Called Interfaces (Caller and Called are IMS Subscribers)

Called User Equipment
- Home IMS 1
- Visited IMS 1
- Home IMS 2
- Term P-CSCF
- Term I-CSCF
- Term S-CSCF
- Term P-CSCF
- Called

Caller User Equipment
- Orig P-CSCF
- Orig S-CSCF

Calling UE
- Orig P-CSCF
- Orig S-CSCF
- Visited IMS 1
- Home IMS 1
- Called UE
- Home IMS 2
- Term I-CSCF
- Term S-CSCF
- Term P-CSCF

IMS Routing of Initial SIP INVITE

INVITE
INVITE CALLED-IP SIP/2.0,
P-Asserted-Identity: <caller@hims1.net>,
tel: +13015556666,
Via: <Term P-CSCF>; port <Term S-CSCF> <Term I-CSCF> <Orig S-CSCF> <Orig P-CSCF>
Calling-UE,
Route: <Term P-CSCF>; port,
Record-Route: <Term S-CSCF> <Orig S-CSCF> <Orig P-CSCF>,
Contact: <Called UE IP> :Port,
SDP: <Caller Supported Codec List>,
P-Media-Authorization

The P-CSCF updates the Via and Route-Record headers and forwards the request to the Called UE. Note that the secure port is included in the Via address specification. The message also includes the media authorization token. This token will have to be passed to the GGSN in the PDP context activation request.

Prepare a list of Codecs common between the Caller and the Called subscriber

The Caller examines the SDP list of available codec. It prunes the list by excluding codecs that are not supported by the called subscriber. This list will be included in the 183 message sent to the caller.

IMS Routing of First Response to the SIP Invite

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: <Called 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.

PDP Context Activation and Audio/Video Path Setup

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

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.

200 OK
SDP: <Selected Codec>, <Local-QOS: none>

The final codec at the called side is decided. So initiate the PDP context activation to allocate resources for meeting the QoS of the terminating leg of the call.

begin
Called PDP Context Activation

UPDATE
SDP: <Local-QOS: sendrecv>

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.

200 OK
SDP: <Local-QOS: none>

The caller replies back to the called user. Note that the Local QoS is still set to none as the
Called Interfaces (Caller and Called are IMS Subscribers)

<table>
<thead>
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<th>Calling UE Equipment</th>
<th>IMS Network</th>
<th>Called UE Equipment</th>
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<tr>
<td>Visited IMS 1</td>
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<td>Term P-CSCF</td>
<td>Called</td>
</tr>
</tbody>
</table>

Called PDP context activation has not been completed.

The called PDP context activation has been completed. At this point, the caller and the called PDP contexts are both active. The QoS for the call can now be met.

Ringing

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

180 Ringing

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.

PRACK

The caller acknowledges the ringing message.

200 OK

The called subscriber acknowledges the PRACK.

Answer

The called subscriber answers the call.

200 OK

Notify the caller that the call has been answered.

AOK

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.