Mid-call Signaling Consumption

The Cisco Unified Border Element BE Mid-call Signaling support aims to reduce the interoperability issues that arise due to consuming mid-call RE-INVITES/UPDATES.

Mid-call Re-INVITEs/UPDATEs can be consumed in the following ways:

- Mid-call Signaling Passthrough - Media Change
- Mid-call Signaling Block
- Mid-call Signaling Codec Preservation

Note This feature should be used as a last resort only when there is no other option in CUBE. This is because configuring this feature can break video-related features. For Delay-offer Re-INVITE, the configured codec will be passed as an offer in 200 message to change the codec, the transcoder is added in the answer.

Feature Information for Mid-call Signaling

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 https://cfnng.cisco.com/. An account on Cisco.com is not required.

Table 1: Feature Information for Mid-call Signaling

Feature Name	Releases	Feature Information
Mid-call Re-INVITE Consumption	Cisco IOS 15.2(1)T Cisco IOS XE 3.6S	The Mid-call Re-INVITE consumption feature consumes mid-call Re-INVITEs from CUBE and helps to avoid interoperability issues because of these re-invites The following commands were introduced or modified: **midcall-signaling**.
Mid-call Codec Preservation	Cisco IOS 15.3(2)T Cisco IOS XE 3.9S	The Mid-call Codec Preservation feature helps to disables codec negotiation in the middle of a call and preserves the codec negotiated before the call. The following commands were introduced or modified: **midcall-signaling preserve-codec** , **voice-class sip midcall-signaling preserve-codec** .
Mid-call Re-INVITE Consumption Enhancements	Cisco IOS 15.5(3)M Cisco IOS XE 3.16S	Mid-call signaling Re-INVITE consumption is enhanced to support: • Re-INVITE based call transfer • Call transfer with REFER Consume • Normalization of call hold in a call set-up

Prerequisites

- Enable CUBE application on a device

Mid-call Signaling Passthrough - Media Change

Passthrough media change method optimizes or consumes mid-call, media-related signaling within the call. Mid-call signaling changes will be passed through only when bidirectional media like T.38 or video is added.

The command **midcall-signaling passthru media-change** needs to be configured to enable passthrough media change.

Restrictions for Mid-Call Signaling Passthrough - Media Change

- SIP-H.323 calls are not supported.
- TDM Gateways are not supported.
- Session Description Protocol (SDP) -passthrough is not supported.
- When **codec T** is configured, the offer from CUBE has only audio codecs, and so the video codecs are not consumed.
- Re-invites are not consumed if media flow-around is configured.
- Re-invites are not consumed if media anti-tromboning is configured.
- De-escalation re-invites are consumed. So, one call leg might be de-escalated to audio only while the other call leg continues to support audio and video.
- Re-invites with media direction changes are consumed.
- Video transcoding is not supported.
- Multicast Music On Hold (MMOH) is not supported.
- When the **midcall-signaling passthru media-change** command is configured and high-density transcoder is enabled, there might be some impact on Digital Signal Processing (DSP) resources as the transcoder might be used for all the calls.
- Session timer is handled leg by leg whenever this feature is configured and it includes session timer negotiation for initial INVITE/200 OK transaction as well.
- More than two m-lines in the SDP is not supported.
- Alternative Network Address Types (ANAT) is not supported.
- Video calls and Application streams are not supported when mid-call signaling block is configured.
- In the SRTP-RTP scenario, re-invites are not consumed.

Behavior of Mid-call Re-INVITE Consumption

- If mid-call signaling block is enabled on either of call-legs, video parameters and application streams are not negotiated, and are rejected in the answer.
- When flow around and offer-all is configured, CUBE performs codec renegotiation even if mid-call signaling block is configured globally.
- The following behavior is for refer consume scenario:
  - REFER consume is supported for blind, alert and consult call transfers.
  - Existing codecs or DTMF is used for local bridging of new call legs. No Re-INVITE or UPDATE is sent for media re-negotiation after REFER.

    - Call gets dropped when DSP is required but not available.
    - A call can be escalated to video only if transferee and transfer-to dial-peers do not have mid-call signaling block configured.
    - Video calls are de-escalated if mid-call signaling block configuration on transfer-to dial-peer.
    - For Re-INVITE based call-transfer involving Cisco Unified Communications Manager, all Re-INVITE are locally answered and transcoder is invoked if negotiated codecs are different than the codecs before call-transfer.

