv)# exit	Exits the current mode.

Configuring Dial-Peer-Level SIP: Substitution of the Calling Number for Display Name When the Display Name Is Unavailable

The dial-peer-specific steps for substituting the Calling Number for the Display Name when it is unavailable in the Remote-Party-ID and From headers are as follows:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *dial-peer-number* **voip**
4. **clid substitute name**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice <i>dial-peer-number</i> voip Example: Router(config-dial-peer)# dial-peer voice 100 voip	Enters dial-peer configuration mode.
Step 4	clid substitute name Example: Router(config-dial-peer)# clid substitute name	Substitutes the calling number for the display name when the display name is unavailable in dial-peer configuration mode.
Step 5	exit Example: Router(config-dial-peer)# exit	Exits the current mode.

Configuring Globally the SIP: Pass-Through of the Passing Calling Number as Network-Provided

This field shows whether the Call-ID information was supplied by the network or not. This is for screening purposes.

Formerly the Calling Number from the session initiation protocol to public switched telephone network (SIP-to-PSTN) was always translated to user-provided. This feature introduces a CLI switch to toggle between branding numbers as user-provided or network-provided.

The steps for globally setting set the Screening Indicator to network-provided are as follows:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **clid network-provided**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice-service-VoIP configuration mode.
Step 4	clid network-provided Example: Router(config-voip-serv)# clid network-provided	Enters the network-provided calling number in voice-service-VoIP configuration mode.
Step 5	exit Example: Router(config-voip-serv)# exit	Exits the current mode.

Configuring at the Dial-Peer Level the SIP: Pass-Through of Passing the Calling Number as Network-Provided

The dial-peer specific command to set the Screening Indicator to network-provided is as follows:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice *dial-peer-number* voip**
4. **clid network-provided**

5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice <i>dial-peer-number</i> voip Example: Router(config)# dial-peer voice 100 voip	Enters dial-peer configuration mode.
Step 4	clid network-provided Example: Router(config-dial-peer)# clid network-provided	Enters the network-provided calling number in dial-peer configuration mode.
Step 5	exit Example: Router(config-dial-peer)# exit	Exits the current mode.

Configuring Globally the SIP: Pass-Through of the Passing Calling Number as User-Provided

The steps for globally setting set the Screening Indicator to user-provided are as follows:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **no clid network-provided**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice service voip Example: Router(config)# voice service voip	Enters voice-service-VoIP configuration mode.
Step 4	no clid network-provided Example: Router(config-voip-serv)# no clid network-provided	Enters the network-provided calling number in voice-service-VoIP configuration mode.
Step 5	exit Example: Router(config-voip-serv)# exit	Exits the current mode.

Configuring at the Dial-Peer Level the SIP: Pass-Through of Passing the Calling Number as User-Provided

The dial-peer specific command to set the Screening Indicator to user-provided is as follows:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice *dial-peer-number* voip**
4. **no clid network-provided**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice dial-peer-number voip Example: Router(config)# dial-peer voice 100 voip	Enters dial-peer configuration mode.
Step 4	no clid network-provided Example: Router(config-dial-peer)# no clid network-provided	Enters the user-provided calling number in dial-peer configuration mode.
Step 5	exit Example: Router(config-dial-peer)# exit	Exits the current mode.

