📡 SIP Proxy | 📞 SIP Phone A | 📞 SIP Phone D | 🏠 Router | 📞 SIP Phone B

📁 **rtp_sip_g711alaw_30ms.pcapng**

📞 SIP INVITE

📡 Via	SIP/2.0/UDP 10.180.110.58:5060;branch=z9hG4bK264667
🔗 Call-ID	B92CB3AB-6DAF11E9-9959E3AD-C1443EBC@10.180.110.58
🖥️ UA	Cisco-SIPGateway/IOS-16.9.2
🔢 CSeq	101 INVITE
🔢 Max-Fwd	70
📞 Contact	<sip:10.180.110.58:5060>
👤 Owner	CiscoSystemsSIP-GW-UserAgent 3210 2880 IN IP4 10.180.110.58
Session Name (s)	SIP Call
🌐 Connection	IN IP4 10.180.110.58
🎵 Media	audio 8456 RTP/AVP 8 101
🌐 Connection	IN IP4 10.180.110.58
📋 Attr	rtpmap:8 PCMA/8000
📋 Attr	rtpmap:101 telephone-event/8000
📋 Attr	fmtp:101 0-16
📋 Attr	ptime:30

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

🚫 SIP 422 Session Timer too small

Frame 2 | 2019-05-04T14:28:30.494969Z

📋 SIP REGISTER

🔗 Call-ID	15569800866931884@10.180.110.59
🔢 CSeq	1 REGISTER
📞 Contact	<sip:csta@10.180.110.59:5060;transport=udp>
🖥️ UA	

💡 SIP REGISTER binds a contact URI to an AOR (Address of Record) at the registrar; Expires=0 means unregister; 401 triggers Digest auth challenge

✅ SIP ACK

🔗 Call-ID	B92CB3AB-6DAF11E9-9959E3AD-C1443EBC@10.180.110.58
🔢 CSeq	101 ACK

Frame 4 | 2019-05-04T14:28:30.495961Z

📞 SIP INVITE

📡 Via	SIP/2.0/UDP 10.180.110.58:5060;branch=z9hG4bK2651C51
🔗 Call-ID	B92CB3AB-6DAF11E9-9959E3AD-C1443EBC@10.180.110.58
🖥️ UA	Cisco-SIPGateway/IOS-16.9.2
🔢 CSeq	102 INVITE
🔢 Max-Fwd	70
📞 Contact	<sip:10.180.110.58:5060>
👤 Owner	CiscoSystemsSIP-GW-UserAgent 3210 2880 IN IP4 10.180.110.58
Session Name (s)	SIP Call
🌐 Connection	IN IP4 10.180.110.58
🎵 Media	audio 8456 RTP/AVP 8 101
🌐 Connection	IN IP4 10.180.110.58
📋 Attr	rtpmap:8 PCMA/8000
📋 Attr	rtpmap:101 telephone-event/8000
📋 Attr	fmtp:101 0-16
📋 Attr	ptime:30

💡 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

📡 SIP Proxy | 📞 SIP Phone A | 📞 SIP Phone D | 🏠 Router | 📞 SIP Phone B

✅ SIP 200 OK

📞 Contact	<sip:1234@10.180.110.48:5060>;+u.sip!devicename.ccm.cisco.com="SEP2C31246A214B"
🌐 Connection	IN IP4 10.10.214.57
🎵 Media	audio 22682 RTP/AVP 8 101
📋 Attr	rtpmap:8 PCMA/8000
📋 Attr	rtpmap:101 telephone-event/8000
📋 Attr	fmtp:101 0-15

💡 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	B92CB3AB-6DAF11E9-9959E3AD-C1443EBC@10.180.110.58
🔢 CSeq	102 ACK

Frame 9 | 2019-05-04T14:28:31.583969Z

🎵 RTP

Setup frame	5
Marker	False
🎵 Codec	ITU-T G.711 PCMA (8)
Sequence number	9431
Timestamp	2398809032
🔢 SSRC	0x00007a3e (31294)

Frame 10 | 2019-05-04T14:28:31.626966Z

🎵 RTP

Setup frame	5
Marker	True
🎵 Codec	ITU-T G.711 PCMA (8)
Sequence number	14524
Timestamp	2940969109
🔢 SSRC	0x97d5b2f9 (2547364601)

Frame 14 | 2019-05-04T14:28:31.744956Z

📋 SIP REGISTER

🔗 Call-ID	15569800866931884@10.180.110.59
🔢 CSeq	1 REGISTER
📞 Contact	<sip:csta@10.180.110.59:5060;transport=udp>
🖥️ UA	

💡 SIP REGISTER binds a contact URI to an AOR (Address of Record) at the registrar; Expires=0 means unregister; 401 triggers Digest auth challenge