

Voice GW B Voice GW A CUCM DTMF DTMF Phone A

dtmf_relay_no_iwrk.pcapng

SIP INVITE

Via	SIP/2.0/UDP 10.10.223.123:5060;branch=z9hG4bKA818F1
Call-ID	183BD61B-F76D11E9-8118C92B-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 3216 8308 IN IP4 10.10.223.123
Session Name (s)	SIP Call
Connection	IN IP4 10.10.223.123
Media	audio 16464 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

Via	SIP/2.0/UDP 10.10.223.107:5060;branch=z9hG4bKB322C2
Call-ID	183A707F-F76D11E9-813DCBC9-8D39E71C@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 5677 8908 IN IP4 10.10.223.107
Session Name (s)	SIP Call
Connection	IN IP4 10.10.223.107
Media	audio 8220 RTP/AVP 0 101
Connection	IN IP4 10.10.223.107
Attr	rtpmap:0 PCMU/8000
Attr	rtpmap:101 telephone-event/8000
Attr	fmt:101 0-16
Attr	ptime:20

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SIP 100 Trying

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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

Call-ID	183BD61B-F76D11E9-8118C92B-D2F22CC5@10.10.223.123
CSeq	102 PRACK
RAck	6890 101 INVITE

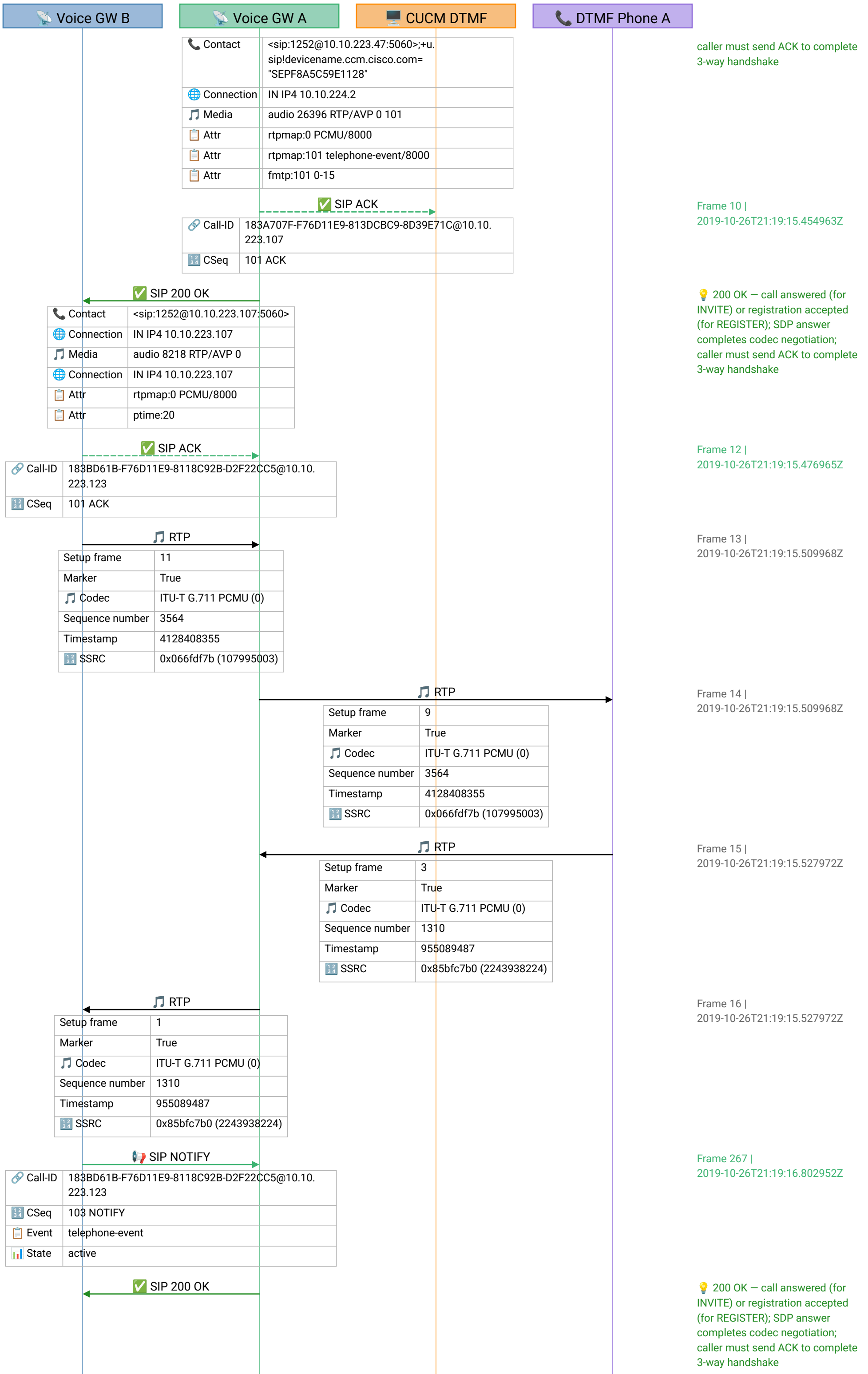
Frame 7 | 2019-10-26T21:19:14.302978Z

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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Frame 10 | 2019-10-26T21:19:15.454963Z

