

rtp_sip_g711ulaw_10ms.pcapng

SIP INVITE

Via	SIP/2.0/UDP 10.180.110.58:5060;branch=z9hG4bK25B1258
Call-ID	2B452FEA-6DAF11E9-9946E3AD-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 1183 5020 IN IP4 10.180.110.58
Session Name (s)	SIP Call
Connection	IN IP4 10.180.110.58
Media	audio 8450 RTP/AVP 0 101
Connection	IN IP4 10.180.110.58
Attr	rtpmap:0 PCMU/8000
Attr	rtpmap:101 telephone-event/8000
Attr	fmp:101 0-16
Attr	ptime:10

💡 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 ACK

Call-ID	2B452FEA-6DAF11E9-9946E3AD-C1443EBC@10.180.110.58
CSeq	101 ACK

Frame 2 | 2019-05-04T14:24:32.419976Z

Frame 3 | 2019-05-04T14:24:32.420968Z

SIP INVITE

Via	SIP/2.0/UDP 10.180.110.58:5060;branch=z9hG4bK25C629
Call-ID	2B452FEA-6DAF11E9-9946E3AD-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 1183 5020 IN IP4 10.180.110.58
Session Name (s)	SIP Call
Connection	IN IP4 10.180.110.58
Media	audio 8450 RTP/AVP 0 101
Connection	IN IP4 10.180.110.58
Attr	rtpmap:0 PCMU/8000
Attr	rtpmap:101 telephone-event/8000
Attr	fmp:101 0-16
Attr	ptime:10

💡 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

SIP 180 Ringing

SIP 200 OK

💡 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

💡 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:1234@10.180.110.48:5060>;+u.sip!devicename.ccm.cisco.com="SEP2C31246A214B"
Connection	IN IP4 10.10.214.57
Media	audio 24878 RTP/AVP 0 101
Attr	rtpmap:0 PCMU/8000
Attr	rtpmap:101 telephone-event/8000
Attr	fmp:101 0-15

SIP ACK

Call-ID	2B452FEA-6DAF11E9-9946E3AD-C1443EBC@10.180.110.58
CSeq	102 ACK

Frame 8 | 2019-05-04T14:24:33.506962Z

RTP

Setup frame	4
Marker	False
Codec	ITU-T G.711 PCMU (0)

Frame 9 | 2019-05-04T14:24:33.535967Z

SIP Proxy

SIP Phone A

SIP Phone B

Sequence number	5601
Timestamp	1710275606
SSRC	0x00007a50 (31312)

RTP

Setup frame	4
Marker	True
Codec	ITU-T G.711 PCMU (0)
Sequence number	24130
Timestamp	1747414691
SSRC	0x82913f1e (2190556958)

Frame 21 |
2019-05-04T14:24:33.64996Z