Exploring dtmf_relay_asymm.pcapng

📡 Voice GW B | 📡 Voice GW A | 🖥️ CUCM DTMF | 📞 DTMF Phone A

📁 **dtmf_relay_asymm.pcapng**

📞 SIP INVITE

📡 Via	SIP/2.0/UDP 10.10.223.123:5060;branch=z9hG4bKD8381
🔗 Call-ID	FCC58C96-F77611E9-8152C92B-D2F22CC5@10.10.223.123
🖥️ UA	Cisco-SIPGateway/IOS-15.8.3.M2
🔢 CSeq	101 INVITE
🔢 Max-Fwd	70
📞 Contact	<sip:10.10.223.123:5060>
👤 Owner	CiscoSystemsSIP-GW-UserAgent 8743 4180 IN IP4 10.10.223.123
Session Name (s)	SIP Call
🌐 Connection	IN IP4 10.10.223.123
🎵 Media	audio 16482 RTP/AVP 0 98
🌐 Connection	IN IP4 10.10.223.123
📋 Attr	rtpmap:0 PCMU/8000
📋 Attr	rtpmap:98 telephone-event/8000
📋 Attr	fmtp:98 0-16
📋 Attr	ptime:20

💡 SIP INVITE initiates a call — carries SDP offer with codec/media proposals; callee responds with 180 Ringing, then 200 OK with SDP answer

⏳ SIP 100 Trying

💡 100 Trying — proxy accepted the request and is searching for the callee; stops retransmission timer; no end-to-end significance

📞 SIP INVITE

📡 Via	SIP/2.0/UDP 10.10.223.107:5060;branch=z9hG4bKDB7A8
🔗 Call-ID	FCC8CAD6-F77611E9-8185CBC9-8D39E71C@10.10.223.107
🖥️ UA	Cisco-SIPGateway/IOS-16.12.1a
🔢 CSeq	101 INVITE
📞 Contact	<sip:10.10.223.107:5060>
🔢 Max-Fwd	69
👤 Owner	CiscoSystemsSIP-GW-UserAgent 6649 6049 IN IP4 10.10.223.107
Session Name (s)	SIP Call
🌐 Connection	IN IP4 10.10.223.107
🎵 Media	audio 8268 RTP/AVP 0 98
🌐 Connection	IN IP4 10.10.223.107
📋 Attr	rtpmap:0 PCMU/8000
📋 Attr	rtpmap:98 telephone-event/8000
📋 Attr	fmtp:98 0-15
📋 Attr	ptime:20

💡 SIP INVITE initiates a call — carries SDP offer with codec/media proposals; callee responds with 180 Ringing, then 200 OK with SDP answer

⏳ SIP 100 Trying

💡 100 Trying — proxy accepted the request and is searching for the callee; stops retransmission timer; no end-to-end significance

🔔 SIP 180 Ringing

💡 180 Ringing — callee's phone is alerting; caller hears locally generated ringback; 183 with SDP offers early media (in-band ringback) instead

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✅ SIP PRACK

Frame 7 | 2019-10-26T22:30:03.241976Z

🔗 Call-ID	FCC58C96-F77611E9-8152C92B-D2F22CC5@10.10.223.123
🔢 CSeq	102 PRACK
🔢 RAck	1567 101 INVITE

✅ SIP 200 OK

💡 200 OK — call answered (for INVITE) or registration accepted (for REGISTER); SDP answer completes codec negotiation; caller must send ACK to complete 3-way handshake

✅ SIP 200 OK

💡 200 OK — call answered (for INVITE) or registration accepted

Voice GW B	Voice GW A	CUCM DTMF	DTMF Phone A

📞 Contact	<sip:1252@10.10.223.47:5060>;+u.sip!devicename.ccm.cisco.com="SEPF8A5C59E1128"
🌐 Connection	IN IP4 10.10.224.2
🎵 Media	audio 27156 RTP/AVP 0 101
📋 Attr	rtpmap:0 PCMU/8000
📋 Attr	rtpmap:101 telephone-event/8000
📋 Attr	fmtp:101 0-15

(for REGISTER); SDP answer completes codec negotiation; caller must send ACK to complete 3-way handshake

✅ SIP ACK

Frame 10 | 2019-10-26T22:30:04.433968Z

🔗 Call-ID	FCC8CAD6-F77611E9-8185CBC9-8D39E71C@10.10.223.107
🔢 CSeq	101 ACK

✅ SIP 200 OK

💡 200 OK — call answered (for INVITE) or registration accepted (for REGISTER); SDP answer completes codec negotiation; caller must send ACK to complete 3-way handshake

