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IMS/MMD Call Flow Examples

Revision History

Revision	Changes	Date
v1.0	Initial Publication	December, 2007

1   IMS/MMD Call Flow Examples

2   **CONTENTS**

1 LIST OF FIGURES

1 Introduction

The document provides examples of signaling flows for the IP multimedia call control based on SIP and SDP. The signaling flows specified in this document are only for informational purposes. If there is ambiguity between this specification and [1], the text specified in [1] shall be followed.

The call flows describe the behavior of the mobile stations under various conditions for setting up real time services like VoIP and PSVT.

In this document, several key words are used to signify the requirements. The key words "shall", "shall not", "should", "should not" and "may" are to be interpreted as described in [6] and the TIA Engineering Style Manual.

2   Glossary and Definitions

2.1   Acronyms

AAA	Authentication, Authorization, and Accounting
AN	Access Network
AT	Access Terminal
EMPA	Enhanced Multi-Flow Packet Application
HRPD	High Rate Packet Data
HSS	Home Subscriber Server
IMS	IP Multimedia Subsystem
IP	Internet Protocol
IP-CAN	IP-Connectivity Access Network
MMD	Multi Media Domain
MPA	Multi-Flow Packet Application
PDSN	Packet Data Serving Node
PFO	PPP Free Operation
PPP	Point to Point Protocol
PSVT	Packet Switched Video Telephony
QoS	Quality of Service
RAN	Radio Access Network
RSVP	Resource ReSerVation Protocol
SBBC	Service Based Bearer Control
SDP	Session Description Protocol
SIP	Session Initiation Protocol
TFT	Traffic Flow Template
UDP	User Datagram Protocol
UE	User Equipment
VoIP	Voice Over IP

2.2   Definitions

Caller	The person placing a call
Called party	The recipient or destination of a call
Alert	The audible notification given to the Called party of an incoming call.
Ring Back	The audible notification given to a Caller to indicate that the Called party has been located and is being alerted

3 References

The following documents contain provisions, which, through reference in this text, constitute provisions of this document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.

- For a specific reference, subsequent revisions do not apply.

- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP2 document, a non-specific reference implicitly refers to the latest version of that document in the same Release as the present document.

3.1 Normative References

[1] 3GPP2 X.S0013-004-A v1.0: "IP Multimedia Call Control Protocol Based on SIP and SDP Stage 3".

[2] 3GPP2 X.S0013-002-A v1.0: "IP Multimedia (IM) Subsystem – Stage 2".

[3] IETF RFC 2429, " RTP Payload Format for the 1998 Version of ITU-T Rec. H.263 Video (H.263+)".

[4] IETF RFC 3558, "RTP Payload Format for Enhanced Variable Rate Codecs (EVRC) and Selectable Mode Vocoders (SMV)".

[5] IETF RFC 3262, "Reliability of provisional Responses in the Session Initiation Protocol (SIP)"

[6] IETF RFC 2119, "Key words for use in RFCs to Indicate Requirement Levels", March 1997.

[7] 3GPP2 X.S0013-005-A v1.0: "All-IP Core Network Multimedia Domain IP Multimedia Subsystem Cx Interface Signaling flows and Message Contents" – Annex A.

3.2 Informative References

[8] 3GPP2 X.S0013-012-0 v1.0: "All-IP Core Network Multimedia Domain; Service Based Bearer Control – Stage 2".

4   Methodology

4.1   Functional Entities covered by call flows

The flows show the signaling exchanges between the following functional entities:

- ▪   User Equipment (UE)
- ▪   Proxy-CSCF (P-CSCF)
- ▪   Interrogating-CSCF (I-CSCF)
- ▪   Serving-CSCF (S-CSCF)

The call flows mainly show the interactions between the UE-1 and UE-2 for call origination and termination. The procedures and the message exchanged between these elements are as described in [1].

4.2   Identities

4.2.1   User and Network Identities

The public identity of UE-1 is sip:UE-1@domain1.net. The public identity of UE-2 is sip:UE-2@domain2.net

The following are network entities associated with UE-1:

- ▪   Home Domain of UE-1: domain1.net
- ▪   P-CSCF serving UE-1:   p-cscf1.domain1.net
- ▪   I-CSCF serving UE-1:   i-cscf1.domain1.net
- ▪   S-CSCF serving UE-1:   s-cscf1.domain1.net

The following are the network entities associated with UE-2:

- ▪   Home Domain of UE-2: domain2.net
- ▪   P-CSCF serving UE-2:   p-cscf2.domain2.net
- ▪   I-CSCF serving UE-2:   i-cscf2.domain2.net
- ▪   S-CSCF serving UE-2:   s-cscf2.domain2.net

In the example session establishment flows, both UE-1 and UE-2 are assumed to be in the home network. So, the P-CSCF1 and P-CSCF2 are in UE-1's and UE-2's respective home networks. However, according to [2], the P-CSCF can be either in the home or visited network. The session establishment call flows described in this document are not affected whether the P-CSCF is in the visited or home network.

For brevity, I-CSCF and S-CSCF are shown together in the session establishment call flows.

4.3 Notation for call flows

Offer/Answer exchange process is also shown in the call flows. For brevity, the following notation is used to represent offer and answer:

- "O" and "A" in the call flows represent "Offer" and "Answer".
- "O_n" represents the nth SDP offer. For example, if there are two offer/answer exchanges during the call setup process, then the first offer will be noted as "O_1" and the second as "O_2".
- "A_n" represents the nth SDP Answer. Answer "A_n" corresponds to Offer "O_n".

5 Registration Procedures

The registration procedures for UE-1 and UE-2 are as specified in [1] and [7]. UE-1 registers with S-CSCF1 through P-CSCF1 and UE-2 registers with S-CSCF2 through P-CSCF2.

6 Signaling flows for session establishment

In all of the call flows provided in this document, registration procedures are assumed to have already been completed.

