

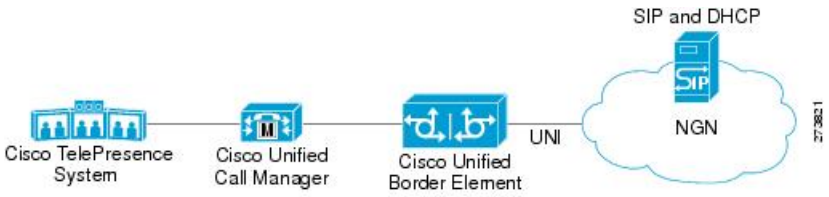
Support for PAID, PPID, Privacy, PCPID, and PAURI Headers

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Overview

The figure below shows a typical network topology where the Cisco Unified Border Element (CUBE) is configured to route messages between a call manager system (such as the Cisco Unified Call Manager) and a Next Generation Network (NGN).

Figure 1: Cisco Unified Border Element and Next Generation Topology



Devices that connect to an NGN must comply with the User-Network Interface (UNI) specification. The CUBE supports the NGN UNI specification and can be configured to interconnect NGN with other call manager systems, such us the Cisco Unified Call Manager.

The CUBE supports the following:

- the use of P-Preferred Identity (PPID), P-Asserted Identity (PAID), Privacy, P-Called Party Identity (PCPID), in INVITE messages
- the translation of PAID headers to PPID headers and vice versa
- the translation of RPID headers to PAID or PPID headers and vice versa
- the configuration and/or pass through of privacy header values
- the use of the PCPID header to route INVITE messages
- the use of multiple PAURI headers in the response messages (200 OK) it receives to REGISTER messages

P-Preferred Identity and P-Asserted Identity Headers

NGN servers use the PPID header to identify the preferred number that the caller wants to use. The PPID is part of INVITE messages sent to the NGN. When the NGN receives the PPID, it authorizes the value, generates a PAID based on the preferred number, and inserts it into the outgoing INVITE message towards the called party.

However, some call manager systems, such as Cisco Unified Call Manager 5.0, use the Remote-Party Identity (RPID) value to send calling party information. Therefore, the Cisco Unified Border Element must support building the PPID value for an outgoing INVITE message to the NGN, using the RPID value or the From: value received in the incoming INVITE message. Similarly, CUBE supports building the RPID and/or From: header values for an outgoing INVITE message to the call manager, using the PAID value received in the incoming INVITE message from the NGN.

In non-NGN systems, the CUBE can be configured to translate between PPID and PAID values, and between From: or RPID values and PAID/PPID values, at global and dial-peer levels.

In configurations where all relevant servers support the PPID or PAID headers, the Cisco Unified Border Element can be configured to transparently pass the header.

Note If the NGN sets the From: value to anonymous, the PAID is the only value that identifies the caller.

The table below describes the types of INVITE message header translations supported by the Cisco Unified Border Element. It also includes information on the configuration commands to use to configure P-header translations.

The table below shows the P-header translation configuration settings only. In addition to configuring these settings, you must configure other system settings (such as the session protocol).

Table 1: P-header Configuration Settings

Incoming Header	Outgoing Header	Configuration Notes
From:	RPID	To enable the translation to RPID headers in the outgoing header, use the **remote-party-id** command in SIP user-agent configuration mode. For example: Router(config-sip-ua)# **remote-party-id** This is the default system behavior. **Note** If both, **remote-party-id** and **asserted-id** commands are configured, then the **asserted-id** command takes precedence over the **remote-part-id** command.
PPID	PAID	To enable the translation to PAID privacy headers in the outgoing header at a global level, use the **asserted-id pai** command in voice service VoIP SIP configuration mode. For example: Router(conf-serv-sip)# **asserted-id pai** To enable the translation to PAID privacy headers in the outgoing header on a specific dial peer, use the **voice-class sip asserted-id pai** command in dial peer voice configuration mode. For example: Router(config-dial-peer)# **voice-class sip asserted-id pai**

Incoming Header	Outgoing Header	Configuration Notes
PPID	RPID	To enable the translation to RPID headers in the outgoing header, use the **remote-party-id** command in SIP user-agent configuration mode. For example: Router(config-sip-ua)# **remote-party-id** This is the default system behavior.
PAID	PPID	To enable the translation to PPID privacy headers in the outgoing header at a global level, use the **asserted-id ppi** command in voice service VoIP SIP configuration mode. For example: Router(conf-serv-sip)# **asserted-id ppi** To enable the translation to PPID privacy headers in the outgoing header on a specific dial peer, use the **voice-class sip asserted-id ppi** command in dial peer voice configuration mode. For example: Router(config-dial-peer)# **voice-class sip asserted-id ppi**
PAID	RPID	To enable the translation to RPID headers in the outgoing header, use the **remote-party-id** command in SIP user-agent configuration mode. For example: Router(config-sip-ua)# **remote-party-id** This is the default system behavior.
RPID	PPID	To enable the translation to PPID privacy headers in the outgoing header at a global level, use the **asserted-id ppi** command in voice service VoIP SIP configuration mode. For example: Router(conf-serv-sip)# **asserted-id ppi** To enable the translation to PPID privacy headers in the outgoing header on a specific dial peer, use the **voice-class sip asserted-id ppi** command in dial peer voice configuration mode. For example: Router(config-dial-peer)# **voice-class sip asserted-id ppi**
RPID	PAID	To enable the translation to PAID privacy headers in the outgoing header at a global level, use the **asserted-id pai** command in voice service VoIP SIP configuration mode. For example: Router(conf-serv-sip)# **asserted-id pai** To enable the translation to PAID privacy headers in the outgoing header on a specific dial peer, use the **voice-class sip asserted-id pai** command in dial peer voice configuration mode. For example: Router(config-dial-peer)# **voice-class sip asserted-id pai**
RPID	From:	By default, the translation to RPID headers is enabled and the system translates PPID headers in incoming messages to RPID headers in the outgoing messages. To disable the default behavior and enable the translation from PPID to From: headers, use the **no remote-party-id** command in SIP user-agent configuration mode. For example: Router(config-sip-ua)# **no remote-party-id**

Note Privacy functions are not initialized on Unified Border Element without configuring **asserted-id pai** or **asserted-id ppi**. Ensure that you configure **asserted-id pai** or **asserted-id ppi** to support privacy functions on Unified Border Element.

Just configuring **asserted-id pai** in dial-peer or global configuration mode is sufficient to process both PPID and PAID headers.

The CUBE can be configured to transparently pass the PAID and PPID headers in the incoming and outgoing Session Initiation Protocol (SIP) requests or response messages from end-to-end.

- Requests include: INVITEs and UPDATEs
- Responses include:18x and 200OK

Note The priority of P-headers are in the following order: PAID, PPID, and RPID.

Table 2: PAID and PPID header configuration settings for mid-call requests and responses

Incoming Header	Outgoing Header	Configuration Notes
PAID	PPID	To enable the translation to PPID headers in the outgoing header at a global level, use the **asserted-id ppi** command in voice service VoIP SIP configuration mode. For example: Router(conf-serv-sip)# **asserted-id ppi** To enable the translation to PPID headers in the outgoing header on a specific dial peer, use the **voice-class sip asserted-id ppi** command in dial peer voice configuration mode. For example: Router(config-dial-peer)# **voice-class sip asserted-id ppi**

