### Caller 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>Visited IMS 1</td>
<td>Home IMS 1</td>
<td>Home IMS 2</td>
</tr>
<tr>
<td>Orig P-CSCF</td>
<td>Orig S-CSCF</td>
<td>Term I-CSCF</td>
</tr>
<tr>
<td>Term S-CSCF</td>
<td>Term P-CSCF</td>
<td>Called</td>
</tr>
</tbody>
</table>

**EventStudio System Designer 4.0**

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### IMS Routing of Initial SIP INVITE

**Initiate Call**
called@hims2.net

Prepare a list of supported voice and video codecs

**INVITE**

```
INVITE called@hims2.net SIP/2.0,
P-Preferred-Identity: <caller@hims1.net>,
Via: <Calling UE IP>:Port,
Route: <P-CSCF address>,
Route: <S-CSCF address>,
Contact: <Calling UE IP>:Port,
SDP: <Caller Supported Codec List>
```

The user initiates a call to called@hims2.net.
The calling includes all supported codecs. This information is included as the first SDP offer in the initial invite.
The SIP phone sends the invite to called@hims2.net. The message contains Route entries for the terminal and the S-CSCF address that was extracted from the Service-Route header in the registration "200 OK" message. Security ports setup for IPSec SA establishment are used. "To" and "From" headers are also included in the message. These headers do not play a role in call processing.
The P-CSCF just acknowledges the INVITE to the UE. The "100 Trying" message indicates that the call setup is in progress.

**100 Trying**

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### IMS Routing of First Response to the SIP Invite

**83 Session Progress**

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

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.

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

**Select one Codec from the common codec list**

**PRACK**

```
PRACK <Selected Codec>,
Local-QOS: none
```

The Caller examines the received common codec list and selects the codec to activate.
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.

Now that the codec to be used has been selected, the PDP context activation is initiated for allocating resources for meeting the Quality of Service (QoS) requirements for the codec.
The called subscriber acknowledges the PRACK. The message also indicates that quality of service for the session is not met for the called subscriber.

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**200 OK**

```
SDP: <Selected Codec>,
Local-QOS: none
```

The called subscriber acknowledges the PRACK. The message also indicates that quality of service for the session is not met for the called subscriber.
The caller PDP context activation has been completed.

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.

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.

Notify the caller that the call has been answered.

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