6.1 General assumptions

All the call flows shown in this document assume the following:

- The originating UE and the terminating UE both support precondition and reliable provisional responses 100 (rel).
- The originating UE will only include "Supported: precondition" in the SIP INVITE it sends out to its peer.
  - o If the originating UE wishes to both send and receive media with its peer, and the resources are reserved for the stream, the originating UE marks the stream as sendrecv using the "a=sendrecv" attribute. (Note: for sendonly streams, the originating UE marks the stream as "a=sendonly" and for recvonly streams, the originating UE marks the stream as "a=recvonly").
  - o If the originating UE wishes to communicate with its peer, and the resources are not reserved for the stream, the originating UE marks the particular stream as inactive using the "a=inactive" attribute.
- If the "Supported" header in the incoming INVITE includes "precondition" and the terminating UE decides to use precondition, the terminating UE includes "Require: precondition" in all reliable responses that carry SDP (i.e., offer or answer).
- The terminating UE sends 180 (Ringing) response reliably. Note that sending the 180 (Ringing) response reliably does not increase the call setup time experienced by the caller.
- If the 180 (Ringing) response contains an answer (or offer), the called party is alerted only after receiving a PRACK request for the 180 (Ringing) response. This enhances the caller/called party user experience. For example, if the called party picks up the phone before the 180 (Ringing) response reaches the caller, the caller will only be able to hear the called party but will not be able to respond,
- Both UEs have the required resources ready before the terminating UE can alert the called party of an incoming call.
- For brevity, 100 (Trying) messages are not shown in the figures.
- The SBBC interactions [8] are not shown in the call flows.

6.2 Scenarios

The following scenarios are considered for the session establishment process:

- Scenario 1: Originating UE has resources ready before sending INVITE, terminating UE has resources ready before sending the first provisional response;
- Scenario 2: Originating UE has resources ready before sending INVITE, terminating UE does not have resources ready before sending the first provisional response;
- Scenario 3: Originating UE does not have resources ready before sending INVITE, terminating UE has resources ready before sending the first provisional response;
- Scenario 4: Originating UE does not have resources ready before sending INVITE, terminating UE does not have resources ready before sending the first provisional response.

The call flows for the above four scenarios are depicted in the subsequent subsections.

6.2.1 Scenario 1

This section covers the scenario where the originating UE's resources are ready before sending INVITE, and the

terminating UE's resources are ready before sending the first provisional response.

6.2.1.1 Assumptions

6.2.1.2 Call Flow

The following section applies to the case where UE-1 has its resource ready before sending the INVITE request to UE-2. The terminating UE, UE-2, also has its resource ready before answering the INVITE request with the first provisional response.

Note: This flow assumes UE-2 has sufficient resources available prior to receiving the INVITE.
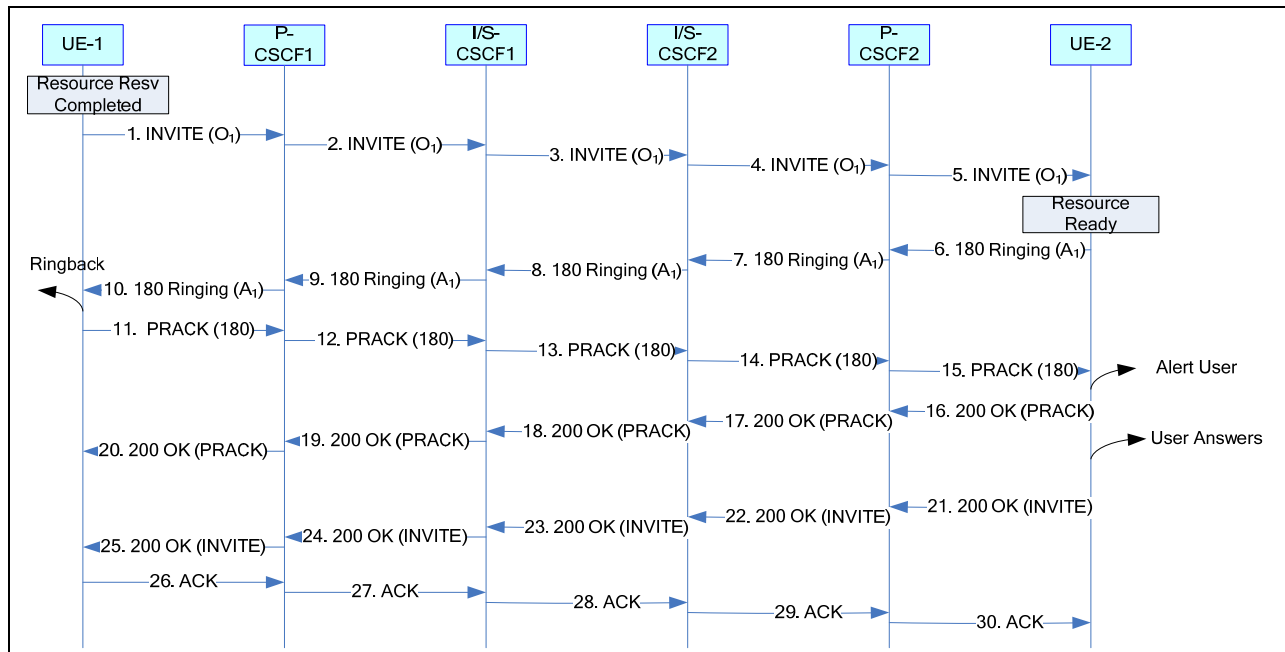
Figure 1 Originating UE resource ready, terminating UE resource ready

1. INVITE (UE-1 to P-CSCF1)

UE-1 determines the set of codecs or media streams that it wishes to support for the session. It builds an SDP offer containing characteristics of each codec, and assigns local port numbers for each possible media flow. Multiple media flows may be offered, and for each media flow (m= line in SDP), there may be multiple codec choices offered.

For this example, it is assumed that UE-1 is willing to establish a multimedia session comprising a video stream and an audio stream. The video stream supports one H.263 codec as specified in [3]. The audio stream supports both EVRC and SMV codecs as specified in [4].

In addition, UE-1 indicates that precondition is supported for this session. In the SDP offer, it indicates that resource is already available at the local end point.

