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

| Calling UE | IMS Network | Called UE | EventStudio System Designer 4.0 |
|---|---|---|---|

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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### IMS Routing of Initial SIP INVITE

**Initiate Call**

```
called@hims2.net
```

The user initiates a call to called@hims2.net.

**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 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.

**100 Trying**

The P-CSCF just acknowledges the INVITE to the UE. The "100 Trying" message indicates that the call setup is in progress.

**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.

### 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: <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>
```

**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>
```

**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>
```

**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>
```

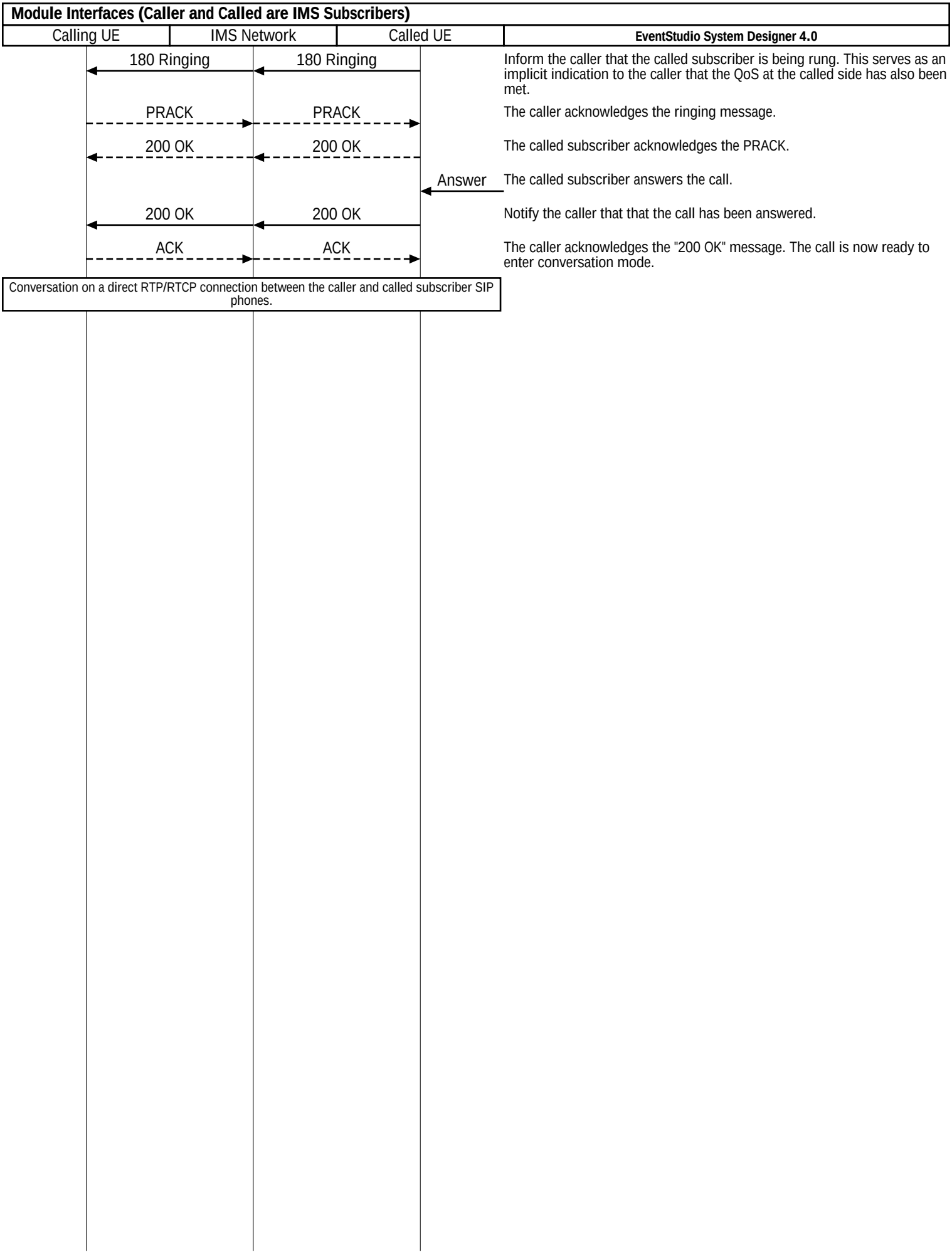
**200 OK**

```
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.


Module Interfaces (Caller and Called are IMS Subscribers)
Calling UE
IMS Network
Called UE
EventStudio System Designer 4.0
180 Ringing
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
PRACK
The caller acknowledges the ringing message.
200 OK
200 OK
The called subscriber acknowledges the PRACK.
Answer
The called subscriber answers the call.
200 OK
200 OK
Notify the caller that that the call has been answered.
ACK
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.