Exploring sip_dtmf_nte_rfc4733.pcapng

📡 SIP Proxy | 📞 SIP Phone A | 📞 SIP Phone B

📁 **sip_dtmf_nte_rfc4733.pcapng**

📞 SIP INVITE

📡 Via	SIP/2.0/UDP 10.180.110.58:5060;branch=z9hG4bK22A664
🔗 Call-ID	196B51FB-6D0C11E9-97A4E3AD-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 9721 2207 IN IP4 10.180.110.58
Session Name (s)	SIP Call
🌐 Connection	IN IP4 10.180.110.58
🎵 Media	audio 8384 RTP/AVP 0 101
🌐 Connection	IN IP4 10.180.110.58
📋 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

💡 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 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 25126 RTP/AVP 0 101
📋 Attr	rtpmap:0 PCMU/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	196B51FB-6D0C11E9-97A4E3AD-C1443EBC@10.180.110.58
🔢 CSeq	101 ACK

Frame 5 | 2019-05-03T18:57:20.807956Z

🔢 DTMF

🏁 End	False
⏱️ Duration	0

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