Exploring rtp_sip_opus.pcapng

🎵 Media E	🎵 Media B	🎵 Media A

📁 **rtp_sip_opus.pcapng**

📡 SIP INVITE sip:2f231a59-524c-49d3-a058-9d6d4317a18d@10.10.214.57:51546;transport=tcp SIP/2.0

Frame 1 | 2019-08-29T21:22:04.491788Z

📡 Via	SIP/2.0/TCP 172.18.110.48:5060;branch=z9hG4bK1045455dd08
🔗 Call-ID	ab3eb80-1ea1e8a6-1018c-306e12ac@172.18.110.48
🔢 CSeq	101 INVITE
📞 Contact	<sip:1026@172.18.110.48:5060;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP0C75BD110CA4"

⏳ SIP 100 Trying

Frame 2 | 2019-08-29T21:22:04.495095Z

🔔 SIP 180 Ringing

💡 180 Ringing — callee's phone is alerting; caller hears locally generated ringback

✅ SIP 200 OK

💡 200 OK — call answered or registration accepted; SDP answer completes codec negotiation; ACK required to complete INVITE 3-way handshake

📞 Contact	<sip:2f231a59-524c-49d3-a058-9d6d4317a18d@10.10.214.57:51546;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP2C31246A214B"
🎵 Media	audio 22018 RTP/AVP 114 9 124 113 115 0 8 116 18 101
🌐 Connection	IN IP4 10.10.214.57

📡 SIP ACK sip:2f231a59-524c-49d3-a058-9d6d4317a18d@10.10.214.57:51546;transport=tcp SIP/2.0

Frame 6 | 2019-08-29T21:22:05.641962Z

📡 Via	SIP/2.0/TCP 172.18.110.48:5060;branch=z9hG4bK10455116ad69f
🔗 Call-ID	ab3eb80-1ea1e8a6-1018c-306e12ac@172.18.110.48
🔢 CSeq	101 ACK
👤 Owner	CiscoSystemsCCM-SIP 234985 1 IN IP4 172.18.110.48
🌐 Connection	IN IP4 10.10.214.56
🎵 Media	audio 22018 RTP/AVP 114 101
📋 Attr	rtpmap:114 opus/48000/2
📋 Attr	fmtp:114 maxplaybackrate=16000;sprop-maxcapturerate=16000;maxaveragebitrate=64000;stereo=0;sprop-stereo=0;usedtx=0
📋 Attr	rtpmap:101 telephone-event/8000
📋 Attr	fmtp:101 0-15

📡 SIP UPDATE sip:2f231a59-524c-49d3-a058-9d6d4317a18d@10.10.214.57:51546;transport=tcp SIP/2.0

Frame 7 | 2019-08-29T21:22:05.647624Z

📡 Via	SIP/2.0/TCP 172.18.110.48:5060;branch=z9hG4bK1045742670f3c
🔗 Call-ID	ab3eb80-1ea1e8a6-1018c-306e12ac@172.18.110.48
🔢 CSeq	102 UPDATE
📞 Contact	<sip:c28713b5-c24d-40f2-bca1-cc485c749fdf@172.18.110.48:5060;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP0C75BD110CA4"

✅ SIP 200 OK

💡 200 OK — call answered or registration accepted; SDP answer completes codec negotiation; ACK required to complete INVITE 3-way handshake

📞 Contact	<sip:2f231a59-524c-49d3-a058-9d6d4317a18d@10.10.214.57:51546;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP2C31246A214B"

🎵 RTP

Frame 9 | 2019-08-29T21:22:05.785365Z

Setup frame	1
Marker	True
🎵 Codec	opus (114)
Sequence number	54086
Timestamp	2101902535
🔢 SSRC	0xf7835471 (4152579185)

🎵 RTP

Frame 11 | 2019-08-29T21:22:05.807122Z

Setup frame	6
Marker	True
🎵 Codec	opus (114)
Sequence number	54086
Timestamp	2101902535
🔢 SSRC	0xfd1fd070 (4246720624)

🔢 DTMF

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

🏁 End	False

🎵 Media E | 🎵 Media B | 🎵 Media A

⏱️ Duration	0

transcoding

📡 SIP BYE sip:2f231a59-524c-49d3-a058-9d6d4317a18d@10.10.214.57:51546;transport=tcp SIP/2.0

Frame 1244 | 2019-08-29T21:22:17.415752Z

📡 Via	SIP/2.0/TCP 172.18.110.48:5060;branch=z9hG4bK104583f98d8fc
🔗 Call-ID	ab3eb80-1ea1e8a6-1018c-306e12ac@172.18.110.48
🔢 CSeq	103 BYE

✅ SIP 200 OK

💡 200 OK — call answered or registration accepted; SDP answer completes codec negotiation; ACK required to complete INVITE 3-way handshake