1   **Table 6.2.1.2-1 INVITE (UE-1 to P-CSCF1)**

```
INVITE sip:UE-2@domain2.net SIP/2.0
From: <sip:UE-1@domain1.net>;tag=169f498-0-13c4-78e-2044f20e-78e
To: <sip:UE-2@domain2.net>
Call-ID: 16a1b80-0-13c4-78e-6a540802-78e@10.20.1.100
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.20.1.100:1357;comp=sigcomp;branch=z9hG4bK-78f-1d86b6-24d29348
Max-Forwards: 70
Route: <sip:pcscf1.domain1.net:7531;lr;comp=sigcomp>, <sip:scscf1.domain1.net;lr>
P-Preferred-Identity: "User-1" <sip:UE-1@domain1.net>
P-Access-Network-Info: 3GPP2-1X-HRPD; ci-3gpp2=12341234123412341234123412341234411
Privacy: none
Require: sec-agree
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96;spi-c=98765432; spi-s=76543210;
port-c=13579; port-s=23456
Contact: <sip:UE-1@10.20.1.100:1357;comp=sigcomp>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Supported: 100rel, precondition
Content-Type: application/sdp
Content-Length: xxx

v=0
o=- 3323527065117000 3323527065117000 IN IP4 10.20.1.100
s=-
c=IN IP4 10.20.1.100
t=0 0
m=audio 49500 RTP/AVP 97 99
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
a=rtpmap:97 EVRC/8000
a=ptime:20
a=rtpmap:99 SMV/8000
m=video 49600 RTP/AVP 34
b=AS:75
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
a=rtpmap:34 H263/90000
a=cif:1
```

2
3   2.   INVITE (P-CSCF1 to I/S-CSCF1)
4   The P-CSCF1 adds itself to the Record-Route header and Via header. As the request is forwarded to an interface that
5   is not compressed, the P-CSCF1 SIP URI does not contain the "comp=sigcomp" parameter. The P-CSCF1 removes
6   the Security-Verify header and associated "sec-agree" option-tags prior to forwarding the request. As the Require
7   and Proxy-Require headers are empty, P-CSCF1 removes those headers completely.
8
9   The INVITE request is forwarded to the I/S-CSCF1.
10
11 3.   INVITE (I/S-CSCF1 to I/S-CSCF2)
12 S-CSCF1 performs an analysis of the destination address, and determines the network operator to whom the
13 destination subscriber belongs. Since the originating operator does not desire to keep their internal configuration
14 hidden, S-CSCF1 forwards the INVITE request directly to I-CSCF2 in the destination network.
15
16 The I-CSCF2 sends a query to the HSS to find out the S-CSCF2 of the called user. The HSS responds with the

address of the current S-CSCF2 for the terminating subscriber. I-CSCF2 forwards the INVITE request to the S-CSCF2 that will handle the session termination. S-CSCF2 validates the service profile of this subscriber and evaluates the initial filter criteria.

4-5.   INVITE (I/S-CSCF2 to UE-2)
S-CSCF2 forwards the INVITE request to UE-2 via P-CSCF2.

6.   180 Ringing (UE-2 to P-CSCF2)
UE-2 has accepted both video and audio streams, and EVRC is the chosen codec for the audio stream. Since resources are already available for UE-2, it sends a 180 (Ringing) response reliably with the SDP answer indicating that resources are reserved at both endpoints.

7-10.   180 Ringing (P-CSCF2 to UE-1)
P-CSCF2 forwards the 180 (Ringing) response to UE-1   via I/S-CSCF2, I/S-CSCF1, and P-CSCF1. Table 6.2.1.2-2 shows the 180 (Ringing) response in detail.

Table 6.2.1.2-2 180 Ringing (P-CSCF1 to UE-1)

```
SIP/2.0 180 Ringing
From: <sip:UE-1@domain1.net>;tag=169f498-0-13c4-78e-2044f20e-78e
To: <sip:UE-2@domain2.net>;tag=169f7f8-0-13c4-9f6-33835972-9f6
Call-ID: 16a1b80-0-13c4-78e-6a540802-78e@10.20.1.100
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.20.1.100:1357;comp=sigcomp;branch=z9hG4bK-78f-1d86b6-24d29348
Record-Route:<sip:pcscf2.domain2.net;lr>,<sip:scscf2.domain2.net;lr>,<sip:scscf1.domain1.net;lr>,
<sip:pcscf1.domain1.net:7531;lr;comp=sigcomp>
Contact: <sip:UE-2@10.30.1.24:8805;comp=sigcomp>
P-Asserted-Identity: "User 2" <sip:UE-2@domain2.net>, <tel:+1-858-335-7341>
Privacy: none
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Require: 100rel, precondition
RSeq: 1000
Content-Type: application/sdp
Content-Length: xxx


v=0
o=- 33235270718000 33235270718000 IN IP4 10.30.1.24
s=-
c=IN IP4 10.30.1.24
t=0 0
m=audio 49700 RTP/AVP 97
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
a=rtpmap:97 EVRC/8000
a=ptime:20
m=video 49702 RTP/AVP 34
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
a=rtpmap:34 H263/90000
a=cif:1
```

11. PRACK (UE-1 to P-CSCF1)
UE-1 acknowledges the 180 (Ringing) response from UE-2 with a PRACK request as specified in [5]. (Table 6.2.1.2-3)

1   If UE-1 determines to make any change in media flows, it includes a new SDP offer in the PRACK request sent to
2   UE-2. Otherwise, no SDP offer is included in the PRACK request.
3
4   **Table 6.2.1.2-3 PRACK (UE-1 to P-CSCF1)**

```
PRACK sip:UE-2@10.30.1.24:8805;comp=sigcomp SIP/2.0
From: <sip:UE-1@domain1.net>;tag=169f498-0-13c4-78e-2044f20e-78e
To: <sip:UE-2@domain2.net>; tag=169f7f8-0-13c4-9f6-33835972-9f6
Call-ID: 16a1b80-0-13c4-78e-6a540802-78e@10.20.1.100
P-Access-Network-Info: 3GPP2-1X-HRPD;ci-3gpp2=12341234123412341234123412341234123411
CSeq: 2 PRACK
Via: SIP/2.0/UDP 10.20.1.100:1357;comp=sigcomp;branch=z9hG4bK-78f-1d86b6-24d29348
Max-Forwards: 70
Route:
<sip:pcscf1.domain1.net:7531;lr;comp=sigcomp>,<sip:scscf1.domain1.net;lr>,<sip:scscf2.d
omain2.net;lr>,
<sip:pcscf2.domain2.net;lr>
RAck: 1000 1 INVITE
Content-Length: 0
```