📞 Contact	<sip:1252@10.10.223.107:5060>
🌐 Connection	IN IP4 10.10.223.107
🎵 Media	audio 8266 RTP/AVP 0 101
🌐 Connection	IN IP4 10.10.223.107
📋 Attr	rtpmap:0 PCMU/8000
📋 Attr	rtpmap:101 telephone-event/8000
📋 Attr	fmtp:101 0-16
📋 Attr	ptime:20

✅ SIP ACK

Frame 12 | 2019-10-26T22:30:04.467962Z

🔗 Call-ID	FCC58C96-F77611E9-8152C92B-D2F22CC5@10.10.223.123
🔢 CSeq	101 ACK

🎵 RTP

Frame 13 | 2019-10-26T22:30:04.503971Z

Setup frame	11
Marker	True
🎵 Codec	ITU-T G.711 PCMU (0)
Sequence number	2870
Timestamp	494229034
🔢 SSRC	0x0465df7b (73785211)

🎵 RTP

Frame 14 | 2019-10-26T22:30:04.503971Z

Setup frame	9
Marker	True
🎵 Codec	ITU-T G.711 PCMU (0)
Sequence number	2870
Timestamp	494229034
🔢 SSRC	0x0465df7b (73785211)

🎵 RTP

Frame 21 | 2019-10-26T22:30:04.565964Z

Setup frame	3
Marker	True
🎵 Codec	ITU-T G.711 PCMU (0)
Sequence number	13312
Timestamp	1613845641
🔢 SSRC	0x22dec981 (585025921)

🎵 RTP

Frame 22 | 2019-10-26T22:30:04.565964Z

Setup frame	1
Marker	True
🎵 Codec	ITU-T G.711 PCMU (0)
Sequence number	13312
Timestamp	1613845641
🔢 SSRC	0x22dec981 (585025921)

📊 RTCP SR

💡 RTCP Sender Report — media sender reports packet/byte counts and NTP timestamp for lip-sync; receivers use this to calculate jitter and loss

📦 Packets	76
📦 Bytes	12768

📊 RTCP SR

💡 RTCP Sender Report — media sender reports packet/byte counts and NTP timestamp for lip-sync; receivers use this to calculate jitter and loss

📦 Packets	76
📦 Bytes	12768

🔢 DTMF

💡 DTMF digit sent as RTP Named Telephone Event (RFC 4733) —

📡 Voice GW B | 📡 Voice GW A | 🖥️ CUCM DTMF | 📞 DTMF Phone A

🏁 End	False
⏱ Duration	80

out-of-band signaling preserves digit accuracy across codec transcoding

🔢 DTMF

🏁 End	False
⏱ Duration	80

💡 DTMF digit sent as RTP Named Telephone Event (RFC 4733) — out-of-band signaling preserves digit accuracy across codec transcoding

📊 RTCP SR

📦 Packets	248
📦 Bytes	39680

💡 RTCP Sender Report — media sender reports packet/byte counts and NTP timestamp for lip-sync; receivers use this to calculate jitter and loss

📊 RTCP SR

📦 Packets	248
📦 Bytes	39680

💡 RTCP Sender Report — media sender reports packet/byte counts and NTP timestamp for lip-sync; receivers use this to calculate jitter and loss

🔢 DTMF

🏁 End	False
⏱ Duration	0

💡 DTMF digit sent as RTP Named Telephone Event (RFC 4733) — out-of-band signaling preserves digit accuracy across codec transcoding

🔢 DTMF

🏁 End	False
⏱ Duration	0

💡 DTMF digit sent as RTP Named Telephone Event (RFC 4733) — out-of-band signaling preserves digit accuracy across codec transcoding

🛑 SIP BYE

🔗 Call-ID	FCC58C96-F77611E9-8152C92B-D2F22CC5@10.10.223.123
🔢 CSeq	103 BYE
📋 Reason	Q.850;cause=16

💡 SIP BYE terminates an established dialog — both sides send BYE to release media resources; Reason header explains why (e.g., user hangup)

✅ SIP 200 OK

💡 200 OK — call answered (for INVITE) or registration accepted (for REGISTER); SDP answer completes codec negotiation; caller must send ACK to complete 3-way handshake

🛑 SIP BYE

🔗 Call-ID	FCC8CAD6-F77611E9-8185CBC9-8D39E71C@10.10.223.107
🔢 CSeq	102 BYE
📋 Reason	Q.850;cause=16

💡 SIP BYE terminates an established dialog — both sides send BYE to release media resources; Reason header explains why (e.g., user hangup)

✅ SIP 200 OK

💡 200 OK — call answered (for INVITE) or registration accepted (for REGISTER); SDP answer completes codec negotiation; caller must send ACK to complete 3-way handshake

🛑 RTCP BYE

...0 0001 = Source count	1

Frame 2323 | 2019-10-26T22:30:15.94894Z