SDP Use in an IMS-to-IMS Call (SDP Codec Selection and QoS Signaling)

Calling UE	Core Network	Called UE	EventStudio System Designer 4.0
Caller User Equipment	GGSN	Called User Equipment	07-Dec-07 22:30 (Page 1)
Caller	GGSN	Called	

Session Description Protocol (SDP) specifies a format for exchanging streaming related parameters between SIP subscribers. The following sequence diagram focuses on the SDP interactions between two IMS subscribers. The flow covers two phases of the SDP negotiation:

(1) Codec selection between the calling and call IMS subscribers.

(2) SDP signaling involved in exchanging quality of service information.

Codec Selection

Initiate Call (→ Caller)

```
called@hims2.net
```

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

[Caller] Prepare a list of supported voice and video codecs

The calling includes all supported codecs. This information is included as the first SDP offer in the initial invite.

INVITE (Caller → Called)

```
v=0,
o=- <session-id> <version> IN IPV6 <caller-ipaddr>,
c=IN IPV6 <caller-ipaddr>,
s=-,
t=<start-time> 0,
m=audio <caller-port> RTP/AVP 96 97 98,
a=rtpmap:96 H263,
a=rtpmap:97 AMR,
a=rtpmap:98 telephone-event,
a=curr:qos local none,
a=curr:qos remote none,
a=des:qos mandatory local sendrecv,
a=des:qos none remote sendrecv
```

The user terminal sends the initial INVITE with three voice codecs. The terminal also inidicates that it will need to allocate resources to meet the quality of service (QoS) requirements for codecs. The "m=" line specifies the <caller-port>, the transport type (RTP/AVP) and the supported codecs ids(96, 97 and 98). The "a=rtpmap" lines map the codec ids 96, 97 and 98 to H263, AMR and telephone-event (DTMF) respectively. The "a=curr" lines specify the the QoS for the caller (local) and the called (remote) ends are not currently met. The second last line indicates that the caller (local) needs to allocate resources in send and receive directions to meet the quality of service requirements for the codec. The last line indicates that the caller end has no specific requirements for the called (remote) user.

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

The Called subscriber 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.

183 Session Progress (Called → Caller)

```
v=0,
o=- <session-id> <version> IN IPV6 <called-ipaddr>,
c=IN IPV6 <called-ipaddr>,
s=-,
t=<start-time> 0,
m=audio <called-port> RTP/AVP 96 97,
a=rtpmap:96 H263,
a=rtpmap:97 AMR,
a=curr:qos local none,
a=curr:qos remote none,
a=des:qos mandatory local sendrecv,
a=des:qos mandatory remote sendrecv,
a=conf:qos remote sendrecv
```

The UE replies back with 96 and 97 in the "m=" line. 98 is removed as telephone-event (DTMF handling) is not supported. The called UE also uses the "a=curr" lines to specify that QoS for the session is currently not met. Note that the "a=des" lines now signify that called (local) also needs to allocate resources for meeting the QoS. This message also instructs the caller (remote) to inform the called subscriber when the caller acquires the resources for meeting the QoS. This QoS confirmation is being requested in the last line ("a=conf").

PDP Contect Activation QoS Signaling

[Caller] Select one Codec from the common codec list

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

PRACK (Caller → Called)

```
v=0,
o=- <session-id> <version> IN IPV6 <caller-ipaddr>,
c=IN IPV6 <caller-ipaddr>,
s=-,
t=<start-time> 0,
m=audio <caller-port> RTP/AVP 97,
a=rtpmap:97 AMR,
a=curr:qos local none,
a=curr:qos remote none,
a=des:qos mandatory local sendrecv,
a=des:qos mandatory remote sendrecv
```

The Caller now sends a PRACK to inform the called subscriber about the selected Codec. The caller has selected the AMR codec. This is signaled by the "m=" and "a=" lines.

Activate PDP Context (Caller → GGSN)

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.

200 OK (Called → Caller)

```
v=0,
o=- <session-id> <version> IN IPV6 <called-ipaddr>,
c=IN IPV6 <called-ipaddr>,
s=-,
t=<start-time> 0,
m=audio <called-port> RTP/AVP 97,
a=rtpmap:97 AMR,
a=curr:qos local none,
a=curr:qos remote none,
a=des:qos mandatory local sendrecv,
a=des:qos mandatory remote sendrecv
```

The called subscriber acknowledges the PRACK. The message also indicates that quality of service for the session is not met for the called subscriber.

Activate PDP Context (Called → GGSN)

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.

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Caller	GGSN	Called	

Activate PDP Context Accept (GGSN → Called)	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.
Activate PDP Context Accept (GGSN → Caller)	The caller PDP context activation has been completed.
UPDATE (Caller → Called)	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 "a=curr:qos local sendrecv" signals that the caller (local) PDP context has been established. Note that the UPDATE is being sent in response to the QOS confirmation request received in "183 Session Progress" message from the caller.

```
v=0,
o=- <session-id> <version> IN IPV6 <caller-ipaddr>,
c=IN IPV6 <caller-ipaddr>,
s=-,
t=<start-time> 0,
m=audio <caller-port> RTP/AVP 97,
a=rtpmap:97 AMR,
a=curr:qos local sendrecv,
a=curr:qos remote none,
a=des:qos mandatory local sendrecv,
a=des:qos mandatory remote sendrecv
```

200 OK (Called → Caller)	The caller replies back to the called user. Note that the "a=cur" line for the called (local) has been updated to indicate that called end QoS is also met.

```
v=0,
o=- <session-id> <version> IN IPV6 <called-ipaddr>,
c=IN IPV6 <called-ipaddr>,
s=-,
t=<start-time> 0,
m=audio <called-port> RTP/AVP 97,
a=rtpmap:97 AMR,
a=curr:qos local sendrecv,
a=curr:qos remote sendrecv,
a=des:qos mandatory local sendrecv,
a=des:qos mandatory remote sendrecv
```

Ringing (Called →)	Now all the resources for the call are in place. Ring the called subscriber to notify the user about the incoming call.
180 Ringing (Called → Caller)	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 (Caller → Called)	The caller acknowledges the ringing message.
200 OK (Called → Caller)	The called subscriber acknowledges the PRACK.
Answer (→ Called)	The called subscriber answers the call.
200 OK (Called → Caller)	Notify the caller that that the call has been answered.
ACK (Caller → Called)	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.