5
6   12. PRACK (P-CSCF1 to I/S-CSCF1)
7   The P-CSCF forwards the PRACK request to S-CSCF. As the Proxy-Require header is empty, the P-CSCF removes
8   this header completely.
9
10   13-14.   PRACK (I/S-CSCF1 to P-CSCF2)
11   The I/S-CSCF1 forwards the PRACK request to P-CSCF2 via I/S-CSCF2.
12
13   15. PRACK (P-CSCF2 to UE-2)
14   UE-2 starts alerting the user after it receives the PRACK request for the 180 (Ringing) response. UE-2 may also
15   alert the user before receiving the PRACK request, but there may be media clipping if the user answers the call
16   before the 180 (Ringing) response reaches UE-1.
17
18   16. 200 OK (UE-2 to P-CSCF2)
19   The 200 OK response is generated by UE-2 to acknowledge the reception of the PRACK request.
20
21   17-20.   200 OK (P-CSCF2 to UE-1)
22   The P-CSCF2 forwards the 200 OK response to UE-1 via I/S-CSCF2, I/S-CSCF1, and P-CSCF1. Table 6.2.1.2-4
23   shows the 200 OK message going from P-CSCF1 to UE-1.
24
25   **Table 6.2.1.2-4 200 OK (P-CSCF1 to UE-1)**

```
SIP/2.0 200 OK
From: <sip:UE-1@domain1.net>;tag=169f498-0-13c4-78e-2044f20e-78e
To: <sip:UE-2@domain2.net>;tag=169f7f8-0-13c4-9f6-33835972-9f6
Call-ID: 16a1b80-0-13c4-78e-6a540802-78e@10.20.1.100
CSeq: 2 PRACK
Via: SIP/2.0/UDP 10.20.1.100:1357;comp=sigcomp;branch=z9hG4bK-78f-1d86b6-24d29348
Content-Length: 0
```

26
27   21. 200 OK (UE-2 to P-CSCF2)
28   When the user at UE-2 answers the call, UE-2 generates a 200 OK response towards UE-1 to answer the INVITE
29   request.
30
31   22-25.   200 OK (P-CSCF2 to UE-1)
32   The P-CSCF2 forwards the 200 OK response to UE-1   via I/S-CSCF2, I/SCSCF1, and P-CSCF1. Table 6.2.1.2-5
33   shows the 200 OK message going from P-CSCF1 to UE-1.

1 **Table 6.2.1.2-5 200 OK (P-CSCF1 to UE-1)**

```
SIP/2.0 200 OK
From: <sip:UE-1@domain1.net>;tag=169f498-0-13c4-78e-2044f20e-78e
To: <sip:UE-2@domain2.net>;tag=169f7f8-0-13c4-9f6-33835972-9f6
Call-ID: 16a1b80-0-13c4-78e-6a540802-78e@10.20.1.100
CSeq: 1 INVITE
Record-Route:<sip:pcscf2.domain2.net;lr>,<sip:scscf2.domain2.net;lr>,<sip:scscf1.domain1.net;lr>,
<sip:pcscf1.domain1.net:7531;lr;comp=sigcomp>
Via: SIP/2.0/UDP 10.20.1.100:1357;comp=sigcomp;branch=z9hG4bK-78f-1d86b6-24d29348
Contact: <sip:UE-2@10.30.1.24:8805;comp=sigcomp>
Content-Length: 0
```

2
3   26. ACK (UE-1 to P-CSCF1)
4   UE responds to the 200 OK with an ACK request sent to P-CSCF1. (Table 6.2.1.2-6).

5 **Table 6.2.1.2-6 ACK (UE-1 to P-CSCF1)**

6

```
ACK sip:UE-2@10.30.1.24:8805;comp=sigcomp SIP/2.0
From: <sip:UE-1@domain1.net>;tag=169f498-0-13c4-78e-2044f20e-78e
To: <sip:UE-2@domain2.net>;tag=169f7f8-0-13c4-9f6-33835972-9f6
Call-ID: 16a1b80-0-13c4-78e-6a540802-78e@10.20.1.100
CSeq: 1 ACK
Via: SIP/2.0/UDP 10.20.1.100:5060;comp=sigcomp;branch=z9hG4bK-792-1d9290-49ff0c31
Max-Forwards: 70
 Route: <sip:pcscf1.domain1.net:7531;lr;comp=sigcomp>,<sip:scscf1.domain1.net;lr>,
 <sip:scscf2.domain2.net;lr>,
 <sip:pcscf2.domain2.net;lr>
Content-Length: 0
```

7
8   27-30. ACK (P-CSCF1 to UE-2)
9   The P-CSCF1 forwards the ACK response to UE-2 via I/S-CSCF1, I/S-CSCF2, and P-CSCF2.

1

6.2.2   Scenario 2

This section covers the scenario where the originating UE's resources are ready before sending INVITE, and the terminating UE's resources are not ready before sending the first provisional response

6.2.2.1   Assumptions

6.2.2.2   Call Flow

This scenario assumes that the terminating UE, UE-2, does not have the resource ready before sending the first provisional response. UE-2 may send a 183 (Session Progress) response unreliably. At the same time, UE-2 also starts the resource reservation process. Once the resource is ready at UE-2 side, the UE-2 sends 180 (Ringing) response reliably to UE-1, including an SDP answer to indicate that the resource is ready. The SDP offer/answer exchanges are the same as those in Scenario 1.

