

Voice GW B Voice GW A CUCM DTMF DTMF Phone A

dtmf_relay_iwrk_standard.pcapng

SIP INVITE

Via	SIP/2.0/UDP 10.10.223.123:5060;branch=z9hG4bK9E7FD
Call-ID	BE1D3B07-F76C11E9-8112C92B-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 5385 4914 IN IP4 10.10.223.123
Session Name (s)	SIP Call
Connection	IN IP4 10.10.223.123
Media	audio 16462 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=z9hG4bKB010F1
Call-ID	BE1BE87E-F76C11E9-8137CBC9-8D39E71C@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 7474 5824 IN IP4 10.10.223.107
Session Name (s)	SIP Call
Connection	IN IP4 10.10.223.107
Media	audio 8216 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 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

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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	BE1D3B07-F76C11E9-8112C92B-D2F22CC5@10.10.223.123
CSeq	102 PRACK
RAck	6465 101 INVITE

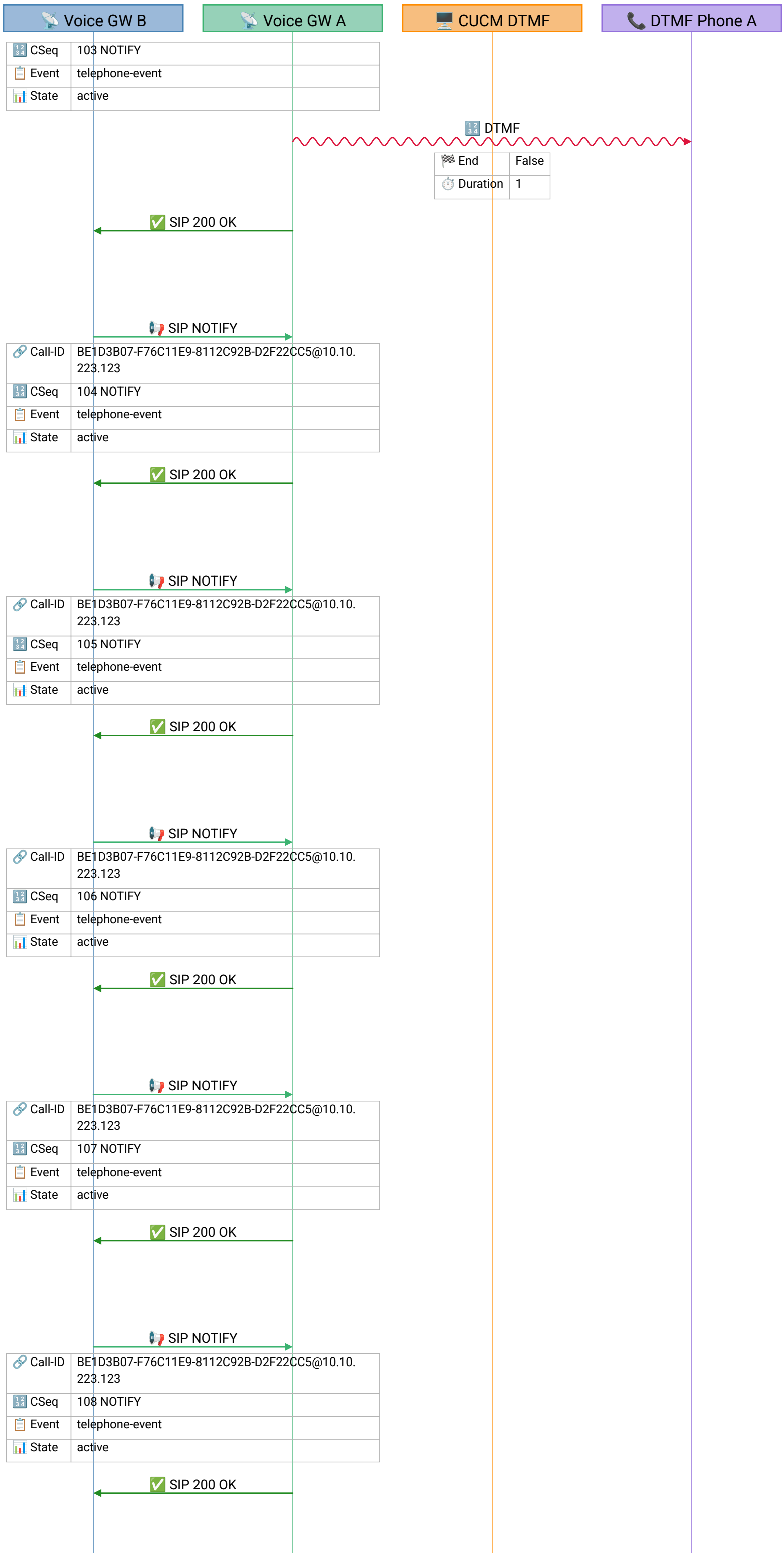
Frame 7 | 2019-10-26T21:16:43.110986Z

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

💡 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 542 |
2019-10-26T21:16:47.008986Z