- The following behavior is for INVITE with REPLACES Header consume scenario:
    - CUBE consumes INVITE with REPLACES Header only when the **handle-replaces** CLI is configured (under **sip-ua** or **voice-class tenant**). In this case, CUBE consumes the INVITE and handles it locally. It triggers an outbound INVITE without replaces header and call gets connected with agent.
    - If the **handle-replaces** CLI is enabled, the 'transfer-to' party must have the same codec that is used for the original call setup. If there is a different codec offer, CUBE rejects the INVITE with 488 error.
    - If the **handle-replaces** CLI is not configured, CUBE does not consume the INVITE with REPLACES Header and the outgoing INVITE holds same replace header which CUBE is received.
    - INVITE with REPLACES Header consumption does not support the following configurations:
        - Delayed Offer INVITE
        - Codec, DTMF attribute changes, and RSVP
        - Mid-call Signaling block
        - IPv6

- The following table provides the details of the behavior when the initial call is establish without 'sendrecv' parameter, that means, the initial call is established with 'sendonly', 'recvonly' or 'inactive'.

Scenario	Behavior
If an Offer is received with 'sendonly' and mid-call block is configured on any or both call legs	Offer is sent with 'sendrecv'.
If an Answer is received with 'sendonly' and the peer leg supports mid-call signaling	Answer is sent with 'sendonly'. Resume transaction is end-to-end.
If an Answer is received with 'sendonly' and the peer leg does not supports mid-call signaling	Answer is sent with 'sendrecv'. Resume transaction is consumed.
If Offer as well as Answer is received with 'sendonly' and Offering leg does not support mid-call signaling	Answer is sent with 'recvonly'. Resume from Offering leg is end-to-end. Resume from answering leg is consumed.
If Offer as well as Answer is received with 'sendonly' and Answering leg does not support mid-call signaling	Answer is sent with 'inactive'. Resume from Offering leg is consumed. Resume from answering leg is end-to-end.

Scenario	Behavior
If Offer as well as Answer is received with 'sendonly' and both legs do not support mid-call signaling	Answer is sent with ' recvonly'. Resume transaction is consumed.

Configuring Passthrough of Mid-call Signalling

Perform this task to configure passthrough of mid-call signaling (as Re-invites) only when bidirectional media is added.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. Configure passthrough of mid-call signaling changes only when bidirectional media is added.
   - In Global VoIP SIP configuration mode

     midcall-signaling passthru media-change
   - In dial-peer configuration mode

     voice-class sip midcall-signaling passthru media-change
4. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Device# configure terminal`	Enters global configuration mode.
Step 3	Configure passthrough of mid-call signaling changes only when bidirectional media is added. • In Global VoIP SIP configuration mode **midcall-signaling passthru media-change** • In dial-peer configuration mode **voice-class sip midcall-signaling passthru media-change** **Example:** In Global VoIP SIP configuration mode:	Re-Invites are passed through only when bidirectional media is added.

	Command or Action	Purpose

	`Device(config)# voice service voip`
	`Device(conf-voi-serv)# sip`
	`Device(conf-serv-sip)# midcall-signaling passthru media-change`
	Example:
	In Dial-peer configuration mode:
	`Device(config)# dial-peer voice 2 voip`
	`Device(config-dial-peer)# voice-class sip midcall-signaling passthru media-change`	

Step 4	**end**	Exits to privileged EXEC mode.

Example Configuring Passthrough SIP Messages at Dial Peer Level

The following example shows how to passthrough SIP messages at the dial peer Level:

```
dial-peer voice 600 voip
 destination-pattern 2222222222
 session protocol sipv2
 session target ipv4:9.45.38.39:9001
voice-class sip midcall-signaling passthru media-change
 incoming called-number 1111111111
voice-class codec  2 offer-all
dial-peer voice 400 voip
 destination-pattern 1111111111
 session protocol sipv2
 session target ipv4:9.45.38.39:9000
 incoming called-number 2222222222
voice-class codec 1 offer-all
```

Example Configuring Passthrough SIP Messages at the Global Level

The following example shows how to passthrough SIP messages at the global level:

```
Device(config)# voice service voip
Device(conf-voi-serv)# no ip address trusted authenticate
Device(conf-voi-serv)# allow-connections sip to sip
Device(conf-voi-serv)# sip
Device(conf-serv-sip)# midcall-signaling passthru media-change
```

Mid-call Signaling Block

The Block method blocks all mid-call media-related signaling to the specific SIP trunk. The command **midcall-signaling block** needs to be configured to enable this behavior. Video escalation and T.38 call flow are rejected when the **midcall-signaling block** command is configured. This command should be configured only when basic call is the focus and mid-call can be consumed.

Restrictions for Mid-Call Signaling Block

- SIP-H.323 calls are not supported.

- TDM Gateways are not supported.
- Session Description Protocol (SDP) -passthrough is not supported
- Video calls and Application streams are not supported.
- When media flow-around is configured, Mid-call INVITE is rejected with 488 error message.
- Re-invites are not consumed if media anti-tromboning is configured.
- Multicast Music On Hold (MMOH) is not supported.
- When the **midcall-signaling passthru media-change** command is configured and high-density transcoder is enabled, there might be some impact on Digital Signal Processing (DSP) resources as the transcoder might be used for all the calls.
- Session timer is handled leg by leg whenever this feature is configured.
- More than two m-lines in the SDP is not supported.
- Alternative Network Address Types (ANAT) is not supported.
- When mid-call signaling block is configured, you can either configure REFER consume or enable TCL script. Mid-call signaling block is not supported if both REFER consume and TCL script are enabled. We also recommend not to configure **supplementary-service media-renegotiate** command.
- In the SRTP-RTP scenario, re-invites are not consumed.

