Processor Interfaces (Caller and Called are IMS Subscribers)					
Calling UE	IMS Network			Called UE	EventStudio System Designer 4.0
Caller User Equipment	Visited IMS 1	Home IMS 1	Home IMS 2	Called User Equipment	15-Dec-07 07:55 (Page 1)

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:

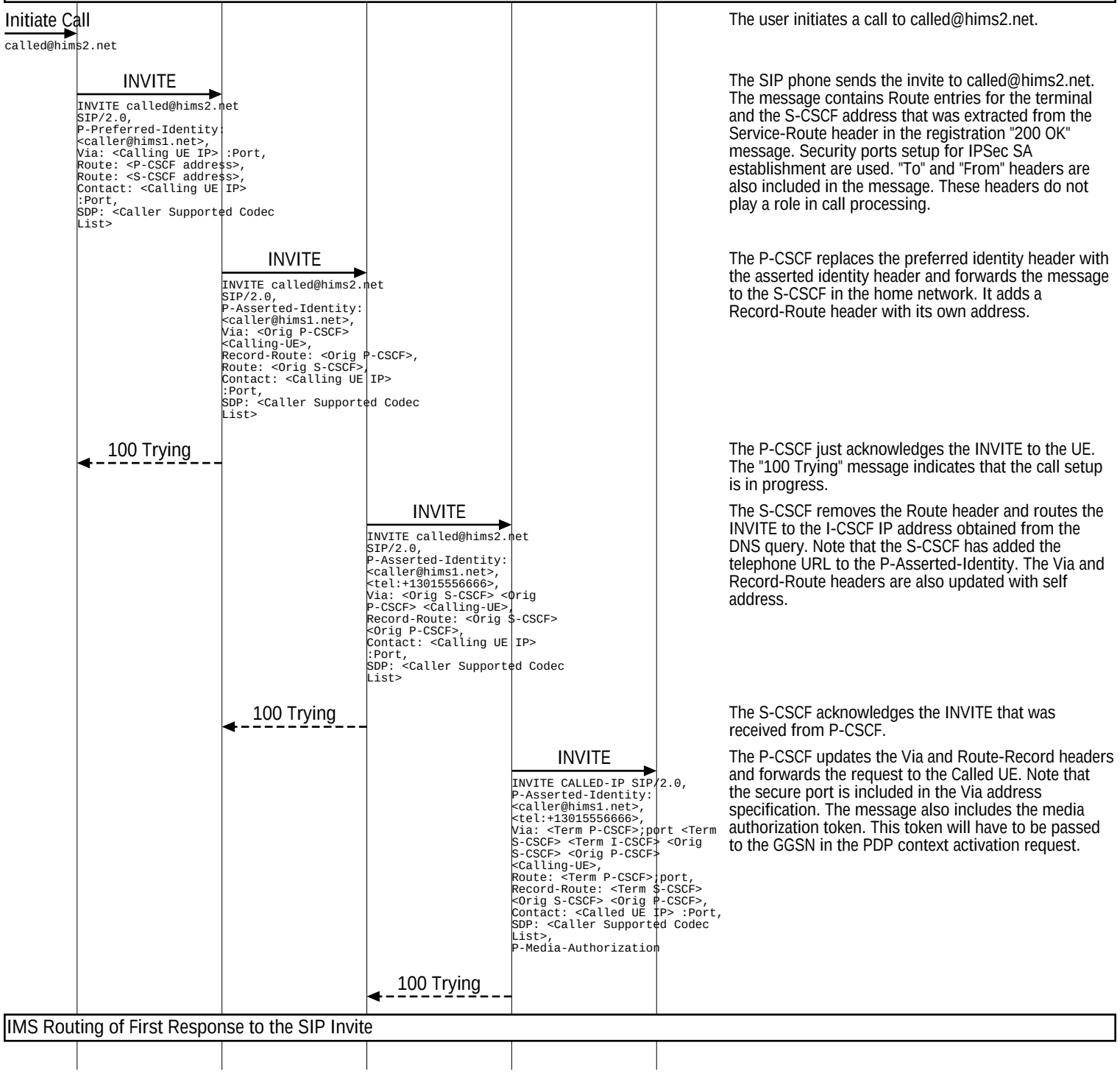
(1) IMS Routing of Initial SIP INVITE.

(2) IMS Routing of First Response to the SIP Invite.

(3) PDP Context Activation and Audio/Video Path Setup.

This sequence diagram was generated with EventStudio System Designer 4.0 (http://www.EventHelix.com/EventStudio). Copyright © 2007 EventHelix.com Inc. All Rights Reserved.

IMS Routing of Initial SIP INVITE



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: <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.

183 Session Progress

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

183 Session Progress

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

183 Session Progress

```
Via: <Calling-UE>,
Record-Route: <Term
S-CSCF>;port <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.

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 requiements of the session are not available.

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.

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 PDP context activation has not been completed.

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 that the call has been answered.

ACK

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.