Exploring dtmf_relay_digitdrop.pcapng

📡 Voice GW B | 📡 Voice GW A | 🖥️ CUCM DTMF | 📞 DTMF Phone A

📁 **dtmf_relay_digitdrop.pcapng**

📞 SIP INVITE

📡 Via	SIP/2.0/UDP 10.10.223.123:5060;branch=z9hG4bK7B1CC
🔗 Call-ID	1D24FBB5-F76A11E9-80F4C92B-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 9861 5101 IN IP4 10.10.223.123
Session Name (s)	SIP Call
🌐 Connection	IN IP4 10.10.223.123
🎵 Media	audio 16452 RTP/AVP 0 101
🌐 Connection	IN IP4 10.10.223.123
📋 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 INVITE

📡 Via	SIP/2.0/UDP 10.10.223.107:5060;branch=z9hG4bK9D208E
🔗 Call-ID	1D24661B-F76A11E9-8113CBC9-8D39E71C@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 8691 6419 IN IP4 10.10.223.107
Session Name (s)	SIP Call
🌐 Connection	IN IP4 10.10.223.107
🎵 Media	audio 8184 RTP/AVP 0
🌐 Connection	IN IP4 10.10.223.107
📋 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 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 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 PRACK

🔗 Call-ID	1D24FBB5-F76A11E9-80F4C92B-D2F22CC5@10.10.223.123
🔢 CSeq	102 PRACK
🔢 RAck	9490 101 INVITE

Frame 7 | 2019-10-26T20:57:54.02899Z

✅ 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

✅ SIP 200 OK

💡 200 OK — call answered (for INVITE) or registration accepted (for REGISTER); SDP answer completes codec negotiation;

📡 Voice GW B | 📡 Voice GW A | 🖥️ CUCM DTMF | 📞 DTMF Phone A

📞 Contact	<sip:1252@10.10.223.47:5060>;+u.sip!devicename.ccm.cisco.com="SEPF8A5C59E1128"
🌐 Connection	IN IP4 10.10.224.2
🎵 Media	audio 21432 RTP/AVP 0
📋 Attr	rtpmap:0 PCMU/8000

caller must send ACK to complete 3-way handshake

✅ SIP ACK

Frame 10 | 2019-10-26T20:57:55.137978Z

🔗 Call-ID	1D24661B-F76A11E9-8113CBC9-8D39E71C@10.10.223.107
🔢 CSeq	101 ACK

✅ 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

📞 Contact	<sip:1252@10.10.223.107:5060>
🌐 Connection	IN IP4 10.10.223.107
🎵 Media	audio 8182 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 ACK

Frame 12 | 2019-10-26T20:57:55.158988Z

🔗 Call-ID	1D24FBB5-F76A11E9-80F4C92B-D2F22CC5@10.10.223.123
🔢 CSeq	101 ACK

🎵 RTP

Frame 13 | 2019-10-26T20:57:55.193974Z

Setup frame	11
Marker	True
🎵 Codec	ITU-T G.711 PCMU (0)
Sequence number	7713
Timestamp	1802622694
🔢 SSRC	0x090ddf7b (151904123)

🎵 RTP

Frame 14 | 2019-10-26T20:57:55.193974Z

Setup frame	9
Marker	True
🎵 Codec	ITU-T G.711 PCMU (0)
Sequence number	7713
Timestamp	1802622694
🔢 SSRC	0x090ddf7b (151904123)

🎵 RTP

Frame 25 | 2019-10-26T20:57:55.295974Z

Setup frame	3
Marker	True
🎵 Codec	ITU-T G.711 PCMU (0)
Sequence number	23770
Timestamp	3733792091
🔢 SSRC	0xc639c9d4 (3325675988)

🎵 RTP

Frame 26 | 2019-10-26T20:57:55.295974Z

Setup frame	1
Marker	True
🎵 Codec	ITU-T G.711 PCMU (0)
Sequence number	23770
Timestamp	3733792091
🔢 SSRC	0xc639c9d4 (3325675988)