Incoming Header	Outgoing Header	Configuration Notes
RPID	PPID	To enable the translation to PPID headers in the outgoing header at a global level, use the **asserted-id ppi** command in voice service VoIP SIP configuration mode. For example: Router(conf-serv-sip)# **asserted-id ppi** To enable the translation to PPID headers in the outgoing header on a specific dial peer, use the **voice-class sip asserted-id ppi** command in dial peer voice configuration mode. For example: Router(config-dial-peer)# **voice-class sip asserted-id ppi**
PPID	PPID	To enable the translation to PPID headers in the outgoing header at a global level, use the **asserted-id ppi** command in voice service VoIP SIP configuration mode. To enable the translation to PPID headers in the outgoing header on a specific dial peer, use the **voice-class sip asserted-id ppi** command in dial peer voice configuration mode. For example: Router(config-dial-peer)# **voice-class sip asserted-id ppi**
PAID	PAID	To enable the translation to PAID headers in the outgoing header at a global level, use the **asserted-id pai** command in voice service VoIP SIP configuration mode. To enable the translation to PAID headers in the outgoing header on a specific dial peer, use the **voice-class sip asserted-id pai** command in dial peer voice configuration mode. For example: Router(config-dial-peer)# **voice-class sip asserted-id pai**

Incoming Header	Outgoing Header	Configuration Notes
RPID	PAID	To enable the translation to PAID headers in the outgoing header at a global level, use the **asserted-id pai** command in voice service VoIP SIP configuration mode. To enable the translation to PAID headers in the outgoing header on a specific dial peer, use the **voice-class sip asserted-id pai** command in dial peer voice configuration mode.
PPID	PAID	To enable the translation to PAID headers in the outgoing header at a global level, use the **asserted-id pai** command in voice service VoIP SIP configuration mode. To enable the translation to PAID headers in the outgoing header on a specific dial peer, use the **voice-class sip asserted-id pai** command in dial peer voice configuration mode.
PAID	RPID	To enable the translation to RPID headers in the outgoing header, use the **remote-party-id** command in SIP user-agent configuration mode. For example: Router(config-sip-ua)# **remote-party-id**. **Note** PAID and PPID headers are not configured in this case.
RPID	RPID	To enable the translation to RPID headers in the outgoing header, use the **remote-party-id** command in SIP user-agent configuration mode. For example: Router(config-sip-ua)# **remote-party-id**. **Note** PAID and PPID headers are not configured in this case.

Incoming Header	Outgoing Header	Configuration Notes
PPID	RPID	To enable the translation to RPID headers in the outgoing header, use the **remote-party-id** command in SIP user-agent configuration mode. For example: Router(config-sip-ua)# **remote-party-id**
FROM	FROM	No configuration required except for the **remote-party-id** header.
FROM	RPID	To enable the translation to RPID headers in the outgoing header, use the **remote-party-id** command in SIP user-agent configuration mode. For example: Router(config-sip-ua)# **remote-party-id**
PAID	PAID	Enables PPID headers on the incoming dial-peer and PAID headers on the outgoing dial-peer.
RPID	PAID	Enables PPID headers on incoming dial-peer and PAID headers on outgoing dial-peer.
PPID	PAID	Enables PPID headers on incoming dial-peer and PAID headers on outgoing dial-peer.
PAID	PAID	Enables RPID headers on incoming dial-peer and PAID headers on outgoing dial-peer.
RPID	PAID	Enables RPID headers on incoming dial-peer and PAID headers on outgoing dial-peer.
PPID	PAID	Enables RPID headers on incoming dial-peer and PAID headers on outgoing dial-peer.
PAID	PPID	Enables PAID headers on incoming dial-peer and PPID headers on outgoing dial-peer.
RPID	PPID	Enables PAID headers on incoming dial-peer and PPID headers on outgoing dial-peer.