💡 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 619 |
2019-10-26T21:16:47.448966Z

💡 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 636 |
2019-10-26T21:16:47.568955Z

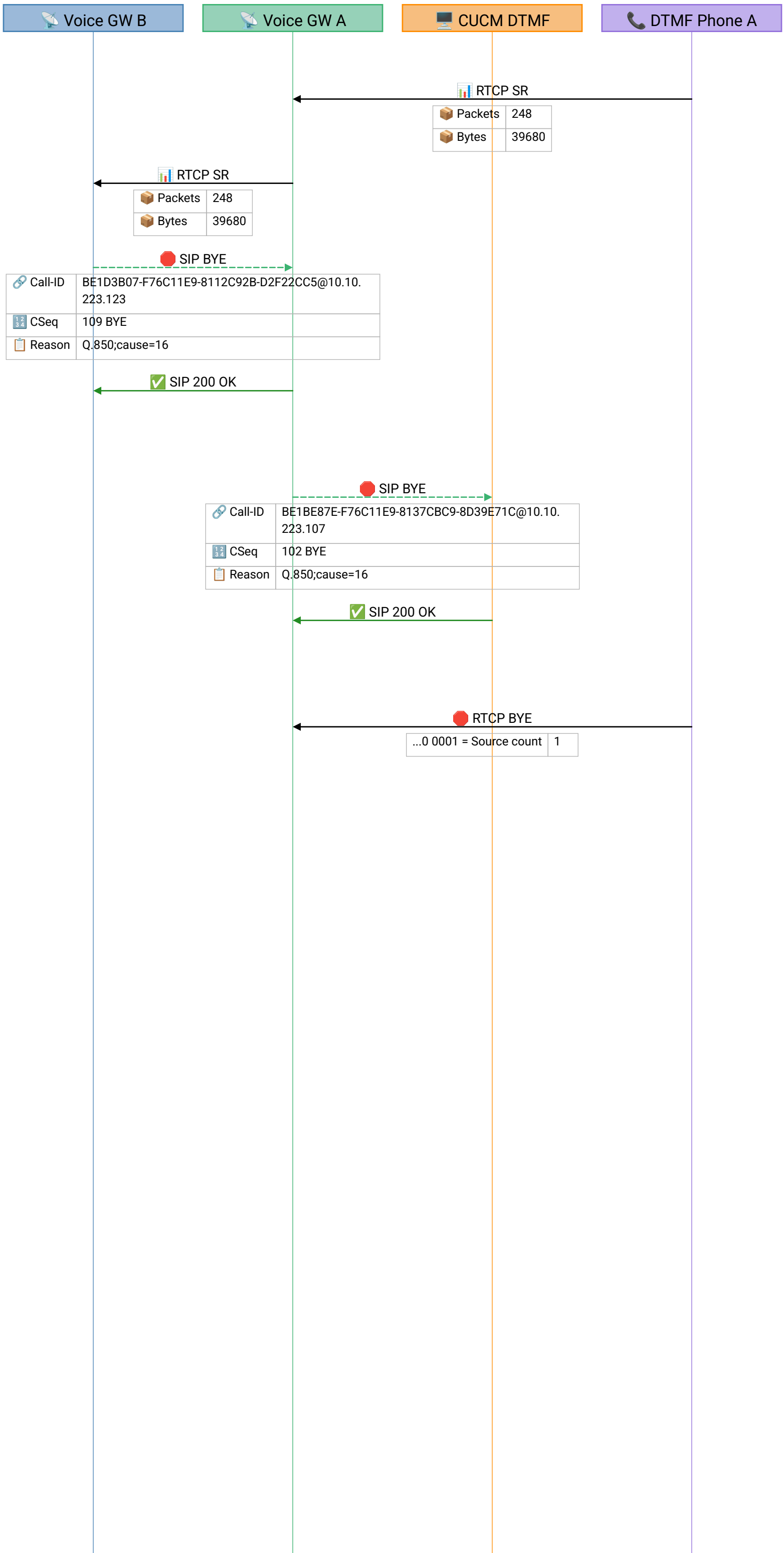
💡 200 OK – call answered (for INVITE) or registration accepted (for REGISTER); SDP answer completes codec negotiation; caller must send ACK to complete 3-way handshake

Frame 695 |
2019-10-26T21:16:47.908949Z

💡 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 712 |
2019-10-26T21:16:48.02899Z

💡 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

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Frame 1379 | 2019-10-26T21:16:51.430977Z