🔢 DTMF

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

🏁 End	False
⏱️ Duration	80

📢 SIP NOTIFY

Frame 350 | 2019-10-26T20:57:56.934949Z

🔗 Call-ID	1D24661B-F76A11E9-8113CBC9-8D39E71C@10.10.223.107
🔢 CSeq	102 NOTIFY
📋 Event	telephone-event
📊 State	active

📡 Voice GW B | 📡 Voice GW A | 🖥️ CUCM DTMF | 📞 DTMF Phone A

✅ 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

📢 SIP NOTIFY

🔗 Call-ID	1D24661B-F76A11E9-8113CBC9-8D39E71C@10.10.223.107
🔢 CSeq	103 NOTIFY
📋 Event	telephone-event
📊 State	active

Frame 369 | 2019-10-26T20:57:57.063992Z

✅ 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

📢 SIP NOTIFY

🔗 Call-ID	1D24661B-F76A11E9-8113CBC9-8D39E71C@10.10.223.107
🔢 CSeq	104 NOTIFY
📋 Event	telephone-event
📊 State	active

Frame 462 | 2019-10-26T20:57:57.553956Z

✅ 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

📢 SIP NOTIFY

🔗 Call-ID	1D24661B-F76A11E9-8113CBC9-8D39E71C@10.10.223.107
🔢 CSeq	105 NOTIFY
📋 Event	telephone-event
📊 State	active

Frame 479 | 2019-10-26T20:57:57.663951Z

✅ 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

📢 SIP NOTIFY

🔗 Call-ID	1D24661B-F76A11E9-8113CBC9-8D39E71C@10.10.223.107
🔢 CSeq	106 NOTIFY
📋 Event	telephone-event
📊 State	active

Frame 564 | 2019-10-26T20:57:58.10499Z

✅ 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

📢 SIP NOTIFY

🔗 Call-ID	1D24661B-F76A11E9-8113CBC9-8D39E71C@10.10.223.107
🔢 CSeq	107 NOTIFY
📋 Event	telephone-event
📊 State	active

Frame 581 | 2019-10-26T20:57:58.223987Z

✅ 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

📊 RTCP SR

📦 Packets	176
📦 Bytes	29568

💡 RTCP Sender Report — media sender reports packet/byte counts and NTP timestamp for lip-sync; receivers use this to calculate jitter and loss

📊 RTCP SR

💡 RTCP Sender Report — media sender reports packet/byte counts and NTP timestamp for lip-sync;

Voice GW B | Voice GW A | CUCM DTMF | DTMF Phone A

📦 Packets	176
📦 Bytes	29568

receivers use this to calculate jitter and loss

📊 RTCP SR

📦 Packets	248
📦 Bytes	39680

💡 RTCP Sender Report — media sender reports packet/byte counts and NTP timestamp for lip-sync; receivers use this to calculate jitter and loss

📊 RTCP SR

📦 Packets	248
📦 Bytes	39680

💡 RTCP Sender Report — media sender reports packet/byte counts and NTP timestamp for lip-sync; receivers use this to calculate jitter and loss

🛑 SIP BYE

🔗 Call-ID	1D24FBB5-F76A11E9-80F4C92B-D2F22CC5@10.10.223.123
🔢 CSeq	103 BYE
📋 Reason	Q.850;cause=16

💡 SIP BYE terminates an established dialog — both sides send BYE to release media resources; Reason header explains why (e.g., user hangup)

✅ 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

🛑 SIP BYE

🔗 Call-ID	1D24661B-F76A11E9-8113CBC9-8D39E71C@10.10.223.107
🔢 CSeq	108 BYE
📋 Reason	Q.850;cause=16

💡 SIP BYE terminates an established dialog — both sides send BYE to release media resources; Reason header explains why (e.g., user hangup)

✅ 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

🛑 RTCP BYE

...0 0001 = Source count	1

Frame 1456 | 2019-10-26T20:58:02.620954Z