Figure 2 Originating UE resource ready, terminating UE resource not ready

6.2.3 Scenario 3
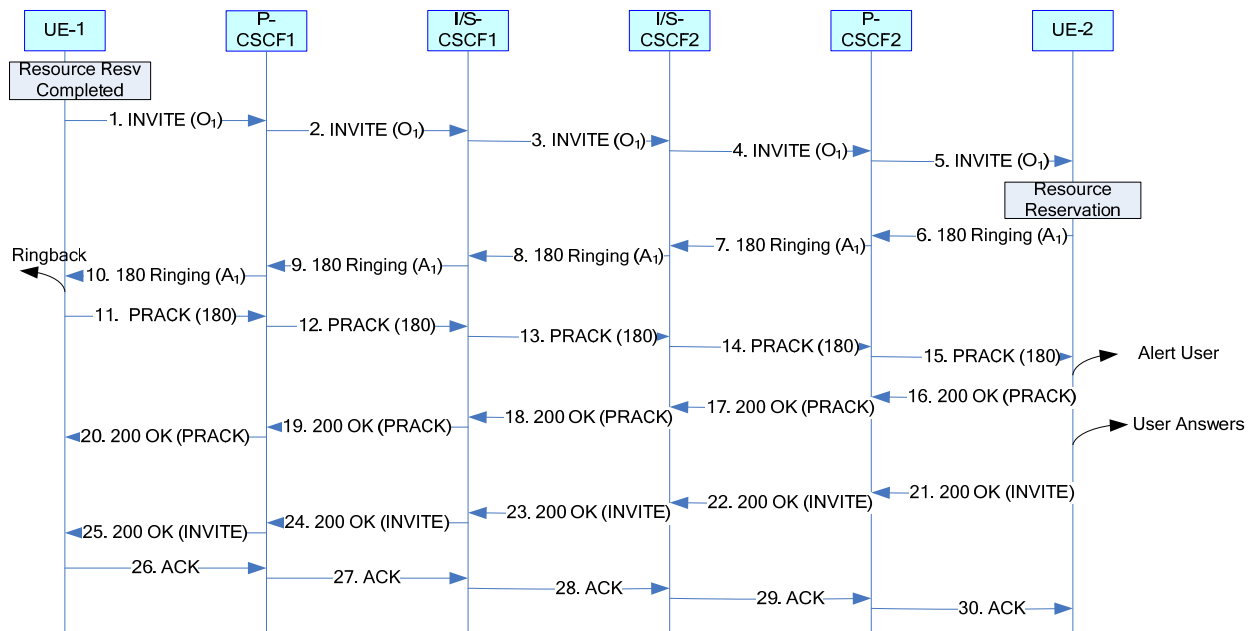
This section covers the scenario where the originating UE's resources are not ready before sending INVITE, and terminating UE's resources are ready before sending the first provisional response.

6.2.3.1 Assumptions

The call flow in this section assumes that the originating UE, UE-1, has resources ready before sending the PRACK request to the first provisional response. In case UE-1's resources are not ready before sending the PRACK request for the first provisional response (183), then UE-1 needs to send an UPDATE request to UE-2 to indicate that resources are ready, after the resource reservation is completed.

6.2.3.2 Call flow

The following section applies to the case where UE-1 has no resource reserved before sending the initial INVITE request to UE-2.

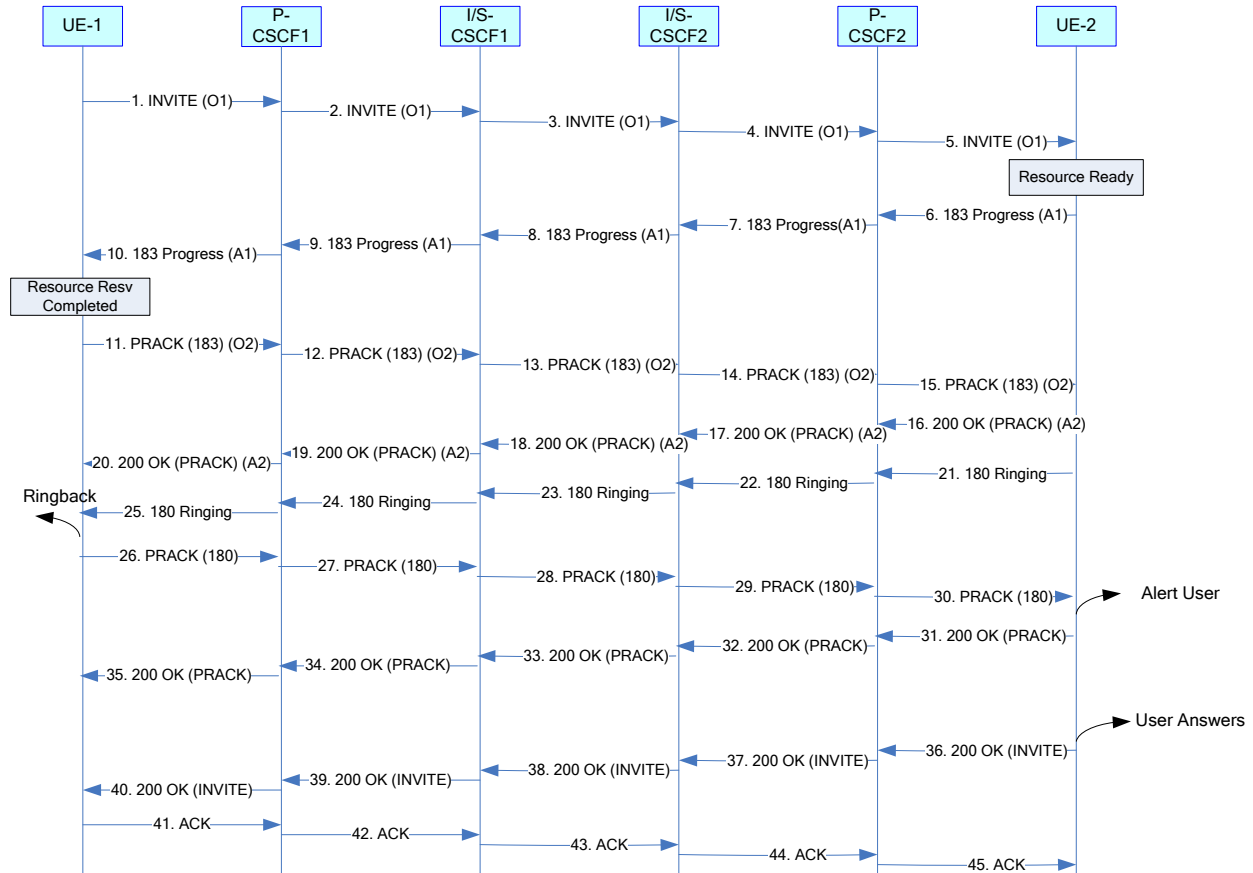**Figure 3 Originating UE not ready, Terminating UE ready**

1-5. INVITE (UE-1 to P-CSCF1)

UE-1 determines the set of codecs or media streams that it wishes to support for the session. It builds a SDP containing characteristics of each codec, and assigns local port numbers for each possible media flow. Multiple media flows may be offered, and for each media flow (m= line in SDP), there may be multiple codec choices offered.

