Exploring dtmf_relay_iwrk_rtpnte.pcapng

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

📁 **dtmf_relay_iwrk_rtpnte.pcapng**

📞 SIP INVITE

📡 Via	SIP/2.0/UDP 10.10.223.123:5060;branch=z9hG4bKB21CA7
🔗 Call-ID	A39C4DD3-F76D11E9-811EC92B-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 4710 2021 IN IP4 10.10.223.123
Session Name (s)	SIP Call
🌐 Connection	IN IP4 10.10.223.123
🎵 Media	audio 16466 RTP/AVP 0
🌐 Connection	IN IP4 10.10.223.123
📋 Attr	rtpmap:0 PCMU/8000
📋 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

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

📡 Via	SIP/2.0/UDP 10.10.223.107:5060;branch=z9hG4bKB61046
🔗 Call-ID	A39AAB6C-F76D11E9-8143CBC9-8D39E71C@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 5201 9383 IN IP4 10.10.223.107
Session Name (s)	SIP Call
🌐 Connection	IN IP4 10.10.223.107
🎵 Media	audio 8224 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 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 | 2019-10-26T21:23:08.138985Z

🔗 Call-ID	A39C4DD3-F76D11E9-811EC92B-D2F22CC5@10.10.223.123
🔢 CSeq	102 PRACK
🔢 RAck	4270 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

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📡 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 19104 RTP/AVP 0 101
📋 Attr	rtpmap:0 PCMU/8000
📋 Attr	rtpmap:101 telephone-event/8000
📋 Attr	fmtp:101 0-15

caller must send ACK to complete 3-way handshake

✅ SIP ACK

🔗 Call-ID	A39AAB6C-F76D11E9-8143CBC9-8D39E71C@10.10.223.107
🔢 CSeq	101 ACK

Frame 10 | 2019-10-26T21:23:09.292984Z

✅ SIP 200 OK

📞 Contact	<sip:1252@10.10.223.107:5060>
🌐 Connection	IN IP4 10.10.223.107
🎵 Media	audio 8222 RTP/AVP 0
🌐 Connection	IN IP4 10.10.223.107
📋 Attr	rtpmap:0 PCMU/8000
📋 Attr	ptime:20

💡 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 ACK

🔗 Call-ID	A39C4DD3-F76D11E9-811EC92B-D2F22CC5@10.10.223.123
🔢 CSeq	101 ACK

Frame 12 | 2019-10-26T21:23:09.31497Z

🎵 RTP

Setup frame	11
Marker	True
🎵 Codec	ITU-T G.711 PCMU (0)
Sequence number	9243
Timestamp	2854117503
🔢 SSRC	0x1808df7b (403234683)

Frame 13 | 2019-10-26T21:23:09.343976Z

🎵 RTP

Setup frame	9
Marker	True
🎵 Codec	ITU-T G.711 PCMU (0)
Sequence number	9243
Timestamp	2854117503
🔢 SSRC	0x1808df7b (403234683)

Frame 14 | 2019-10-26T21:23:09.343976Z

🎵 RTP

Setup frame	3
Marker	True
🎵 Codec	ITU-T G.711 PCMU (0)
Sequence number	15308
Timestamp	2396507749
🔢 SSRC	0xb5ad61ec (3048038892)

Frame 17 | 2019-10-26T21:23:09.365978Z

🎵 RTP

Setup frame	1
Marker	True
🎵 Codec	ITU-T G.711 PCMU (0)
Sequence number	15308
Timestamp	2396507749
🔢 SSRC	0xb5ad61ec (3048038892)

Frame 18 | 2019-10-26T21:23:09.365978Z

📢 SIP NOTIFY

🔗 Call-ID	A39C4DD3-F76D11E9-811EC92B-D2F22CC5@10.10.223.123
🔢 CSeq	103 NOTIFY
📋 Event	telephone-event
📊 State	active

Frame 315 | 2019-10-26T21:23:10.876953Z

🔢 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

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

✅ SIP 200 OK

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Frame 332 | 2019-10-26T21:23:10.986948Z

📢 SIP NOTIFY

🔗 Call-ID	A39C4DD3-F76D11E9-811EC92B-D2F22CC5@10.10.223.123
🔢 CSeq	104 NOTIFY
📋 Event	telephone-event
📊 State	active

✅ 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

Frame 405 | 2019-10-26T21:23:11.406976Z

📢 SIP NOTIFY

🔗 Call-ID	A39C4DD3-F76D11E9-811EC92B-D2F22CC5@10.10.223.123
🔢 CSeq	105 NOTIFY
📋 Event	telephone-event
📊 State	active

✅ 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

Frame 422 | 2019-10-26T21:23:11.526965Z

📢 SIP NOTIFY

🔗 Call-ID	A39C4DD3-F76D11E9-811EC92B-D2F22CC5@10.10.223.123
🔢 CSeq	106 NOTIFY
📋 Event	telephone-event
📊 State	active

