This sequence diagram describes the call setup of a call from one IMS subscriber to another IMS subscriber. The calling subscriber is roaming in another IMS supporting network. The called subscriber is in the home IMS network.

The call flow focuses on the IMS routing of SIP dialog. The major steps in the call flow are:

1. IMS Routing of Initial SIP INVITE
2. IMS Routing of First Response to the SIP Invite.
3. PDP Context Activation and Audio/Video Path Setup.

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### Processor Interfaces (Caller and Called are IMS Subscribers)

<table>
<thead>
<tr>
<th>Calling UE</th>
<th>IMS Network</th>
<th>Called UE</th>
</tr>
</thead>
<tbody>
<tr>
<td>Caller User Equipment</td>
<td>Visited IMS 1</td>
<td>Home IMS 1</td>
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<tr>
<th>EventStudio System Designer 4.0</th>
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#### 183 Session Progress

```
Via: <Term P-CSCF>;port <Term S-CSCF> <Term I-CSCF> <Orig S-CSCF> <Orig P-CSCF> <Calling-UE>,
Record-Route: <Term S-CSCF>;port <Orig S-CSCF> <Orig P-CSCF>,
Contact: <Calling UE IP> Port,
SDP: <Codecs supported by Caller and Called>
```

The UE replies indicating that the session is in progress. The contact address is set its own IP address. The Via and the Record-Route headers are copied from the received INVITE.

Just like other nodes, the Orig P-CSCF removes its own entry from the Via header. The P-CSCF also updates the Record-Route header to include the protected port number in its entry. This forces the terminal to send all responses using the protected IPsec SA. The message also includes the media authorization token. This token will have to be passed to the GGSN in the PDP context activation request.

#### PDP Context Activation and Audio/Video Path Setup

<table>
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<tr>
<th>PRACK</th>
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<tr>
<td>200 OK</td>
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The Caller now sends a PRACK to inform the called subscriber about the selected Codec. The message also indicates that currently the resources needed for meeting the quality of service requirements of the session are not available.

The called subscriber acknowledges the PRACK. The message also indicates that quality of service for the session is not met for the called subscriber.

Since the caller PDP context has been activated, notify the called end that the caller can now meet the quality of service in the send and receive direction.

The caller replies back to the called user. Note that the Local QoS is still set to none as the called PDP context activation has not been completed.

#### Ringing

```
180 Ringing
```

Now all the resources for the call are in place. Ring the called subscriber to notify the user about the incoming call.

Inform the caller that the called subscriber is being rung. This serves as an implicit indication to the caller that the QoS at the called side has also been met.

The caller acknowledges the ringing message.

The called subscriber acknowledges the PRACK.

The caller acknowledges the "200 OK" message. The call is now ready to enter conversation mode.

### Conversation on a direct RTP/RTCP connection between the caller and called subscriber SIP phones.