Incoming Header	Outgoing Header	Configuration Notes
PPID	PPID	Enables PAID headers on incoming dial-peer and PPID on outgoing dial-peer.
PAID	PPID	Enables RPID headers on incoming dial-peer and PPID headers on outgoing dial-peer.
RPID	PPID	Enables RPID headers on incoming dial-peer and PPID headers on outgoing dial-peer.
PPID	PPID	Enables RPID headers on incoming dial-peer and PPID headers on outgoing dial-peer.
PAID	RPID	Enables PPID headers on incoming dial-peer and RPID headers on outgoing dial-peer. **Note** PAID headers will be given priority and RPID headers will be created using the PAID header information.
RPID	RPID	Enables PPID headers on incoming dial-peer and RPID headers on outgoing dial-peer.
PPID	RPID	Enables PPID headers on incoming dial-peer and RPID headers on outgoing dial-peer. **Note** PPID headers will be given priority and RPID headers will be created using the PPID header information.
PAID	RPID	Enables PAID headers on incoming dial-peer and RPID headers on outgoing dial-peer. **Note** PAID headers will be given priority and RPID headers will be created using the PAID header information.

Incoming Header	Outgoing Header	Configuration Notes
RPID	RPID	Enables PAID headers on incoming dial-peer and RPID headers on outgoing dial-peer.
PPID	RPID	Enables PAID headers on incoming dial-peer and RPID headers on outgoing dial-peer. **Note** PPID headers will be given priority and RPID headers will be created using the PPID header information.

Privacy

If the user is subscribed to a privacy service, the Cisco Unified Border Element can support privacy using one of the following methods:

- Using prefixes

The NGN dial plan can specify prefixes to enable privacy settings. For example, the dial plan may specify that if the caller dials a prefix of 184, the calling number is not sent to the called party.

The dial plan may also specify that the caller can choose to send the calling number to the called party by dialing a prefix of 186. Here, the Cisco Unified Border Element transparently passes the prefix as part of the called number in the INVITE message.

The actual prefixes for the network are specified in the dial plan for the NGN, and can vary from one NGN to another.

- Using the Privacy header

If the Privacy header is set to None, the calling number is delivered to the called party. If the Privacy header is set to a Privacy:id value, the calling number is not delivered to the called party.

- Using Privacy values from the peer call leg

If the incoming INVITE has a Privacy header or a RPID with privacy on, the outgoing INVITE can be set to Privacy: id. This behavior is enabled by configuring **privacy pstn** command globally or **voice-class sip privacy pstn** command on the selected dial-per.

Incoming INVITE can have multiple privacy header values, id, user, session, and so on. Configure the **privacy-policy passthru** command globally or **voice-class sip privacy-policy passthru** command to transparently pass across these multiple privacy header values.

Some NGN servers require a Privacy header to be sent even though privacy is not required. In this case the Privacy header must be set to none. The Cisco Unified Border Element can add a privacy header with the value None while forwarding the outgoing INVITE to NGN. Configure the **privacy-policy send-always** globally or **voice-class sip privacy-policy send-always** command in dial-peer to enable this behavior.

If the user is not subscribed to a privacy service, the Cisco Unified Border Element can be configured with no Privacy settings.

Note For the Privacy functions to work as intended, the command **asserted-id {pai|ppi}** must be configured.

P-Called Party Identity

The Cisco Unified Border Element can be configured to use the PCPID header in an incoming INVITE message to route the call, and to use the PCPID value to set the To: value of outgoing INVITE messages.

The PCPID header is part of the INVITE messages sent by the NGN, and is used by Third Generation Partnership Project (3GPP) networks. The Cisco Unified Border Element uses the PCPID from incoming INVITE messages (from the NGN) to route calls to the Cisco Unified Call Manager.

Note The PCPID header supports the use of E.164 numbers only.

P-Associated URI

The Cisco Unified Border Element supports the use of PAURI headers sent as part of the registration process. After the Cisco Unified Border Element sends REGISTER messages using the configured E.164 number, it receives a 200 OK message with one or more PAURIs. The number in the first PAURI (if present) must match the contract number. The Cisco Unified Border Element supports a maximum of six PAURIs for each registration.

Note The Cisco Unified Border Element performs the validation process only when a PAURI is present in the 200 OK response.

