



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	fntp: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

SIP REGISTER

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

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

💡 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

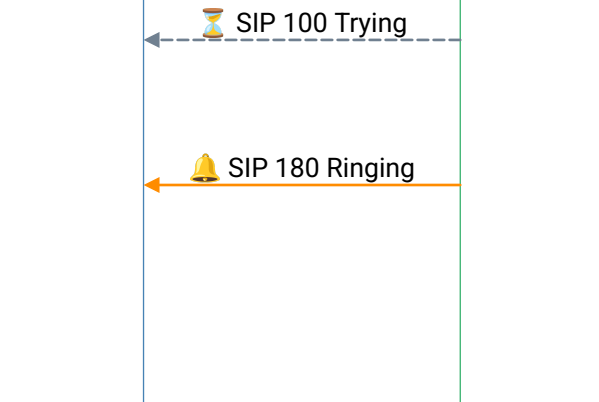
Call-ID	B92CB3AB-6DAF11E9-9959E3AD-C1443EBC@10.180.110.58
CSeq	101 ACK

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	fntp: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



💡 100 Trying – proxy accepted the request and is searching for the callee; stops retransmission timer; no end-to-end significance

💡 180 Ringing – callee's phone is alerting; caller hears locally generated ringback; 183 with SDP offers early media (in-band ringback) instead



✓ SIP 200 OK

Contact	<sip:1234@10.180.110.48:5060>;+u.sipdeviceName.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	fmp: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