For this example, it is assumed that UE-1 desires to establish a multimedia session comprising a video stream and an audio stream. The video stream supports one H.263 codec. The audio stream supports both EVRC and SMV codec.

UE-1 does not indicate that precondition is required for this session, but indicates that it is supported. (This approach optimizes compatibility and performance when interworking with 3rd party (non-MMD) terminals). In the SDP body, UE-1 indicates the current resource status and that the desired resource status is optional. UE-1 also sets every media stream to inactive mode by using the 'a=inactive' SDP attribute in the SDP offer. Detecting the QoS precondition content in the SDP, UE-2 indicates resource reservation, but the session can continue regardless of whether or not this reservation is possible.

UE-1 does not indicate that reliable provisional responses are required, but indicates that they are supported. This gives UE-2 the ability to reliably send only those responses that are most appropriate.

17

1

2   **Table 6.2.3.2-7 INVITE (UE-1 to P-CSCF1)**

```
INVITE sip:UE-2@domain2.net SIP/2.0
Via: SIP/2.0/UDP 100.200.1.1:1357;comp=sigcomp;branch=z9hG4bK4d29348
From: <sip:UE-1@domain1.net>;tag= a48s
To: <sip:UE-2@domain2.net>
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@domain1.net
CSeq: 1 INVITE
Max-Forwards: 70
Route: <sip:pcscf1.domain1.net:7531;lr;comp=sigcomp>, <sip:scscf1.domain1.net;lr>
P-Preferred-Identity: "User-1" <sip:UE-1@domain1.net>
P-Access-Network-Info: 3GPP2-1X-HRPD; ci-3gpp2=1234123412341234123412341234123411
Privacy: none
Contact: <sip:UE-1@100.200.1.1:1357;comp=sigcomp>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Require: sec-agree
Proxy-Require: sec-agree
Supported: 100rel, precondition
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=9876543; spi-s=3456789;
  port-c=8642; port-s=7531
Content-Type: application/sdp
Content-Length: xxx


v=0
o=- 2987935614 2987935614 IN IP4 100.200.1.1
s=-
c=IN IP4 100.200.1.1
t=0 0
m=audio 10500 RTP/AVP 97 99
b=AS:25.4
a=inactive
a=curr:qos local none
a=curr:qos remote none
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
a=rtpmap:97 EVRC/8000
a=ptime:20
a=rtpmap:99 SMV/8000
m=video 10600 RTP/AVP 34
b=AS:75
a=inactive
a=curr:qos local none
a=curr:qos remote none
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
a=rtpmap:34 H263/90000
a=cif:1
```

3
4   6-10. 183 (P-CSCF1 to UE-1)
5   UE-2 accepts both video and audio streams, and chooses EVRC as the codec for the audio stream. An SDP answer is
6   included to assist UE-1 in completing resource reservation as early as possible. The SDP answer includes
7   precondition status.
8

1   **Table 6.2.3.2-8 183 Session Progress (P-CSCF1 to UE-1)**

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 100.200.1.1:1357;comp=sigcomp;branch= z9hG4bK4d29348
From: <sip:UE-1@domain1.net>;tag=a48s
To: <sip:UE-2@domain2.net>;tag=a9f6
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@domain1.net
CSeq: 1 INVITE
Record-Route:<sip:pcscf2.domain2.net;lr>,<sip:scscf2.domain2.net;lr>,
  <sip:scscf1.domain1.net;lr>,<sip:pcscf1.domain1.net:7531;lr;comp=sigcomp>
Contact: <sip:UE-2@100.300.1.2:2345;comp=sigcomp>
P-Asserted-Identity: "User 2" <sip:UE-2@domain2.net>, <tel:+1-972-321-9876>
Privacy: none
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Require: 100rel, precondition
RSeq: 1000
Content-Type: application/sdp
Content-Length: xxx

v=0
o=- 35270718123 35270718123 IN IP4 100.300.1.2
s=-
c=IN IP4 100.300.1.2
t=0 0
m=audio 10700 RTP/AVP 97
b=AS:25.4
a=inactive
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
a=conf:qos remote sendrecv
a=rtpmap:97 EVRC/8000
a=ptime:20
m=video 10702 RTP/AVP 34
b=AS:75
a=inactive
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
a=conf:qos remote sendrecv
a=rtpmap:34 H263/90000
a=cif:1
```

2
3   11-15. PRACK (UE-1 to P-CSCF1)
4   UE-1 acknowledges the 183 (Session Progress) response from UE-2 with a PRACK request. Resource reservation at
5   UE-1 is assumed to have been completed at some point prior to sending the PRACK request. The local resource
6   status is included in the SDP offer.
7
8

Table 6.2.3.2-3 – PRACK (UE-1 to P-CSCF1)

```
PRACK sip:UE-2@100.300.1.2:2345;comp=sigcomp SIP/2.0
Via: SIP/2.0/UDP 100.200.1.1:1357;comp=sigcomp;branch=z9hG4bK4d29348
From: <sip:UE-1@domain1.net>;tag=a48s
To: <sip:UE-2@domain2.net>;tag=a9f6
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@domain1.net
P-Access-Network-Info: 3GPP2-1X-HRPD; ci-3gpp2=12341234123412341234123412341234123411
CSeq: 2 PRACK
Max-Forwards: 70
Route: <sip:pcscf1.domain1.net:7531;lr;comp=sigcomp>,<sip:scscf1.domain1.net;lr>,
<sip:scscf2.domain2.net;lr>, <sip:pcscf2.domain2.net;lr>
Require: precondition
RAck: 1000 1 INVITE
Content-Type: application/sdp
Content-Length: xxx


