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.

The user initiates a call.

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

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.

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

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.
The user is not authorized. The SIP server issues a challenge 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 client acknowledges the receipt of the nonce challenge.

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)
```

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.

The SIP server signals that it is processing the session.

The SIP server successfully authenticates the user.

Resources are assigned on the Media Gateway for handling the bi-directional RTP voice flow.

The client allocates an RTP port and starts listening for RTP packets on that port.

The server and the media gateway allocate an RTP port and starts listening for RTP packets on that port.
Listen on UDP Port 54550

Setup the voice path with RTP

Voice path is now established with RTP. The source and destination port numbers used have been signaled via the SDP in the preceding messages.

Real-Time Transport Protocol

<table>
<thead>
<tr>
<th>Payload type: ITU-T G.711 (0)</th>
<th>Sequence number: 26528</th>
<th>Timestamp: 0</th>
</tr>
</thead>
<tbody>
<tr>
<td>Synchronization Source identifier: 0x2a173650 (706164304)</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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

Real-Time Transport Protocol

<table>
<thead>
<tr>
<th>Payload type: ITU-T G.711 (0)</th>
<th>Sequence number: 18437</th>
<th>Timestamp: 1769305803</th>
</tr>
</thead>
<tbody>
<tr>
<td>Synchronization Source identifier: 0x31be1e0e (834543118)</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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.

SIP call in conversation phase

The called subscriber answers the call.

RTP packets are exchanged to carry the voice session.
SIP client acknowledges the receipt of 200 OK.

Called subscriber initiates a call release.

SIP server initiates call release.

SIP client acknowledges the call release.


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