Configuration Examples for SIP: CLI for Caller ID When Privacy Exists

The following shows an example of the SIP: CLI for Caller ID When Privacy Exists feature when enabled globally and disabled on the dial-peer level:

```
Router# show running-config

Building configuration...
Current configuration: 1234 bytes
!
version 12.4
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
!
hostname pip
!
boot-start-marker
boot system tftp user1/c3660-is-mz 172.18.207.15
boot-end-marker
!
logging buffered 1000000 debugging
enable secret 5 $1$li0u$IkIqPXzKq4uKme.LhzGut0
enable password lab
```

```

!
no aaa new-model
!
resource policy
!
clock timezone GMT 0
clock summer-time EDT recurring
ip subnet-zero
ip tcp path-mtu-discovery
!
ip cef
ip domain name sip.com
ip host sip-server1 172.18.193.100
ip host CALLGEN-SECURITY-V2 10.76.47.38 10.30.0.0
ip name-server 172.18.192.48
no ip dhcp use vrf connected
!
ip vrf btknet
rd 8262:2000
!
voice call send-alert
!
voice service voip <- SIP: CLI for Caller ID When Privacy Exists feature enabled globally
clid substitute name
clid strip pi-restrict all
clid network-provided
sip
!
voice class codec 1
codec preference 1 g729r8
codec preference 2 g711alaw
codec preference 3 g711ulaw
codec preference 4 g729br8
codec preference 5 g726r32
codec preference 6 g726r24
codec preference 7 g726r16
codec preference 8 g723ar53
codec preference 9 g723r53
codec preference 10 g723ar63
codec preference 11 gsmefr
codec preference 12 gsmfr
codec preference 13 g728
!
voice class codec 2
codec preference 1 g729r8
codec preference 2 g711ulaw
codec preference 3 g711alaw
!
voice class codec 99
codec preference 1 g729r8
codec preference 2 g711ulaw
codec preference 3 g711alaw
!
fax interface-type fax-mail
!
interface FastEthernet0/0
ip address 172.18.195.49 255.255.255.0
duplex auto
speed auto
no cdp enable
ip rsvp bandwidth 96 96
!
interface FastEthernet0/1
ip address 172.18.193.190 255.255.255.0

```

```
shutdown
duplex auto
speed auto
no cdp enable
!
no ip http server
!
ip classless
ip route 0.0.0.0 0.0.0.0 FastEthernet0/0
ip route 172.16.0.0 255.0.0.0 172.18.195.1
!
snmp-server community public RO
!
control-plane
!
voice-port 1/0/0
!
voice-port 1/0/1
!
mgcp behavior rsip-range tgcp-only
!
dial-peer cor custom
!
dial-peer voice 100 pots
destination-pattern 9001
!
dial-peer voice 3301 voip
destination-pattern 9002
session protocol sipv2
session target ipv4:172.18.193.87
incoming called-number 9001
codec g711ulaw
no vad
!
dial-peer voice 3303 voip
destination-pattern 777
session protocol sipv2
session target ipv4:172.18.199.94
!
dial-peer voice 36601 voip
destination-pattern 36601
no modem passthrough
session protocol sipv2
session target ipv4:172.18.193.98
!
dial-peer voice 5 voip
destination-pattern 5550100
session protocol sipv2
session target ipv4:172.18.197.182
codec g711ulaw
!
dial-peer voice 36602 voip
destination-pattern 36602
session protocol sipv2
session target ipv4:172.18.193.120
incoming called-number 9001
dtmf-relay rtp-nte
codec g711ulaw
!
dial-peer voice 111 voip
destination-pattern 111
session protocol sipv2
session target ipv4:172.18.193.251
!
```

```

dial-peer voice 5550199 voip <- SIP: CLI for Caller ID When Privacy Exists feature
disabled on dial-peer
destination-pattern 3100801
session protocol sipv2
session target ipv4:10.102.17.208
codec g711ulaw
!
dial-peer voice 333 voip
preference 2
destination-pattern 333
modem passthrough nse codec g711ulaw
voice-class codec 99
session protocol sipv2
session target ipv4:172.18.193.250
dtmf-relay rtp-nte
no vad
!
dial-peer voice 9003 pots
preference 2
destination-pattern 9003
!
dial-peer voice 90032 voip
preference 1
destination-pattern 9003
session protocol sipv2
session target ipv4:172.18.193.97
!
dial-peer voice 1 pots
!
num-exp 5550100 5550199
num-exp 5550199 5550100
gateway
timer receive-rtp 1200
!
sip-ua
srv version 1
retry response 1
!
line con 0
exec-timeout 0 0
line aux 0
line vty 0 4
exec-timeout 0 0
password cisco
login
!
no process cpu extended
no process cpu autoprofile hog
ntp clock-period 17180176
ntp server 172.68.10.150 prefer
!
end

```

The following shows an example of the SIP: CLI for Caller ID When Privacy Exists feature when disabled globally and disabled on the dial-peer level:

```
Router# show running-config
```

```

Building configuration...
Current configuration: 1234 bytes
!
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption

```

```
!
hostname pip
!
boot-start-marker
boot system tftp user1/c3660-is-mz 172.18.207.15
boot-end-marker
!
logging buffered 1000000 debugging
enable secret 5 $1$li0u$IkiqPXzKq4uKme.LhzGut0
enable password lab
!
no aaa new-model
!
resource policy
!
clock timezone GMT 0
clock summer-time EDT recurring
ip subnet-zero
ip tcp path-mtu-discovery
!
ip cef
ip domain name sip.com
ip host sip-server1 172.18.193.100
ip host CALLGEN-SECURITY-V2 10.76.47.38 10.30.0.0
ip name-server 172.18.192.48
no ip dhcp use vrf connected
!
ip vrf btknet
rd 8262:2000
!
voice call send-alert
!
voice service voip <- SIP: CLI for Caller ID When Privacy Exists feature disabled globally
sip
!
voice class codec 1
codec preference 1 g729r8
codec preference 2 g711alaw
codec preference 3 g711ulaw
codec preference 4 g729br8
codec preference 5 g726r32
codec preference 6 g726r24
codec preference 7 g726r16
codec preference 8 g723ar53
codec preference 9 g723r53
codec preference 10 g723ar63
codec preference 11 gsmefr
codec preference 12 gsmfr
codec preference 13 g728
!
voice class codec 2
codec preference 1 g729r8
codec preference 2 g711ulaw
codec preference 3 g711alaw
!
voice class codec 99
codec preference 1 g729r8
codec preference 2 g711ulaw
codec preference 3 g711alaw
!
fax interface-type fax-mail
!
interface FastEthernet0/0
ip address 172.18.195.49 255.255.255.0
```

```

duplex auto
speed auto
no cdp enable
ip rsvp bandwidth 96 96
!
interface FastEthernet0/1
ip address 172.18.193.190 255.255.255.0
shutdown
duplex auto
speed auto
no cdp enable
!
no ip http server
!
ip classless
ip route 0.0.0.0 0.0.0.0 FastEthernet0/0
ip route 172.16.0.0 255.0.0.0 172.18.195.1
!
snmp-server community public RO
!
control-plane
!
voice-port 1/0/0
!
voice-port 1/0/1
!
mgcp behavior rsip-range tgcp-only
!
dial-peer cor custom
!
dial-peer voice 100 pots
destination-pattern 9001
!
dial-peer voice 3301 voip
destination-pattern 9002
session protocol sipv2
session target ipv4:172.18.193.87
incoming called-number 9001
codec g711ulaw
no vad
!
dial-peer voice 3303 voip
destination-pattern 777
session protocol sipv2
session target ipv4:172.18.199.94
!
dial-peer voice 36601 voip
destination-pattern 36601
no modem passthrough
session protocol sipv2
session target ipv4:172.18.193.98
!
dial-peer voice 5 voip
destination-pattern 5550100
session protocol sipv2
session target ipv4:172.18.197.182
codec g711ulaw
!
dial-peer voice 36602 voip
destination-pattern 36602
session protocol sipv2
session target ipv4:172.18.193.120
incoming called-number 9001
dtmf-relay rtp-nte

```

```
codec g711ulaw
!
dial-peer voice 111 voip
destination-pattern 111
session protocol sipv2
session target ipv4:172.18.193.251
!
dial-peer voice 5550199 voip <- SIP: CLI for Caller ID When Privacy Exists feature
disabled on dial-peer
destination-pattern 5550199
session protocol sipv2
session target ipv4:10.102.17.208
codec g711ulaw
!
dial-peer voice 333 voip
preference 2
destination-pattern 333
modem passthrough nse codec g711ulaw
voice-class codec 99
session protocol sipv2
session target ipv4:172.18.193.250
dtmf-relay rtp-nte
no vad
!
dial-peer voice 9003 pots
preference 2
destination-pattern 9003
!
dial-peer voice 90032 voip
preference 1
destination-pattern 9003
session protocol sipv2
session target ipv4:172.18.193.97
!
dial-peer voice 1 pots
!
num-exp 5550100 5550199
num-exp 5550101 5550198
gateway
timer receive-rtp 1200
!
sip-ua
srv version 1
retry response 1
!
!
line con 0
exec-timeout 0 0
line aux 0
line vty 0 4
exec-timeout 0 0
password cisco
login
!
no process cpu extended
no process cpu autopprofile hog
ntp clock-period 17180176
ntp server 171.68.10.150 prefer
!
end
```