v=0
o=- 2987935614 2987935615 IN IP4 100.200.1.1
s=-
c=IN IP4 100.200.1.1
t=0 0
m=audio 10500 RTP/AVP 97
b=AS:25.4
a=sendrecv
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
a=rtpmap:97 EVRC/8000
a=ptime:20
m=video 10600 RTP/AVP 34
b=AS:75
a=sendrecv
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
a=rtpmap:34 H263/90000
a=cif:1
```

16-20. 200 OK (P-CSCF1 to UE-1)

The 200 OK response is generated by UE-2 to acknowledge the reception of the PRACK request. UE-2 has completed resource reservation for both video and audio streams. The response will include an SDP answer to reflect successful resource reservation.

1   **Table 6.2.3.2-4 - 200 OK (P-CSCF1 to UE-1)**

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 100.200.1.1:1357;comp=sigcomp;branch=z9hG4bK4d29348
From: <sip:UE-1@domain1.net>;tag=a48s
To: <sip:UE-2@domain2.net>;tag=a9f6
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@domain1.net
CSeq: 2 PRACK
Require: precondition
Content-Length: xxx


v=0
o=- 35270718123 35270718124 IN IP4 100.300.1.2
s=-
c=IN IP4 100.300.1.2
t=0 0
m=audio 10700 RTP/AVP 97
b=AS:25.4
a=sendrecv
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
a=rtpmap:97 EVRC/8000
a=ptime:20
m=video 10702 RTP/AVP 34
b=AS:75
a=sendrecv
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
a=rtpmap:34 H263/90000
a=cif:1
```

2
3 21-25. 180 Ringing (P-CSCF1 to UE-1)
4 UE-2 generates a 180 (Ringing) response to UE-1 and sends it reliably. UE-2 also alerts the called user of the
5 incoming call.
6
7   **Table 6.2.3.2-5 - 180 Ringing (P-CSCF1 to UE-1)**

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 100.200.1.1:1357;comp=sigcomp;branch= z9hG4bK4d29348
From: <sip:UE-1@domain1.net>;tag=a48s
To: <sip:UE-2@domain2.net>;tag=a9f6
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@domain1.net
CSeq: 1 INVITE
Record-Route:<sip:pcscf2.domain2.net;lr>,<sip:scscf2.domain2.net;lr>,
  <sip:scscf1.domain1.net;lr>,<sip:pcscf1.domain1.net:7531;lr;comp=sigcomp>
Contact: <sip:UE-2@100.300.1.2:2345;comp=sigcomp>
P-Asserted-Identity: "User 2" <sip:UE-2@domain2.net>, <tel:+1-972-321-9876>
Privacy: none
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Require: 100rel
RSeq: 1001
Content-Length: 0
```

8
9 26-30. PRACK (UE-1 to P-CSCF1)
10 UE-1 acknowledges the 180 (Ringing) response from UE-2 with a PRACK request.
11

1 **Table 6.2.3.2-6 - PRACK (UE-1 to P-CSCF1)**

```
PRACK sip:UE-2@100.300.1.2:2345;comp=sigcomp SIP/2.0
Via: SIP/2.0/UDP 100.200.1.1:1357;comp=sigcomp;branch=z9hG4bK4d29348
From: <sip:UE-1@domain1.net>;tag=a48s
To: <sip:UE-2@domain2.net>;tag=a9f6
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@domain1.net
P-Access-Network-Info: 3GPP2-1X-HRPD; ci-3gpp2=12341234123412341234123412341234123411
CSeq: 3 PRACK
Max-Forwards: 70
Route: <sip:pcscf1.domain1.net:7531;lr;comp=sigcomp>,<sip:scscf1.domain1.net;lr>,
<sip:scscf2.domain2.net;lr>, <sip:pcscf2.domain2.net;lr>
RAck: 1001 1 INVITE
Content-Length: 0
```

2
3
4 31-35. 200 OK (P-CSCF1 to UE-1)
5 The 200 OK response is generated by UE-2 to acknowledge the reception of the PRACK request.

6 **Table 6.2.3.2-7 - 200 OK (P-CSCF1 to UE-1)**

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 100.200.1.1:1357;comp=sigcomp;branch=z9hG4bK4d29348
From: <sip:UE-1@domain1.net>;tag=a48s
To: <sip:UE-2@domain2.net>;tag=a9f6
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@domain1.net
CSeq: 3 PRACK
Content-Length: 0
```

7
8 36-40. 200 OK (P-CSCF1 to UE-1)
9 When the user at UE-2 answers the call, UE-2 generates a 200 OK response towards UE-1 to answer the INVITE
10 request.

11 **Table 6.2.3.2-8 - 200 OK (P-CSCF1 to UE-1)**

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 100.200.1.1:1357;comp=sigcomp;branch=z9hG4bK4d29348
From: <sip:UE-1@domain1.net>;tag=a48s
To: <sip:UE-2@domain2.net>;tag=a9f6
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@domain1.net
CSeq: 1 INVITE
Record-Route:<sip:pcscf2.domain2.net;lr>,<sip:scscf2.domain2.net;lr>,<sip:scscf1.domai
n1.net;lr>, <sip:pcscf1.domain1.net:7531;lr;comp=sigcomp>
Contact: <sip:UE-2@100.300.1.2:2345;comp=sigcomp>
Content-Length: 0
```

12
13 41-45. ACK (UE-1 to P-CSCF1)
14 UE-1 sends an ACK request in response to the 200 OK response from UE-2.

15 **Table 6.2.3.2-9 - ACK (UE-1 to P-CSCF1)**

