H.323-to-SIP Interworking on CUBE

This chapter describes how to configure H.323-to-SIP interworking in CUBE and lists the various features supported in this interworking model.

Note H.323 protocol is no longer supported from Cisco IOS XE Bengaluru 17.6.1a onwards. Consider using SIP for multimedia applications.

Prerequisites

- Enable CUBE on the device
- Perform basic H.323 gateway configuration. See Configuring H.323 Gateway (Optional)
- Perform basic H.323 gatekeeper configuration. See Configuring H.323 Gatekeeper (Optional)

Restrictions

- Changing codecs during rotary dial peer selection is not supported.
- Voice class codec is not supported.
- Configure extended capabilities on dial peers for fast start-to-early media scenarios.
- Delayed Offer to Slow-Start is not supported for SRTP-to-SRTP H.323-to-SIP calls.
- During a triggered INVITE scenario the Cisco UBE always generates a delayed offer INVITE.
- Fast-start to delayed-media signal interworking is not supported.

- Fast Start to Early Offer Supplementary Service will not work without extended capabilities configured under dial-peer.
- GSMFR and GSMEFR codecs are not supported.
- Media flow-around is not supported.
- Passing multiple diversion headers or multiple contact header in 302 to the H.323 leg is not supported.
- RSVP for supplementary scenarios is not supported.
- Session refresh is not supported.
- SIP-to-H.323 Supplementary Services based on H.450 is not supported.
- Slow-start to early media signal interworking is not supported.
- Supplementary services are Empty Capability Set (ECS) based supplementary services from the H.323 perspective, not H.450 supplementary services.
- LTI based transcoding is not supported.
- Transcoding for supplementary calls is not supported.
- SCCP based codec transcoding is not support with an exception of Delayed-Offer to Slow-Start with static codec.
- DTMF interworking rtp-nte to inband is supported only with non-high-density transcoding in a delayed-offer to slow-start call.

H.323-to-SIP Basic Call Interworking

This feature enables the IP-to-IP gateway to bridge calls between networks that support different VoIP call-signaling protocols (SIP and H.323). The SIP-to-H.323 protocol interworking capabilities of the CUBE support the following:

Feature	Supported Release	Additional Description
Basic voice calls (G.711 and G.729 codecs)	12.4(11)T	
UDP and TCP transport	12.3(11)T	SIP (UDP)<—>H.323 (TCP) SIP (TCP)<—>H.323 (TCP) SIP (UDP)<—>H.323 (UDP) SIP (TCP)<—>H.323 (UDP) Default SIP protocol is UDP. Default H.323 protocol is TCP.

Feature	Supported Release	Additional Description
Interworking between • H.323 Fast-Start and SIP early-media signaling • H.323 Slow-Start and SIP delayed-media signaling	12.3(11)T	• H.323 Fast Start<—>SIP Early Media • H.323 Slow Start <—>SIP Delay Media **Note** No other combinations are supported. For example, H.323 Slow Start <—>SIP Early Media is not supported and results in call failure.
H.323-to-SIP RSVP Support	12.3(11)T	The following cases are supported (acc-qos and reg_qos): • H.323 to H.323 with only one leg having RSVP • H.323 to H.323 with both legs having RSVP • H.323 to SIP with only one leg having RSVP • H.323 to SIP with both legs having RSVP
DTMF relay interworking:	12.3(11)T 12.4(6)XE	• H.245 alpha/signal<—>SIP Notify • H.245 alpha/signal <—>SIP RFC 2833 • H.245 alpha/signal <—> SIP KPML • G.711 Inband DTMF<—>RFC 2833
Voice call transcoding support	12.3(11)T	• Only voice and DTMF are supported. (G.711-G.729) • Codec transparent and codec filtering is not supported • Cisco Fax Relay and T.38 Fax are not supported

Feature	Supported Release	Additional Description
Calling/called name and number	12.3(11)T	• H.323 IOS FXS/SCCP – IPIPGW – SIP IOS FXS • H.323 IOS FXS/SCCP – IPIPGW – SIP CCME Skinny Phone • H.323 IOS FXS/SCCP – IPIPGW – SIP IP Phone • CCM Phone – IPIPGW – SIP CCME Skinny Phone • CCM Phone – IPIPGW – SIP IP Phone • SIP IOS FXS – IPIPGW – H.323 IOS FXS • SIP IOS FXS – IPIPGW – H.323 CCME Skinny Phone
RADIUS call-accounting records	12.3(11)T	H.323<—>SIP Radius call accounting
TCL IVR 2.0 for SIP, including media playout and digit collection (RFC 2833 DTMF relay)	12.3(11)T 12.4(11)T	12.3(11)T—TCL IVR 2.0 for SIP, including media playout and digit collection (RFC 2833 DTMF relay) 12.4(11)T —TCL IVR support with SIP NOTIFY DTMF
SRTP Passthrough	12.4(15)XY	
Supplementary Services (ECS based).	12.4(11)XJ2	
Codec Transparent	12.4(11)T	
Extended codec support and codec filtering	12.4(11)T	

H.323-to-SIP Supplementary Features Interworking

This interworking provides enhanced termination and re-origination of signaling and media between VoIP and Video Networks in conformance with RFC3261.

Feature	Release
Support H.323-to-SIP Supplementary services for CUCM with MTP on the H.323 Trunk.	12.3(11)T

Feature	Release
ILBC Codec Support	12.3(11)T
Interworking between G.711 inband DTMF to RFC2833	12.3(11)T
VXML 3.x support	12.3(11)T
SIP CDRs and H.323 CDRs Mapping	12.3(11)T
Conference ID can be used to correlate H.323 and SIP Radius records. Conference ID is unique on both H.323 and SIP legs	12.3(11)T
VXML support with SIP Notify	12.4(11)T
Mapping ECS to ReINVITE and ECS to REFER on the Cisco CUBE	12.4(20)T

H.323-to-SIP Codec Progress Indicator Interworking for Media Cut-Through

OGW is the originating gateway and TGW is the terminating gateway.

Table 1: SIP(OGW)—>IPIPGW—>H.323(TGW) calls

SIP at In Leg	H.323 at Out Leg	Comments
183 Session Progress	Progress/Alert PI = 8	Analog phone at TGW
180 Ring	Alert with PI = 0	SCCP phone at TGW

Table 2: H.323(OGW)—>IPIPGW—>SIP(TGW) calls

H.323 at In Leg	SIP at Out Leg	Comments
Progress/Alert PI = 8	183 Session Progress	Analog phone at TGW
Alert with PI = 0	180 Ring	SIP/SCCP phone at TGW

Configuring H.323-to-SIP Interworking

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **allow-connections h.323 to sip**
5. **allow-connections sip to h.323**

6. **end**

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 Global VoIP configuration mode.
Step 4	**allow-connections h.323 to sip** **Example:**	Allows connections from a h.323 endpoint to a SIP endpoint.
Step 5	**allow-connections sip to h.323** **Example:**	Allows connections from a SIP endpoint to a H.323 endpoint.
Step 6	**end** **Example:** `Router(conf-voi-serv)# end`	Exits to previliged EXEC mode.