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 12 | 2019-10-26T21:19:15.476965Z

Frame 13 | 2019-10-26T21:19:15.509968Z

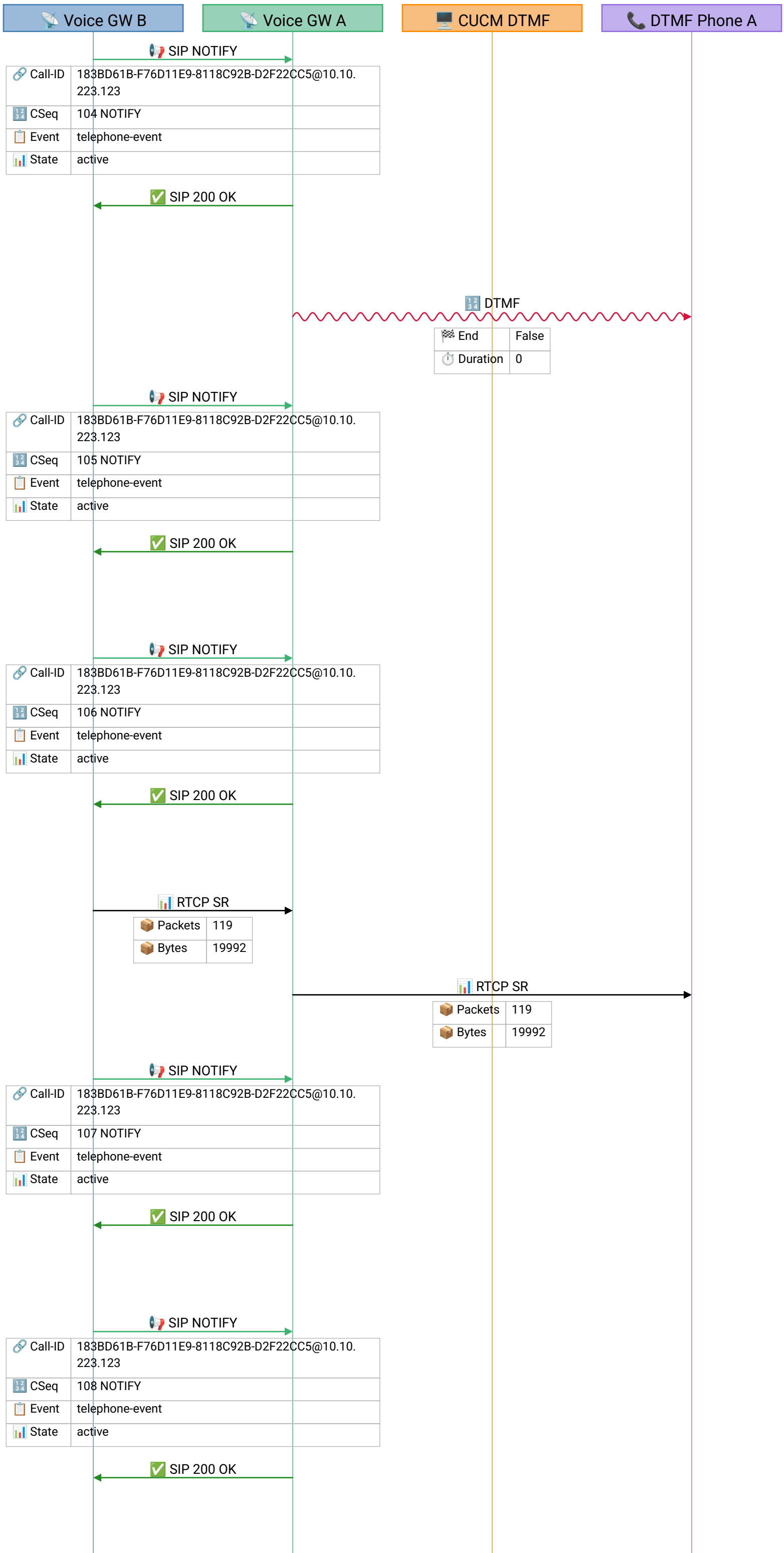
Frame 14 | 2019-10-26T21:19:15.509968Z

Frame 15 | 2019-10-26T21:19:15.527972Z

Frame 16 | 2019-10-26T21:19:15.527972Z

Frame 267 | 2019-10-26T21:19:16.802952Z

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 287 |
2019-10-26T21:19:16.972941Z

