## 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. 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 The user initiates a call to called@hims2.net. Initiate Call called@hims2.net **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 INVITE called@hims2.net SIP/2.0, P-Preferred-Identity: <caller@hims1.net>, Via: <Calling UE IP> :Port, extracted from the Service-Route header in the registration "200 OK" message. Security ports setup for IPSec SA establishment are used. "To" Via: Clairing Us 19 : FOIT, Route: <P-CSCF address>, Route: <S-CSCF address>, Contact: <Calling UE 1P> :Port, SDP: <Caller Supported Codec List> 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. The P-CSCF updates the Via and Route-Record headers and forwards the INVITE request to the Called UE. Note that the secure port is included in the Via INVITE CALLED-IP SIP/2.0, P-Asserted-Identity: <caller@hims1.net>, ctel:+13015556666>, Via: <Term P-CSCF>;port <Term S-CSCF> <calling-UE>, CCILING-UE>, CCILING-U address specification. The message also includes the media authorization token. This token will have to be passed to the GGSN in the PDP context <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 P-Media-Authorization</pre> IMS Routing of First Response to the SIP Invite The UE replies indicating that the session is in progress. The contact 183 Session Progress address is set its own IP address. The Via and the Record-Route headers via: <Term P-CSCF>:port <Term S-CSCF> dddress is set its own IP address. In <Term I-CSCF> <Orig S-CSCF> dorig P-CSCF> are copied from the received INVITE. <Calling-UE>, Record-Route: <Term S-CSCF>:port <Orig S-CSCF> <Orig P-CSCF>, Contact: <Calling UE IP> :Pqrt, SDP: <Codecs supported by Caller and Called> 183 Session Progress 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 Via: <Calling-UE>, 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 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 PRACK** The Caller now sends a PRACK to inform the called subscriber about the selected Codec. The message also indicates that currently the resources SDP: <Selected Codec>, <Selected Codec>, needed for meeting the quality of service requiements of the session are <Local-OOS: none> <Local-OOS: none> not available. The called subscriber acknowledges the PRACK. The message also 200 OK 200 OK indicates that quality of service for the session is not met for the called SDP: <Selected Codec>. SDP: <Selected Codec>. subscriber. <Local-QOS: none> <Local-QOS: Since the caller PDP context has been activated, notify the called end that **UPDATE** UPDATE the caller can now meet the quality of service in the send and receive SDP: <Local-QOS: sendrecv> SDP: <Local-QOS: sendrecv> direction. 200 OK 200 OK 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. SDP: <Local-OOS: none> SDP: <Local-QOS: none> Now all the resources for the call are in place. Ring the called subscriber to notify the user about the incoming call.