The following shows an example of the SIP: CLI for Caller ID When Privacy Exists feature when disabled globally and enabled on the dial-peer level:

```

Router# show running-config

Building configuration...
Current configuration: 1234 bytes
!
version 12.4
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
no service password-encryption
!
hostname pip
!
boot-start-marker
boot system tftp judyg/c3660-is-mz 172.18.207.15
boot-end-marker
!
logging buffered 1000000 debugging
enable secret 5 $1$li0u$IkiqPXzKq4uKme.LhzGut0
enable password lab
!
no aaa new-model
!
resource policy
!
clock timezone GMT 0
clock summer-time EDT recurring
ip subnet-zero
ip tcp path-mtu-discovery
!
ip cef
ip domain name sip.com
ip host sip-server1 172.18.193.100
ip host CALLGEN-SECURITY-V2 10.76.47.38 10.30.0.0
ip name-server 172.18.192.48
no ip dhcp use vrf connected
!
ip vrf btknet
rd 8262:2000
!
voice call send-alert
!
voice service voip <- SIP: CLI for Caller ID When Privacy Exists feature disabled globally
sip
!
voice class codec 1
codec preference 1 g729r8
codec preference 2 g711alaw
codec preference 3 g711ulaw
codec preference 4 g729br8
codec preference 5 g726r32
codec preference 6 g726r24
codec preference 7 g726r16
codec preference 8 g723ar53
codec preference 9 g723r53
codec preference 10 g723ar63
codec preference 11 gsmefr
codec preference 12 gsmfr
codec preference 13 g728
!
voice class codec 2
codec preference 1 g729r8
codec preference 2 g711ulaw
codec preference 3 g711alaw
!

```

```
voice class codec 99
codec preference 1 g729r8
codec preference 2 g711ulaw
codec preference 3 g711alaw
!
fax interface-type fax-mail
!
interface FastEthernet0/0
ip address 172.18.195.49 255.255.255.0
duplex auto
speed auto
no cdp enable
ip rsvp bandwidth 96 96
!
interface FastEthernet0/1
ip address 172.18.193.190 255.255.255.0
shutdown
duplex auto
speed auto
no cdp enable
!
no ip http server
!
ip classless
ip route 0.0.0.0 0.0.0.0 FastEthernet0/0
ip route 172.16.0.0 255.0.0.0 172.18.195.1
!
snmp-server community public RO
!
control-plane
!
voice-port 1/0/0
!
voice-port 1/0/1
!
mgcp behavior rsip-range tgcp-only
!
dial-peer cor custom
!
dial-peer voice 100 pots
destination-pattern 9001
!
dial-peer voice 3301 voip
destination-pattern 9002
session protocol sipv2
session target ipv4:172.18.193.87
incoming called-number 9001
codec g711ulaw
no vad
!
dial-peer voice 3303 voip
destination-pattern 777
session protocol sipv2
session target ipv4:172.18.199.94
!
dial-peer voice 36601 voip
destination-pattern 36601
no modem passthrough
session protocol sipv2
session target ipv4:172.18.193.98
!
dial-peer voice 5 voip
destination-pattern 5550102
session protocol sipv2
```

```

session target ipv4:172.18.197.182
codec g711ulaw
!
dial-peer voice 36602 voip
destination-pattern 36602
session protocol sipv2
session target ipv4:172.18.193.120
incoming called-number 9001
dtmf-relay rtp-nte
codec g711ulaw
!
dial-peer voice 111 voip
destination-pattern 111
session protocol sipv2
session target ipv4:172.18.193.251
!
dial-peer voice 5550100 voip <- SIP: CLI for Caller ID When Privacy Exists feature enabled
on dial-peer
destination-pattern 5550100
session protocol sipv2
session target ipv4:10.102.17.208
codec g711ulaw
clid strip pi-restrict all
clid network-provided
clid substitute name
!
dial-peer voice 333 voip
preference 2
destination-pattern 333
modem passthrough nse codec g711ulaw
voice-class codec 99
session protocol sipv2
session target ipv4:172.18.193.250
dtmf-relay rtp-nte
no vad
!
dial-peer voice 9003 pots
preference 2
destination-pattern 9003
!
dial-peer voice 90032 voip
preference 1
destination-pattern 9003
session protocol sipv2
session target ipv4:172.18.193.97
!
dial-peer voice 1 pots
!
num-exp 5550100 5550199
num-exp 5550101 5550198
gateway
timer receive-rtp 1200
!
sip-ua
srv version 1
retry response 1
!
!
line con 0
exec-timeout 0 0
line aux 0
line vty 0 4
exec-timeout 0 0
password cisco

```

```
login
!
no process cpu extended
no process cpu autoprofile hog
ntp clock-period 17180176
ntp server 172.31.10.150 prefer
!
end
```

Additional References

The following sections provide references related to the SIP: CLI for Caller ID When Privacy Exists feature.

