

sip-short-call-with-rtp.pcap

SIP INVITE

Via	SIP/2.0/UDP 192.168.0.10:59205;branch=z9hG4bKc0a8000a052181bc3f7ea3253c;rport
From	sip:E646657195201@talk4free.com
To	sip:9055551212@talk4free.com
Contact	<sip:E646657195201@192.168.0.10:59205>
Call-ID	C5570127C1A6A1ABF7ED9DB9AD608CE00xc0a8000a
CSeq	1 INVITE
Max-Forwards	70
User-Agent	mJ/2.00.632b.11054E4
SDP-Owner	-2209074887 2209074887 IN IP4 192.168.0.10
SDP-Session	SJphone
SDP-Conn	IN IP4 192.168.0.10
Media	audio 49154 RTP/AVP 0 8 101 13
SDP-Conn	IN IP4 192.168.0.10
Media-Attr	ptime:30
Media-Attr	rtpmap:0 PCMU/8000...
	6 more

💡 [Frame 46] SIP INVITE carries an SDP offer describing media capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams

SIP 100 Trying

To	sip:9055551212@talk4free.com
From	sip:E646657195201@talk4free.com
Call-ID	C5570127C1A6A1ABF7ED9DB9AD608CE00xc0a8000a
CSeq	1 INVITE

Frame 47 | 2012-04-12T15:40:15.755652Z

SIP 401 Unauthorized

To	sip:9055551212@talk4free.com
From	sip:E646657195201@talk4free.com
Call-ID	C5570127C1A6A1ABF7ED9DB9AD608CE00xc0a8000a
CSeq	1 INVITE
Auth-Scheme	Digest
Nonce	"30da0aed2_12170"
Realm	"stratus.com"
Algorithm	MD5

Frame 48 | 2012-04-12T15:40:15.769396Z

SIP ACK

From	sip:E646657195201@talk4free.com
To	sip:9055551212@talk4free.com
Call-ID	C5570127C1A6A1ABF7ED9DB9AD608CE00xc0a8000a
CSeq	1 ACK

Frame 49 | 2012-04-12T15:40:15.882668Z

SIP INVITE

Via	SIP/2.0/UDP 192.168.0.10:59205;branch=z9hG4bKc0a8000a052182706faf2cbf3d;rport
From	sip:E646657195201@talk4free.com
To	sip:9055551212@talk4free.com
Contact	<sip:E646657195201@192.168.0.10:59205>
Call-ID	C5570127C1A6A1ABF7ED9DB9AD608CE00xc0a8000a
CSeq	2 INVITE
Max-Forwards	70
User-Agent	mJ/2.00.632b.11054E4
SDP-Owner	-2209074887 2209074887 IN IP4 192.168.0.10
SDP-Session	SJphone
SDP-Conn	IN IP4 192.168.0.10
Media	audio 49154 RTP/AVP 0 8 101 13
SDP-Conn	IN IP4 192.168.0.10
Media-Attr	ptime:30
Media-Attr	rtpmap:0 PCMU/8000...
	6 more

💡 [Frame 50] SIP INVITE carries an SDP offer describing media capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams

SIP 100 Trying

To	sip:9055551212@talk4free.com
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Frame 51 | 2012-04-12T15:40:15.931983Z

MagicJack

SIP Server

Media Server

From	sip:E646657195201@talk4free.com
Call-ID	C5570127C1A6A1ABF7ED9DB9AD608CE00xc0a8000a
CSeq	2 INVITE

SIP 183 Session Progress

To	sip:9055551212@talk4free.com
From	sip:E646657195201@talk4free.com
Call-ID	C5570127C1A6A1ABF7ED9DB9AD608CE00xc0a8000a
CSeq	2 INVITE
SDP-Owner	- 819596013 819596013 IN IP4 216.234.64.8
SDP-Conn	IN IP4 216.234.64.16
Media	audio 54550 RTP/AVP 0 101

RTP

Setup-Frame	54
Marker	True
PayloadType	ITU-T G.711 PCMU (0)
Seq	26528
Timestamp	0
SSRC	0x2a173650 (706164304)

RTP

Setup-Frame	50
Marker	False
PayloadType	ITU-T G.711 PCMU (0)
Seq	18437
Timestamp	1769305803
SSRC	0x31be1e0e (834543118)

SIP 200 OK

Contact	<sip:9055551212@216.234.64.8:5070>
To	sip:9055551212@talk4free.com
From	sip:E646657195201@talk4free.com
Call-ID	C5570127C1A6A1ABF7ED9DB9AD608CE00xc0a8000a
CSeq	2 INVITE
SDP-Owner	- 819596013 819596013 IN IP4 216.234.64.8
SDP-Conn	IN IP4 216.234.64.16
SDP-Time	0 0
Media	audio 54550 RTP/AVP 0 101
Media-Attr	rtpmap:0 PCMU/8000
Media-Attr	rtpmap:101 telephone-event/8000
Media-Attr	fntp:101 0-11
Media-Attr	ptime:20
Media-Attr	setup:active
Media-Attr	sendrecv

SIP ACK

From	sip:E646657195201@talk4free.com
To	sip:9055551212@talk4free.com
Call-ID	C5570127C1A6A1ABF7ED9DB9AD608CE00xc0a8000a
CSeq	2 ACK

SIP BYE

To	sip:E646657195201@talk4free.com
From	sip:9055551212@talk4free.com
Call-ID	C5570127C1A6A1ABF7ED9DB9AD608CE00xc0a8000a
CSeq	1217001 BYE

SIP 200 OK

From	sip:9055551212@talk4free.com
To	sip:E646657195201@talk4free.com
Contact	<sip:E646657195201@192.168.0.10:59205>
Call-ID	C5570127C1A6A1ABF7ED9DB9AD608CE00xc0a8000a
CSeq	1217001 BYE
Server	mJ/2.00.632b.11054E4

Frame 54 | 2012-04-12T15:40:22.700515Z

Frame 55 | 2012-04-12T15:40:22.765593Z

Frame 59 | 2012-04-12T15:40:22.82158Z

[Frame 925] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

Frame 953 | 2012-04-12T15:40:31.720574Z

[Frame 1324] SIP BYE terminates an established dialog; either party can send it, and the receiving side responds with 200 OK to confirm teardown of the RTP media streams and release resources

[Frame 1329] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin