SIP Client | Media Gateway | SIP Server

SIP call setup with authentication

This call flow shows the SIP call setup between a SIP client (192.168.0.10) and a SIP server (216.234.64.8). The flow also shows the RTP message flow between the SIP client and the Media Gateway (216.234.64.16).

The example covers the following: (1) SIP invite from the client. (2) The SIP server challenges the client to authenticate. (3) The client responds to the authentication challenge. (4) The call is connected. (5) The call enters the conversation phase with RTP traffic. (6) The SIP call is cleared.

Generated with EventStudio (http://www.eventhelix.com/eventstudio/) and VisualEther (http://www.eventhelix.com/visualether/)

Note: You can SIP and RTP message titles in this flow to see complete field level details.

Initiate call

The user initiates a call.

allocate
RTP UDP Port 49154

The client allocates an RTP port. RTP packets will be sent on this port.

Initiating a call with a SIP INVITE with SDP information

SIP INVITE sip:9055551212@talk4free.com SIP/2.0

SIP from address: sip:E646657195201@talk4free.com,
CSeq: 1 INVITE,
Owner/Creator, Session Id (o): - 2209074887 2209074887 IN IP4 192.168.0.10,
Connection Information (c): IN IP4 192.168.0.10,
Time Description, active time (t): 0 0,
Media Description, name and address (m): audio 49154 RTP/AVP 0 8 101 13,
Connection Information (c): IN IP4 192.168.0.10,
Media Attribute (a): ptime:30,
Media Attribute (a): rtpmap:0 PCMU/8000,
Media Attribute (a): rtpmap:8 PCMA/8000,
Media Attribute (a): rtpmap:101 telephone-event/8000,
Media Attribute (a): fmtp:101 0-16,
Media Attribute (a): rtpmap:13 CN/8000,
Media Attribute (a): setup:active,
Media Attribute (a): sendrecv

The SIP client (192.168.0.10) sends a SIP Invite to a SIP Server (216.234.64.8) to initiate the call.

`o=2209074887 2209074887 IN IP4 192.168.0.10`
Specifies that the caller is 2209074887 with 2209074887 session id. The caller uses the Internet (IN). The IPv4 address for the caller is also included.

`m=audio 49154 RTP/AVP 0 8 101 13`
Specifies that port number 49154 is assigned for audio with a list of supported media type formats (0, 8, 101 and 13).

`c=IN IP4 192.168.0.10`
The connection information specifies the connection initiator's IP address.

`a=ptime:30`
Gives the length of time in milliseconds represented by the media in a packet.

`a=rtpmap:0 PCMU/8000`
`a=rtpmap:8 PCMA/8000`
Specifies PMC mu-law (media type: 0) and A-law (media type: 8) codec supported at 8000 samples per second.

`a=rtpmap:101 telephone-event/8000`
`a=fmtp:101 0-16`
Specifies that the payload format 101 supports DTMF digits.

`a=rtpmap:13 CN/8000`
Specifies that payload format 13 is used for comfort noise.

`a=sendrecv`
Specifies that the session will be sending and receiving media.

SIP/2.0 100 Trying

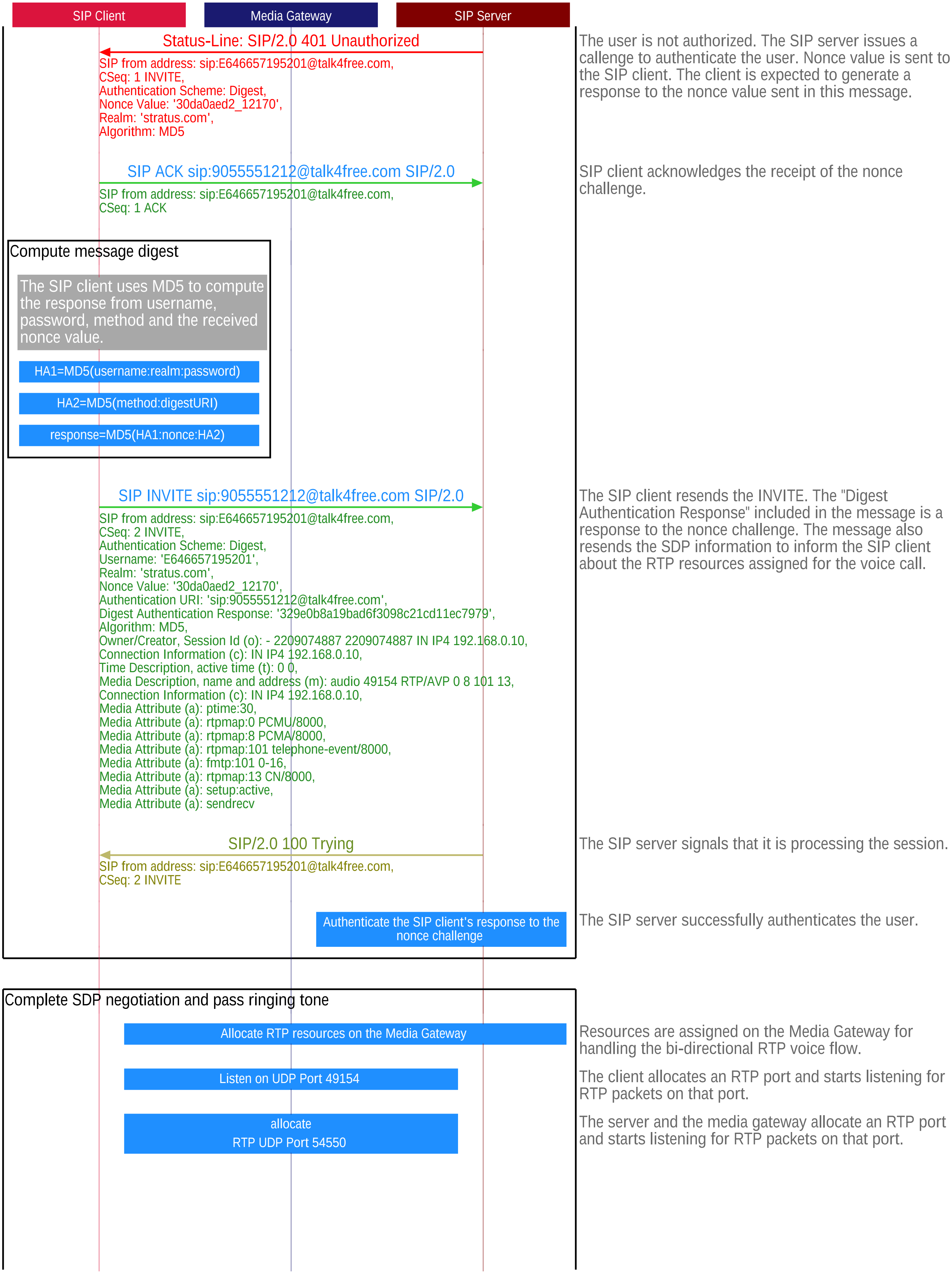
SIP from address: sip:E646657195201@talk4free.com,
CSeq: 1 INVITE

The SIP server acknowledges the receipt of the SIP Invite and informs the client that it is working on the call setup.

SIP server authenticates SIP client

Generate nonce to challenge the user

The SIP client is not authenticated, so the server challenges the client with an nonce value. The client will need to generate a response to the nonce to authenticate itself.


SIP Client
Media Gateway
SIP Server
Status-Line: SIP/2.0 401 Unauthorized
SIP from address: sip:E646657195201@talk4free.com,
CSeq: 1 INVITE,
Authentication Scheme: Digest,
Nonce Value: '30da0aed2_12170',
Realm: 'stratus.com',
Algorithm: MD5
The user is not authorized. The SIP server issues a callenge to authenticate the user. Nonce value is sent to the SIP client. The client is expected to generate a response to the nonce value sent in this message.
SIP ACK sip:9055551212@talk4free.com SIP/2.0
SIP from address: sip:E646657195201@talk4free.com,
CSeq: 1 ACK
SIP client acknowledges the receipt of the nonce challenge.
Compute message digest
The SIP client uses MD5 to compute the response from username, password, method and the received nonce value.
HA1=MD5(username:realm:password)
HA2=MD5(method:digestURI)
response=MD5(HA1:nonce:HA2)
SIP INVITE sip:9055551212@talk4free.com SIP/2.0
SIP from address: sip:E646657195201@talk4free.com,
CSeq: 2 INVITE,
Authentication Scheme: Digest,
Username: 'E646657195201',
Realm: 'stratus.com',
Nonce Value: '30da0aed2_12170',
Authentication URI: 'sip:9055551212@talk4free.com',
Digest Authentication Response: '329e0b8a19bad6f3098c21cd11ec7979',
Algorithm: MD5,
Owner/Creator, Session Id (o): - 2209074887 2209074887 IN IP4 192.168.0.10,
Connection Information (c): IN IP4 192.168.0.10,
Time Description, active time (t): 0 0,
Media Description, name and address (m): audio 49154 RTP/AVP 0 8 101 13,
Connection Information (c): IN IP4 192.168.0.10,
Media Attribute (a): ptime:30,
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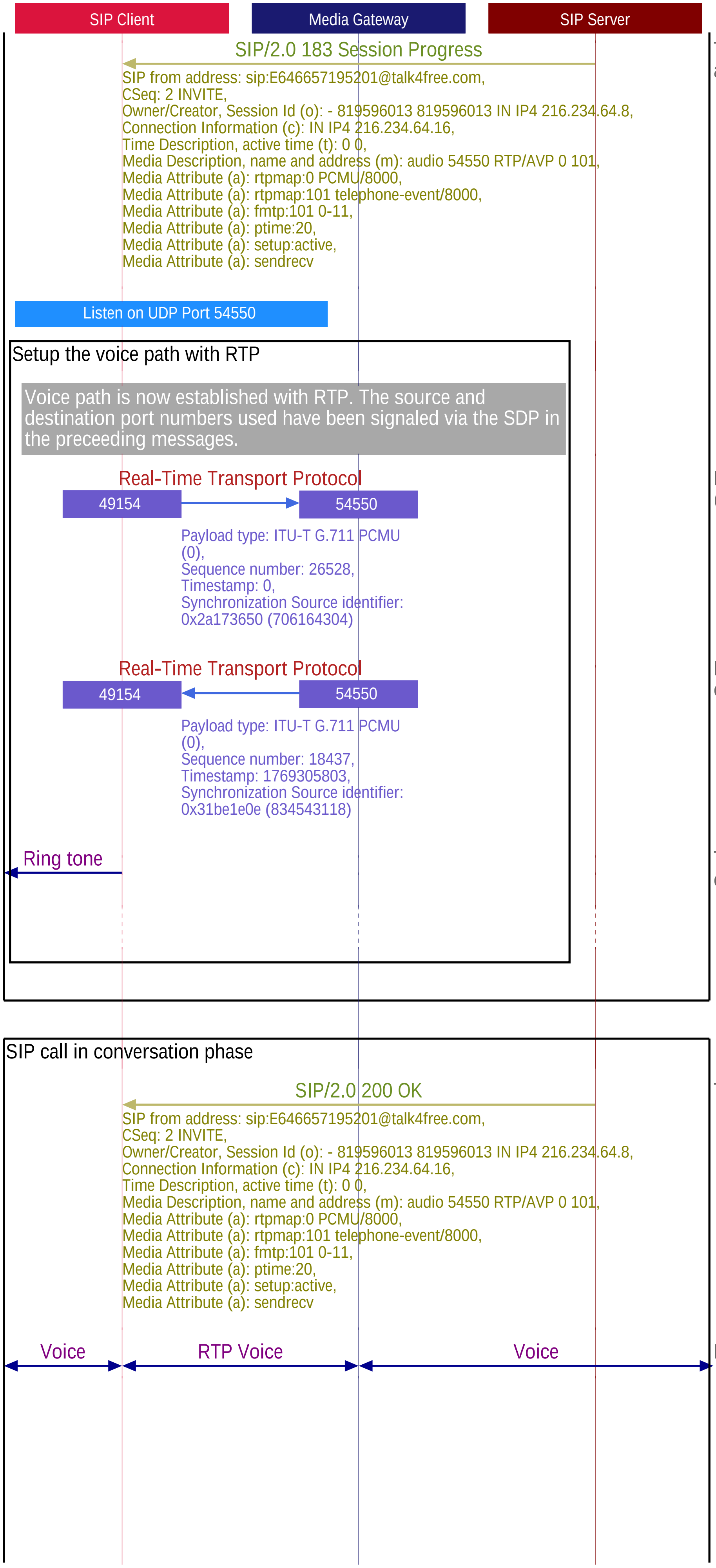
Media Attribute (a): rtpmap:101 telephone-event/8000,
Media Attribute (a): fmtp:101 0-16,
Media Attribute (a): rtpmap:13 CN/8000,
Media Attribute (a): setup:active,
Media Attribute (a): sendrecv
The SIP client resends the INVITE. The "Digest Authentication Response" included in the message is a response to the nonce challenge. The message also resends the SDP information to inform the SIP client about the RTP resources assigned for the voice call.
SIP/2.0 100 Trying
SIP from address: sip:E646657195201@talk4free.com,
CSeq: 2 INVITE
The SIP server signals that it is processing the session.
Authenticate the SIP client's response to the nonce challenge
The SIP server successfully authenticates the user.
Complete SDP negotiation and pass ringing tone
Allocate RTP resources on the Media Gateway
Resources are assigned on the Media Gateway for handling the bi-directional RTP voice flow.
Listen on UDP Port 49154
The client allocates an RTP port and starts listening for RTP packets on that port.
allocate
RTP UDP Port 54550
The server and the media gateway allocate an RTP port and starts listening for RTP packets on that port.



The SIP server responds with the negotiated SDP media attributes.

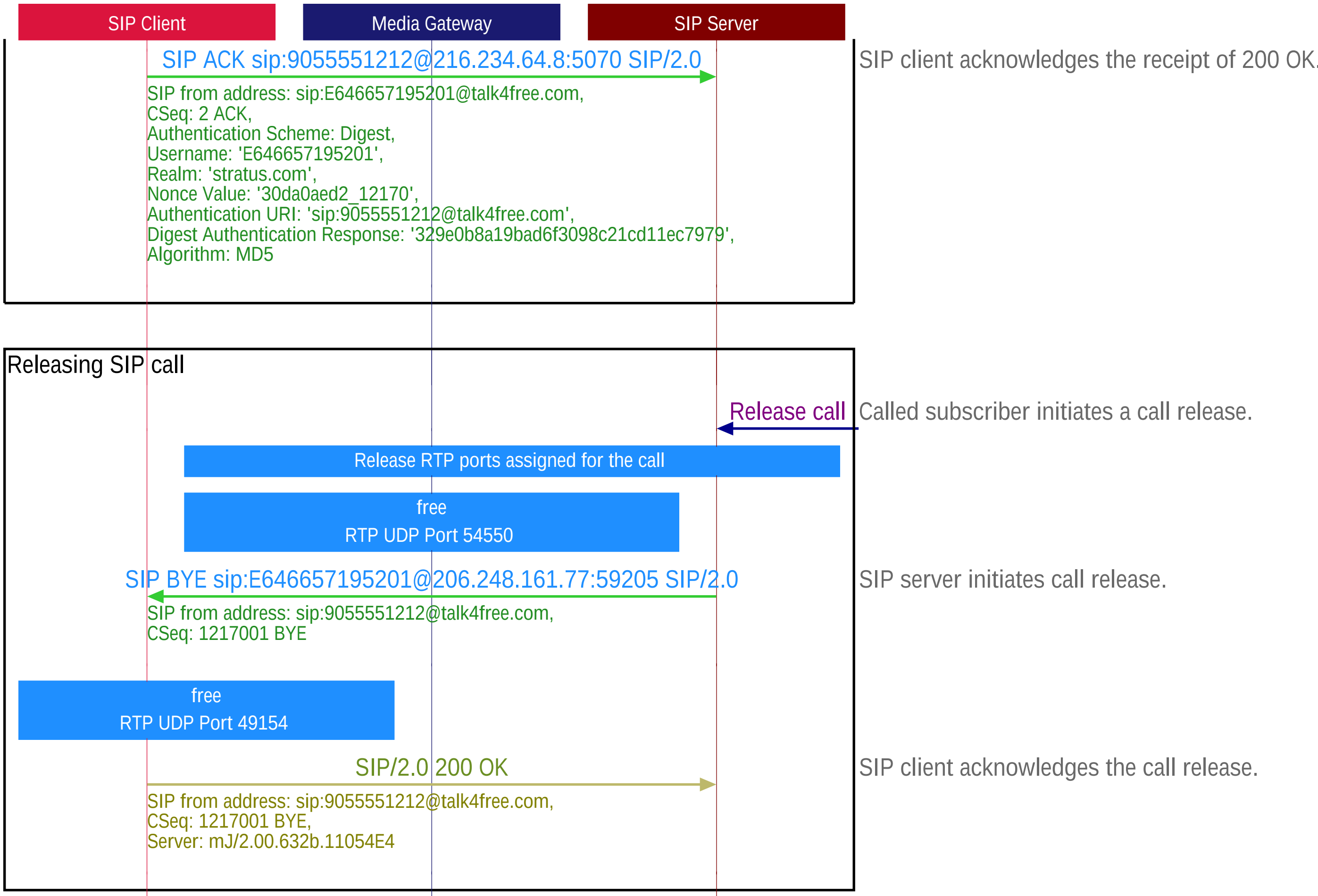
RTP packets from the client (UDP port 49154) to server (UDP port 54550).

RTP packets from the server (UDP port 54550) to the client (UDP port 49154).

The calling subscriber now hears the ringing tone for the called subscriber.

The called subscriber answers the call.

RTP packets are exchanged to carry the voice session.



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