The registration validation process works as follows:

- The Cisco Unified Border Element receives a REGISTER response message that includes PAURI headers that include the contract number and up to five secondary numbers.
- The Cisco Unified Border Element validates the contract number against the E.164 number that it is registering:
  - If the values match, the Cisco Unified Border Element completes the registration process and stores the PAURI value. This allows administration tools to view or retrieve the PAURI if needed.
  - If the values do not match, the Cisco Unified Border Element unregisters and then reregisters the contract number. The Cisco Unified Border Element performs this step until the values match.

Random Contact Support

The Cisco Unified Border Element can use random-contact information in REGISTER and INVITE messages so that user information is not revealed in the contact header.

To provide random contact support, the Cisco Unified Border Element performs SIP registration based on the random-contact value. The Cisco Unified Border Element then populates outgoing INVITE requests with the random-contact value and validates the association between the called number and the random value in

the Request-URI of the incoming INVITE. The Cisco Unified Border Element routes calls based on the PCPID, instead of the Request-URI which contains the random value used in contact header of the REGISTER message.

The default contact header in REGISTER messages is the calling number. The Cisco Unified Border Element can generate a string of 32 random alphanumeric characters to replace the calling number in the REGISTER contact header. A different random character string is generated for each pilot or contract number being registered. All subsequent registration requests will use the same random character string.

The Cisco Unified Border Element uses the random character string in the contact header for INVITE messages that it forwards to the NGN. The NGN sends INVITE messages to the Cisco Unified Border Element with random-contact information in the Request URI. For example: INVITE sip:FefhH3zIHe9i8ImcGjDD1PEc5XfFy51G@10.12.1.46:5060.

The Cisco Unified Border Element will not use the To: value of the incoming INVITE message to route the call because it might not identify the correct user agent if supplementary services are invoked. Therefore, the Cisco Unified Border Element must use the PCPID to route the call to the Cisco Unified Call Manager. You can configure routing based on the PCPID at global and dial-peer levels.

Feature Information

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to www.cisco.com/go/cfn. An account on Cisco.com is not required.

Table 3: Feature Information for PAID and PPID Headers on CUBE

Feature Name	Releases	Feature Information
PAID and PPID Headers in mid-call re-INVITE and UPDATE request and responses on Cisco Unified Border Element	Baseline Fuctionality	This feature enables CUBE platforms to support: • P-Preferred Identity (PPID) and P-Asserted Identity (PAID) in mid-call re-INVITE messages and responses from end-to-end. • P-Preferred Identity (PPID) and P-Asserted Identity (PAID) in mid-call UPDATE messages and responses from end-to-end. • Configuration and/or pass through of PAID and PPID header values.

Restrictions

- To enable random-contact support, you must configure the Cisco Unified Border Element to support SIP registration with random-contact information. In addition, you must configure random-contact support in VoIP voice-service configuration mode or on the dial peer.

- If random-contact support is configured for SIP registration only, the system generates the random-contact information, includes it in the SIP REGISTER message, but does not include it in the SIP INVITE message.
- If random-contact support is configured in VoIP voice-service configuration mode or on the dial peer only, no random contact is sent in either the SIP REGISTER or INVITE message.
- Passing of “+" is not supported with PAID PPID Privacy PCPID and PAURI Headers.

Configure P-Header and Random-Contact Support

To enable random contact support you must configure the CUBE to support Session Initiation Protocol (SIP) registration with random-contact information, as described in this section.

To enable the CUBE to use the PCPID header in an incoming INVITE message to route the call, and to use the PCPID value to set the To: value of outgoing INVITE messages, you must configure P-Header support as described in this section.

Configure P-Header Translation

To configure P-Header translations on a Cisco Unified Border Element, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **asserted-id** *header-type*
6. **exit**

DETAILED STEPS

Procedure

	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.

	Command or Action	Purpose
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters VoIP voice-service configuration mode.
Step 4	**sip** **Example:** `Router(conf-voi-serv)# sip`	Enters voice service VoIP SIP configuration mode.
Step 5	**asserted-id** *header-type* **Example:** `Router(conf-serv-sip)# asserted-id ppi`	Specifies the type of privacy header in the outgoing SIP requests and response messages.
Step 6	**exit** **Example:** `Router(conf-serv-sip)# exit`	Exits the current mode.

Configure P-Header Translation on an Individual Dial Peer

To configure P-Header translation on an individual dial peer, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **voice-class sip asserted-id** *header-type*
5. **exit**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:**	Enters global configuration mode.

	Command or Action	Purpose
	`Router# configure terminal`	
Step 3	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 2611 voip`	Defines the dial peer, specifies the method of voice encapsulation, and enters dial peer voice configuration mode.
Step 4	**voice-class sip asserted-id** *header-type* **Example:** `Router(config-dial-peer)# voice-class sip asserted-id ppi`	Specifies the type of privacy header in the outgoing SIP requests and response messages, on this dial peer.
Step 5	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits the current mode.

Configure P-Called-Party-Id Support

To configure P-Called-Party-Id support on a Cisco Unified Border Element, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **call-route p-called-party-id**
6. **random-request-uri validate**
7. **exit**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:**	Enters global configuration mode.

	Command or Action	Purpose
	`Router# configure terminal`	
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters VoIP voice-service configuration mode.
Step 4	**sip** **Example:** `Router(conf-voi-serv)# sip`	Enters voice service VoIP SIP configuration mode.
Step 5	**call-route p-called-party-id** **Example:** `Router(conf-serv-sip)# call-route p-called-party-id`	Enables the routing of calls based on the PCPID header.
Step 6	**random-request-uri validate** **Example:** `Router(conf-serv-sip)# random-request-uri validate`	Enables the validation of the random string in the Request URI of the incoming INVITE message.
Step 7	**exit** **Example:** `Router(conf-serv-sip)# exit`	Exits the current mode.

Configure P-Called-Party-Id Support on an Individual Dial Peer

To configure P-Called-Party-Id support on an individual dial peer, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **voice-class sip call-route p-called-party-id**
5. **voice-class sip random-request-uri validate**
6. **exit**

DETAILED STEPS

Procedure

	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** *tag* **voip** **Example:** `Router(config)# dial-peer voice 2611 voip`	Defines the dial peer, specifies the method of voice encapsulation, and enters dial peer voice configuration mode.
Step 4	**voice-class sip call-route p-called-party-id** **Example:** `Router(config-dial-peer)# voice-class sip call-route p-called-party-id`	Enables the routing of calls based on the PCPID header on this dial peer.
Step 5	**voice-class sip random-request-uri validate** **Example:** `Router(config-dial-peer)# voice-class sip random-request-uri validate`	Enables the validation of the random string in the Request URI of the incoming INVITE message on this dial peer.
Step 6	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits the current mode.

Configure Privacy Support on a Cisco Unified Border Element

To configure privacy support on a Cisco Unified Border Element, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**

5. **privacy** *privacy-option*
6. **privacy-policy** *privacy-policy-option*
7. **exit**

DETAILED STEPS

Procedure

	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 VoIP voice-service configuration mode.
Step 4	**sip** **Example:** `Router(conf-voi-serv)# sip`	Enters voice service VoIP SIP configuration mode.
Step 5	**privacy** *privacy-option* **Example:** `Router(conf-serv-sip)# privacy id`	Enables the privacy settings for the header.
Step 6	**privacy-policy** *privacy-policy-option* **Example:** `Router(conf-serv-sip)# privacy-policy passthru`	Specifies the privacy policy to use when passing the privacy header from one SIP leg to the next.
Step 7	**exit** **Example:** `Router(conf-serv-sip)# exit`	Exits the current mode.