Related Documents

Related Topic	Document Title
SIP Configuration Information	Cisco IOS SIP Configuration Guide

Standards

Standard	Title
draft-ietf-sip-privacy-02	SIP Extensions for Caller Identity and Privacy

MIBs

MIB	MIBs Link
No new or modified MIBs are supported by this feature, and support for existing MIBs has not been modified by this feature.	To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

RFCs

RFC	Title
RFC 3261	<i>SIP: Session Initiation Protocol</i>

Technical Assistance

Description	Link
The Cisco Technical Support website contains thousands of pages of searchable technical content, including links to products, technologies, solutions, technical tips, and tools. Registered Cisco.com users can log in from this page to access even more content.	http://www.cisco.com/techsupport

Command Reference

This section documents the following new and modified commands:

- [clid \(dial-peer\)](#), page 25
- [clid \(voice-service-voip\)](#), page 28

clid (dial-peer)

To control the presentation and use of calling-line ID (CLID) information, use the **clid** command in dial-peer configuration mode. To remove CLID controls, use the **no** form of this command.

clid { **network-number** *number* [**second-number strip**] | **network-provided** | **override rdnis** | **restrict** | **strip** [**name** | **pi-restrict** [**all**]] | **substitute name** }

no clid { **network-number** *number* [**second-number strip**] | **network-provided** | **override rdnis** | **restrict** | **strip** [**name** | **pi-restrict** [**all**]] | **substitute name** }

Syntax Description	
network-number <i>number</i>	(Optional) Network number. Establishes the calling-party network number in the CLID for this router.
second-number strip	(Optional) Removes a previously configured second network number from the CLID.
network-provided	(Optional) Allows you to set the screening indicator to reflect the number that was provided by the network.
override rdnis	(Optional; supported for POTS dial peers only) Overrides the CLID with the redirected dialed number identification service (RDNIS) if available.
pi-restrict	(Optional) Restricted progress indicator (PI). Causes removal of the calling-party number from the CLID when the PI is restricted.
restrict	(Optional) Restricts presentation of the caller ID in the CLID.
strip	(Optional) Strips the calling-party number from the CLID. name—(Optional) Calling-party name. Causes removal of the calling-party name from the CLID. pi-restrict [all]—(Optional) Restricted PI. Causes removal of all calling-party names and numbers from the CLID when the PI is restricted.
substitute name	(Optional) Copies the calling number into the display name if PI allows it (and the calling name is empty).

Command Default No default behavior or values

Command Modes Dial-peer configuration

Command History	Release	Modification
	12.2(11)T	This command was introduced.
	12.2(13)T	The override rdnis keywords were added.
	12.4(4)T	The following keywords were added: network-provided , pi-restrict all , and substitute name .

Usage Guidelines

The **override rdnis** keywords are supported only for POTS dial peers.

CLID is the collection of information about the billing telephone number from which a call originated. The CLID value might be the entire phone number, the area code, or the area code plus the local exchange. It is also known as caller ID. The various keywords to this command manage the presentation, restriction, or stripping of the various CLID elements.

The **clid network-number** command sets the presentation indicator to “y” and the screening indicator to “network-provided.” The **second-number strip** keyword strips from the H.225 source-address field the original calling-party number, and is valid only if a network number was previously configured.

The **clid override rdnis** command overrides the CLID with the RDNIS if it is available.

The **clid restrict** command causes the calling-party number to be present in the information element, but the presentation indicator is set to “n” to prevent its presentation to the called party.

The **clid strip** command causes the calling-party number to be null in the information element, and the presentation indicator is set to “n” to prevent its presentation to the called party.