💡 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

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

Frame 369 |
2019-10-26T21:19:17.422967Z

💡 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 385 |
2019-10-26T21:19:17.562959Z

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

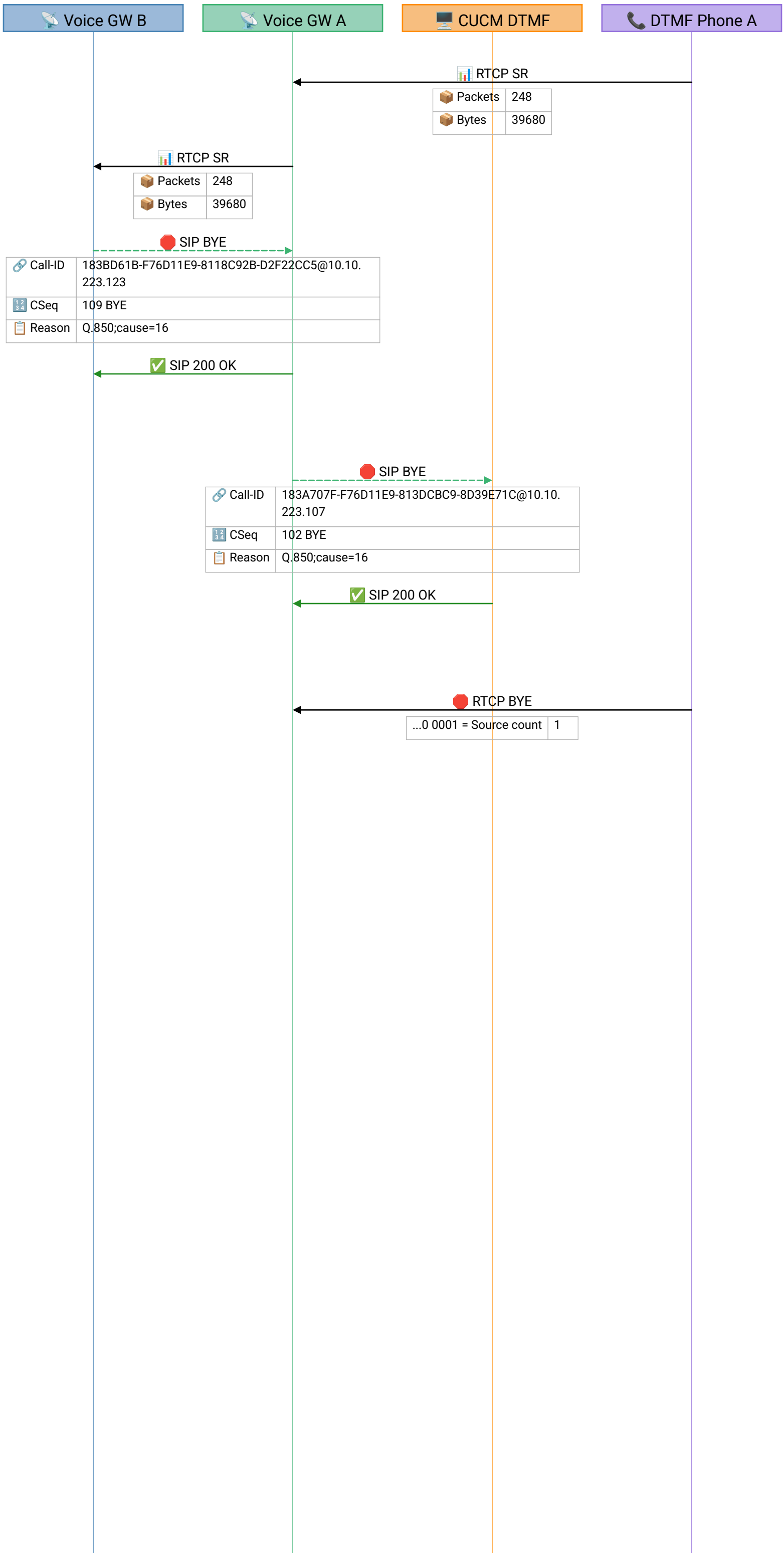
💡 RTCP Sender Report – media sender reports packet/byte counts and NTP timestamp for lip-sync; receivers use this to calculate jitter and loss

Frame 451 |
2019-10-26T21:19:17.922941Z

💡 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 467 |
2019-10-26T21:19:18.052991Z

💡 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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- 💡 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 1297 | 2019-10-26T21:19:22.250978Z