✅ SIP 200 OK

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📊 RTCP SR

📦 Packets	121
📦 Bytes	20328

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

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Frame 483 | 2019-10-26T21:23:11.866944Z

📢 SIP NOTIFY

🔗 Call-ID	A39C4DD3-F76D11E9-811EC92B-D2F22CC5@10.10.223.123
🔢 CSeq	107 NOTIFY
📋 Event	telephone-event
📊 State	active

✅ 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

Frame 496 | 2019-10-26T21:23:11.956935Z

📢 SIP NOTIFY

🔗 Call-ID	A39C4DD3-F76D11E9-811EC92B-D2F22CC5@10.10.223.123
🔢 CSeq	108 NOTIFY
📋 Event	telephone-event
📊 State	active

✅ SIP 200 OK

💡 200 OK — call answered (for INVITE) or registration accepted (for REGISTER); SDP answer completes codec negotiation;

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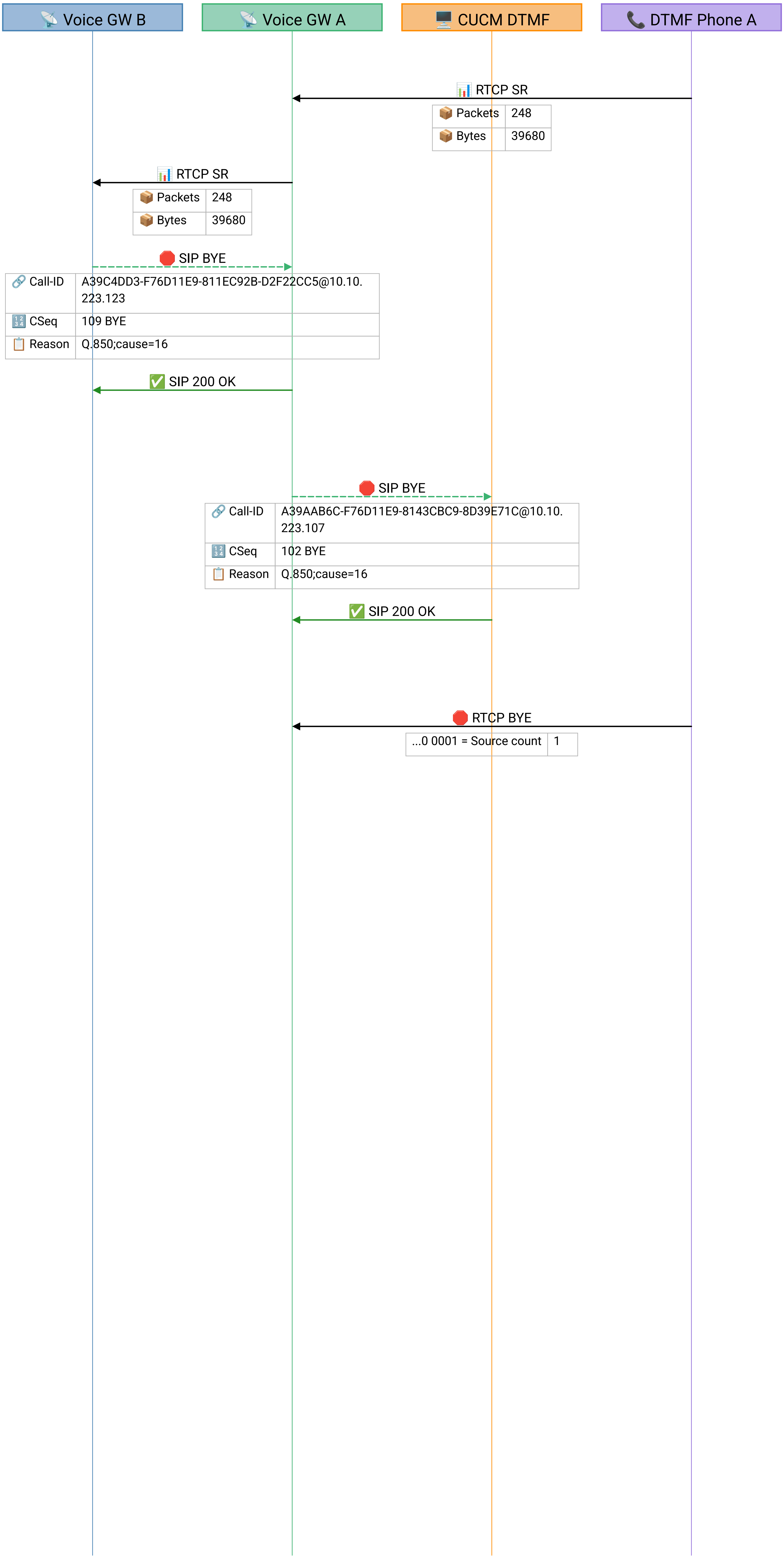

caller must send ACK to complete 3-way handshake

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Frame 1181 | 2019-10-26T21:23:15.438972Z