Blocking Mid-Call Signaling

Perform this task to block mid-call signaling:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. Configure blocking of mid-call signaling changes:
    - In Global VoIP SIP configuration mode

      midcall-signaling block
    - In dial-peer configuration mode

      voice-class sip midcall-signaling block
4. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.

	Command or Action	Purpose
Step 2	**configure terminal** **Example:** `Device# configure terminal`	Enters global configuration mode.

Step 3	Configure blocking of mid-call signaling changes:
	• In Global VoIP SIP configuration mode
	midcall-signaling block
	• In dial-peer configuration mode
	voice-class sip midcall-signaling block
	Example:
	In Global VoIP SIP configuration mode:
	`Device(config)# voice service voip`
	`Device(conf-voi-serv)# sip`
	`Device(conf-serv-sip)# midcall-signaling block`
	Example:
	In Dial-peer configuration mode:
	`Device(config)# dial-peer voice 2 voip`
	`Device(config-dial-peer)# voice-class sip midcall-signaling block`
		Mid-call signaling is always blocked.

Step 4	**end**	Exits to privileged EXEC mode.

Example Blocking SIP Messages at Dial Peer Level

```
dial-peer voice 107 voip
 destination-pattern 74000
 session protocol sipv2
 session target ipv4:9.45.36.9
 incoming called-number 84000
 voice-class codec 1 offer-all
!
dial-peer voice 110 voip
 destination-pattern 84000
 session protocol sipv2
 session target ipv4:9.45.35.2
 incoming called-number 74000
 voice-class codec 1 offer-all
 voice-class sip midcall-signaling block
!
```

Example: Blocking SIP Messages at the Global Level

The following example shows how to block SIP messages at the global Level

```
Device(config)#voice service voip
Device(config-voi-serv)#no ip address trusted authenticate
Device(config-voi-serv)#allow-connections sip to sip
Device(config-voi-serv)#sip
Device(config-serv-sip)#midcall-signaling block
```

Mid Call Codec Preservation

Mid call codec preservation defines whether a codec can be negotiated after a call has been initiated. You can enable or disable codec negotiation in the middle of a call.

Note In the SRTP-RTP scenario, re-invites are not consumed.

Configuring Mid Call Codec Preservation

This tasks disables codec negotiation in the middle of a call and preserves the codec negotiated before the call.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. Enter one of the following to disable midcall codec renegotiation:
   - In Global VoIP SIP configuration mode

     midcall-signaling preserve-codec
   - In dial-peer configuration mode

     voice-class sip midcall-signaling preserve-codec
4. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Device# configure terminal`	Enters global configuration mode.
Step 3	Enter one of the following to disable midcall codec renegotiation: • In Global VoIP SIP configuration mode **midcall-signaling preserve-codec** • In dial-peer configuration mode **voice-class sip midcall-signaling preserve-codec** **Example:**	Disables codec negotiation in the middle of a call and preserves the codec negotiated before the call.

	Command or Action	Purpose
	`Device(config)# voice service voip` `Device(conf-voi-serv)# sip` `Device(conf-serv-sip)# midcall-signaling preserve-codec` **Example:** `Device(config)# dial-peer voice 10 voip` `Device(conf-dial-peer)# voice-class sip midcall-signaling preserve-codec`	
Step 4	**end** **Example:** `Device(conf-serv-sip)# end`	Exits to privileged EXEC mode.

Example: Configuring Mid Call Codec Preservation at the Dial Peer Level

Example: Configuring Mid Call Codec Preservation at the Dial Peer Level

```
dial-peer voice 107 voip
 destination-pattern 74000
 session protocol sipv2
 session target ipv4:9.45.36.9
 incoming called-number 84000
 voice-class codec 1 offer-all
!
dial-peer voice 110 voip
 destination-pattern 84000
 session protocol sipv2
 session target ipv4:9.45.35.2
 incoming called-number 74000
 voice-class codec 1 offer-all
 voice-class sip midcall-signaling preserve-codec
!
```

Example: Configuring Mid Call Codec Preservation at the Global Level

Example: Configuring Mid Call Codec Preservation at the Global Level

```
Device(config)# voice service voip
Device(conf-voi-serv)# no ip address trusted authenticate
Device(conf-voi-serv)# allow-connections sip to sip
Device(conf-voi-serv)# sip
Device(conf-serv-sip)# midcall-signaling preserve-codec
```