Configure Privacy Support on an Individual Dial Peer

To configure privacy support on an individual dial peer, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **voice-class sip privacy** *privacy-option*
5. **voice-class sip privacy-policy** *privacy-policy-option*
6. **exit**

DETAILED STEPS

Procedure

	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** *tag* **voip** **Example:** `Router(config)# dial-peer voice 2611 voip`	Defines the dial peer, specifies the method of voice encapsulation, and enters dial peer voice configuration mode.
Step 4	**voice-class sip privacy** *privacy-option* **Example:** `Router(config-dial-peer)# voice-class sip privacy id`	Enables the privacy settings for the header on this dial peer.
Step 5	**voice-class sip privacy-policy** *privacy-policy-option* **Example:** `Router(config-dial-peer)# voice-class sip privacy-policy passthru`	Specifies the privacy policy to use when passing the privacy header from one SIP leg to the next, on this dial peer.
Step 6	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits the current mode.

Configure Random-Contact Support on a Cisco Unified Border Element

To configure random-contact support on a Cisco Unified Border Element, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **credentials username** *username* **password** *password* **realm** *domain-name*
5. **registrar ipv4:** *destination-address* **random-contact expires** *expiry*
6. **exit**
7. **voice service voip**
8. **sip**
9. **random-contact**
10. **exit**

DETAILED STEPS

Procedure

	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	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters SIP user-agent configuration mode.
Step 4	**credentials username** *username* **password** *password* **realm** *domain-name* **Example:** `Router(config-sip-ua)# credentials username 123456 password cisco realm cisco`	Sends a SIP registration message from the Cisco Unified Border Element. **Note** Ensure that realm name is configured w its name.
Step 5	**registrar ipv4:** *destination-address* **random-contact expires** *expiry*	Enables the SIP gateways to register E.164 numbers on behalf of analog telephone voice ports (FXS), IP phone

	Command or Action	Purpose
	Example: `Router(config-sip-ua)# registrar ipv4:10.1.2.2 random-contact expires 200`	virtual voice ports (EFXS), and Skinny Client Control Protocol (SCCP) phones with an external SIP proxy or SIP registrar. • The **random-contact** keyword configures the Cisco Unified Border Element to send the random string from the REGISTER message to the registrar.
Step 6	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits the current mode.
Step 7	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters VoIP voice-service configuration mode.
Step 8	**sip** **Example:** `Router(conf-voi-serv)# sip`	Enters voice service VoIP SIP configuration mode.
Step 9	**random-contact** **Example:** `Router(conf-serv-sip)# random-contact`	Enables random-contact support on a Cisco Unified Border Element.
Step 10	**exit** **Example:** `Router(conf-serv-sip)# exit`	Exits the current mode.

Configure Random-Contact Support for an Individual Dial Peer

To configure random-contact support for an individual dial peer, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **credentials username** *username* **password** *password* **realm** *domain-name*
5. **registrar ipv4:** *destination-address* **random-contact expires** *expiry*
6. **exit**
7. **dial-peer voice** *tag* **voip**
8. **voice-class sip random-contact**

9. **exit**

DETAILED STEPS

Procedure

	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	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters SIP user-agent configuration mode.
Step 4	**credentials username** *username* **password** *password* **realm** *domain-name* **Example:** `Router(config-sip-ua)# credentials username 123456 password cisco realm cisco`	Sends a SIP registration message from the Cisco Unified Border Element.
Step 5	**registrar ipv4:** *destination-address* **random-contact expires** *expiry* **Example:** `Router(config-sip-ua)# registrar ipv4:10.1.2.2 random-contact expires 200`	Enables the SIP gateways to register E.164 numbers on behalf of FXS, EFXS, and SCCP phones with an external SIP proxy or SIP registrar. • The **random-contact** keyword configures the Cisco Unified Border Element to send the random string from the REGISTER message to the registrar.
Step 6	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits the current mode.
Step 7	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 2611 voip`	Defines the dial peer, specifies the method of voice encapsulation, and enters dial peer voice configuration mode.

	Command or Action	Purpose
Step 8	**voice-class sip random-contact** **Example:** `Router(config-dial-peer)# voice-class sip random-contact`	Enables random-contact support on this dial peer.
Step 9	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits the current mode.