Exploring dtmf_relay_noasymm.pcapng

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

📁 **dtmf_relay_noasymm.pcapng**

📞 SIP INVITE

📡 Via	SIP/2.0/UDP 10.10.223.123:5060;branch=z9hG4bKD4F03
🔗 Call-ID	5C07DA71-F77311E9-8148C92B-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 2550 8549 IN IP4 10.10.223.123
Session Name (s)	SIP Call
🌐 Connection	IN IP4 10.10.223.123
🎵 Media	audio 16480 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

💡 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=z9hG4bKCB7C6
🔗 Call-ID	5C0B85C3-F77311E9-816DCBC9-8D39E71C@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 1596 8619 IN IP4 10.10.223.107
Session Name (s)	SIP Call
🌐 Connection	IN IP4 10.10.223.107
🎵 Media	audio 8252 RTP/AVP 0 102
🌐 Connection	IN IP4 10.10.223.107
📋 Attr	rtpmap:0 PCMU/8000
📋 Attr	rtpmap:102 telephone-event/8000
📋 Attr	fmtp:102 0-15
📋 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 | 2019-10-26T22:04:05.081981Z

🔗 Call-ID	5C07DA71-F77311E9-8148C92B-D2F22CC5@10.10.223.123
🔢 CSeq	102 PRACK
🔢 RAck	400 101 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 30810 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 | 2019-10-26T22:04:06.260972Z

🔗 Call-ID	5C0B85C3-F77311E9-816DCBC9-8D39E71C@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 8250 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 0-16
📋 Attr	ptime:20

✅ SIP ACK

Frame 12 | 2019-10-26T22:04:06.302978Z

🔗 Call-ID	5C07DA71-F77311E9-8148C92B-D2F22CC5@10.10.223.123
🔢 CSeq	101 ACK

🎵 RTP

Frame 13 | 2019-10-26T22:04:06.330976Z

Setup frame	11
Marker	True
🎵 Codec	ITU-T G.711 PCMU (0)
Sequence number	1107
Timestamp	3010652595
🔢 SSRC	0x1fbfdf7b (532668283)

🎵 RTP

Frame 14 | 2019-10-26T22:04:06.330976Z

Setup frame	9
Marker	True
🎵 Codec	ITU-T G.711 PCMU (0)
Sequence number	1107
Timestamp	3010652595
🔢 SSRC	0x1fbfdf7b (532668283)

🎵 RTP

Frame 21 | 2019-10-26T22:04:06.395975Z

Setup frame	3
Marker	True
🎵 Codec	ITU-T G.711 PCMU (0)
Sequence number	61902
Timestamp	4260423999
🔢 SSRC	0xb50ec4bd (3037643965)

🎵 RTP

Frame 22 | 2019-10-26T22:04:06.395975Z

Setup frame	1
Marker	True
🎵 Codec	ITU-T G.711 PCMU (0)
Sequence number	61902
Timestamp	4260423999
🔢 SSRC	0xb50ec4bd (3037643965)

🔢 DTMF

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

🏁 End	False
⏱️ Duration	80

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🏁 End	False
⏱️ Duration	80

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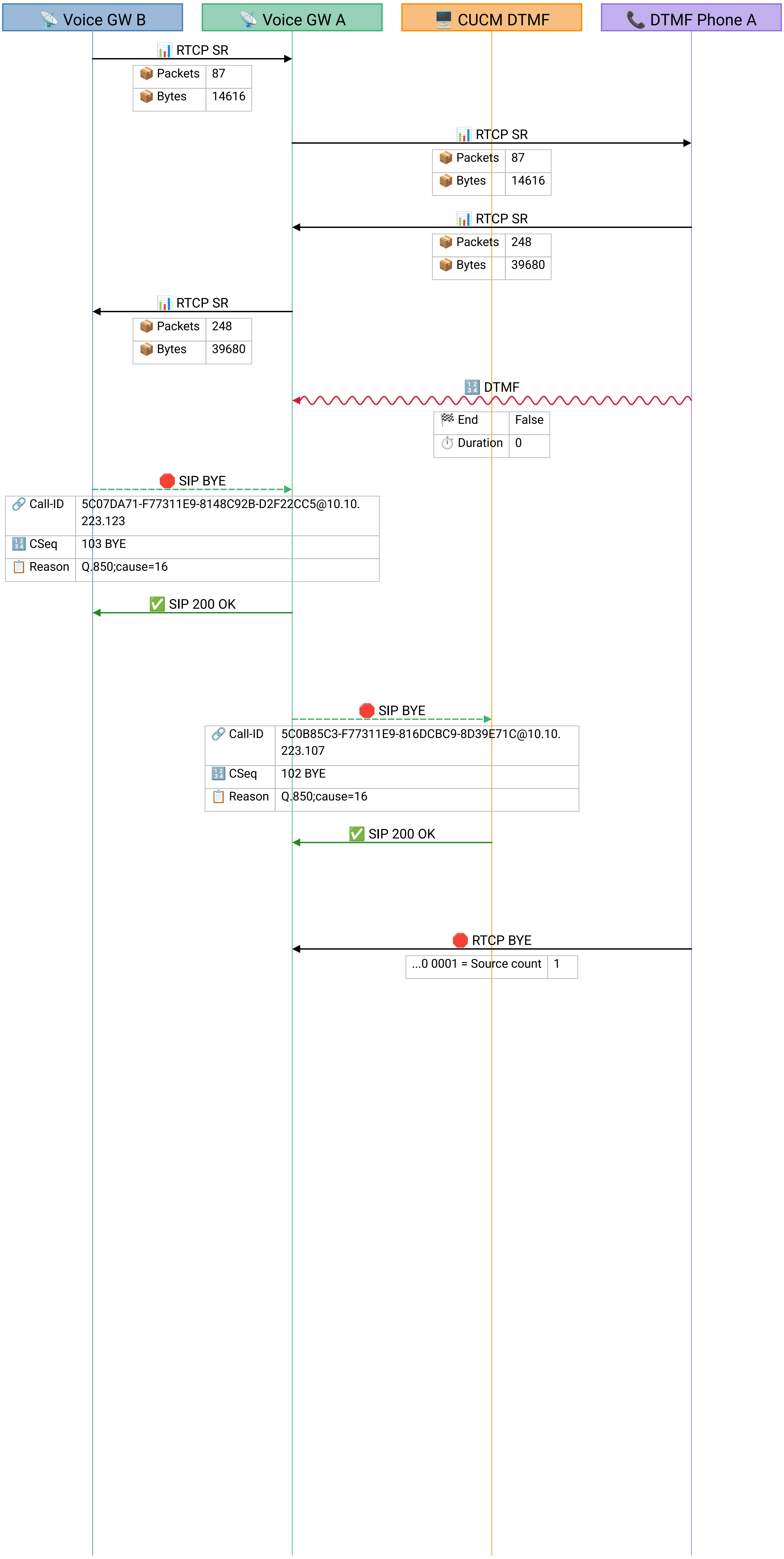

💡 RTCP Sender Report — media sender reports packet/byte counts and NTP timestamp for lip-sync; receivers use this to calculate jitter and loss

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

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

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💡 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 2136 | 2019-10-26T22:04:16.828951Z