Examples

The following example sets the calling-party network number to 98765 for POTS dial peer 4321:

```
Router(config)# dial-peer voice 4321 pots
Router(config-dial-peer)# clid network-number 98765
```

An alternative method of accomplishing this result is to enter the **second-number strip** keywords as part of the **clid network-number** command. The following example sets the calling-party network number to 56789 for VoIP dial peer 1234 and also prevents the second network number from being sent:

```
Router(config)# dial-peer voice 1234 voip
Router(config-dial-peer)# clid network-number 56789 second-number strip
```

The following example overrides the calling-party number with RDNIS if available:

```
Router(config-dial-peer)# clid override rdnis
```

The following example prevents the calling-party number from being presented:

```
Router(config-dial-peer)# clid restrict
```

The following example removes the calling-party number from the CLID information and prevents the calling-party number from being presented:

```
Router(config-dial-peer)# clid strip
```

The following example strips the name from the CLID information and prevents the name from being presented:

```
Router(config-dial-peer)# clid strip name
```

The following example strips the calling party number when PI is set to restrict clid strip from the CLID information and prevents the calling party number from being presented:

```
Router(config-dial-peer)# clid strip pi-restrict
```

The following example strips calling party name and number when the PI is set to the restrict all clid strip from the CLID information and prevents the calling party name and number from being presented:

```
Router(config-dial-peer)# clid strip pi-restrict all
```

The following example substitutes the calling party number into the display name:

```
Router(config-dial-peer)# clid substitute name
```

The following example allows you to set the screening indicator to reflect that the number was provided by the network:

```
Router(config-dial-peer)# clid network-provided
```

Related Commands

Command	Description
clid (voice-service-voip)	Passes the network provided ISDN numbers in an ISDN calling party information element screening indicator field, removes the calling party name and number from the calling-line identifier in voice service voip configuration mode, or allows a presentation of the calling number by substituting for the missing Display Name field in the Remote-Party-ID and From headers.

clid (voice-service-voip)

Passes the network-provided ISDN numbers in an ISDN calling party information element screening indicator field, removes the calling party name and number from the calling-line identifier in voice service voip configuration mode, or allows a presentation of the calling number by substituting for the missing Display Name field in the Remote-Party-ID and From headers.

clid {network-provided | strip pi-restrict all | substitute name}

no {clid network-provided | strip pi-restrict all | substitute name}

Syntax Description	network-provided	Sets the screen indicator as network-provided.
	strip pi-restrict all	Removes the CLID when the progress indicator (PI) is restricted for PSTN to SIP operations and removes the calling party name and number when the PI is restricted for PSTN to SIP operations.
	substitute name	Copies the calling number to the display name if unavailable for PSTN to SIP operations.

Command Default The clid (voice-service-voip) command passes along user-provided ISDN numbers in an ISDN calling party information element screening indicator field.

Command Modes Voice-service-VoIP configuration

Command History	Release	Modification
	12.4(4)T	This command was introduced.

Usage Guidelines

Use the **clid network-provided** command to pass along network-provided ISDN numbers in an ISDN calling party information element screening indicator field.

Use the **clid strip pi-restrict all** command to remove the Calling Party Name and Calling Party Number from the CLID.

Use the **clid substitute name** command to allow a presentation of the Display Name field in the Remote-Party-ID and From headers. The Calling Number is substituted for the Display Name field.

Examples

The following passes along network-provided ISDN numbers in an ISDN calling party information element screening indicator field:

```
Router(config)# clid network-provided
```

The following passes along user-provided ISDN numbers in an ISDN calling party information element screening indicator field:

```
Router(config)# no clid network-provided
```

The following removes the calling party name and number from the calling-line identifier (CLID):

```
Router(config)# clid strip pi-restrict all
```

The following does not remove the calling party name and number from the CLID:

```
Router(config)# no clid strip pi-restrict all
```

The following allows the presentation of the calling number to be substituted for the missing Display Name field in the Remote-Party-ID and From headers:

```
Router(config)# clid substitute name
```

The following disallows the presentation of the calling number to be substituted for the missing Display Name field in the Remote-Party-ID and From headers:

```
Router(config)# no clid substitute name
```

Related Commands

Command	Description
clid (dial-peer)	Controls the presentation and use of CLID information in dial-peer configuration mode.

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■ **clid (voice-service-voip)**