```
ACK sip:UE-2@100.300.1.2:2345;comp=sigcomp SIP/2.0
Via: SIP/2.0/UDP 100.200.1.1:1357;comp=sigcomp;branch=z9hG4bK4d29348
From: <sip:UE-1@domain1.net>;tag=a48s
To: <sip:UE-2@domain2.net>;tag=a9f6
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@domain1.net
CSeq: 1 ACK
Max-Forwards: 70
Route: <sip:pcscf1.domain1.net:7531;lr;comp=sigcomp>,<sip:scscf1.domain1.net;lr>,
<sip:scscf2.domain2.net;lr>, <sip:pcscf2.domain2.net;lr>
Content-Length: 0
```

16

1

6.2.4   Scenario 4

3
4   This section covers the scenario where the originating UE's resources are not ready before sending INVITE,
5   and the terminating UE's resources are not ready before sending the first provisional response.
6

6.2.4.1   Assumptions

8   This scenario assumes that the originating UE, UE-1, completes resource reservation before, or shortly
9   after, the arrival of the 183 (Session Progress) response from UE-2, and that the terminating UE completes
10   local resource reservation prior to sending 180 (Ringing) message.
11
12   In case the originating UE, UE-1, has not completed its resource reservation before sending the PRACK
13   request to the first provisional response (183), the UE-1 needs to send an UPDATE request to indicate
14   resource ready once it completes the resource reservation. Otherwise , if the terminating UE, UE-2, cannot
15   finish resource reservation before the offer/answer wherein the UE-1 indicates resources are ready, UE-2
16   can send a 180 (Ringing) response as soon as its resources are reserved without going through further
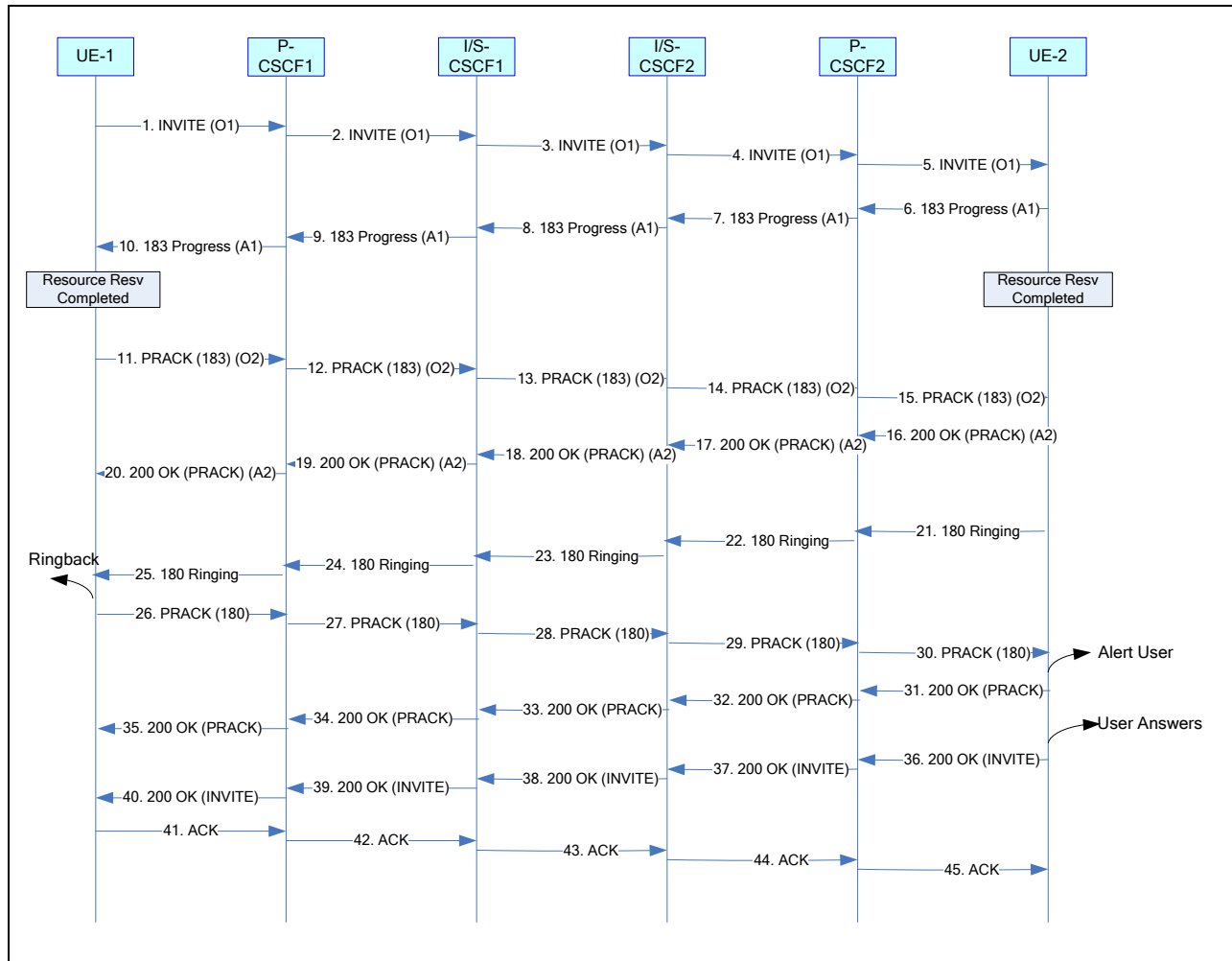17   offer/answer exchanges.
18
19   Note: The UE-1/UE-2 may start reservation anytime after it sends/receives the initial INVITE.

6.2.4.2   Call flow

21

Figure 4 Originating UE not ready, Terminating UE not ready

1-5. INVITE (UE-1 to P-CSCF1)

UE-1 determines the set of codecs or media streams that it wishes to support for the session. It builds a SDP containing characteristics of each codec, and assigns local port numbers for each possible media flow. Multiple media flows may be offered, and for each media flow (m= line in SDP), there may be multiple codec choices offered.

For this example, it is assumed that UE-1 desires to establish a multimedia session comprising a video stream and an audio stream. The video stream supports one H.263 codec. The audio stream supports both EVRC and SMV codecs.

UE-1 does not indicate that precondition is required for this session, but that it is supported. (This approach optimizes compatibility and performance when interworking with 3rd party (non-MMD) terminals). In the SDP body, UE-1 indicates the current resource status and that the desired resource status is optional. UE-1 also sets every media stream to inactive mode by using the 'a=inactive' SDP attribute in the SDP offer. Detecting the QoS precondition content in the SDP, UE-2 indicates resource reservation, but the session can continue regardless of whether or not this reservation is possible.

UE-1 does not indicate that reliable provisional responses are required, but indicates that they are supported. This gives UE-2 the ability to reliably send only those responses that are most appropriate.

UE-1 may perform local resource reservation at any time it feels it has enough information to begin this process.

24

1   This is assumed to be after sending the INVITE request, but could be sooner.

2   **Table 6.2.4.2-1 - INVITE (UE-1 to P-CSCF1)**

```
INVITE sip:UE-2@domain2.net SIP/2.0
Via: SIP/2.0/UDP 100.200.1.1:1357;comp=sigcomp;branch=z9hG4bK4d29348
From: <sip:UE-1@domain1.net>;tag= a48s
To: <sip:UE-2@domain2.net>
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@domain1.net
CSeq: 1 INVITE
Max-Forwards: 70
Route: <sip:pcscf1.domain1.net:7531;lr;comp=sigcomp>, <sip:scscf1.domain1.net;lr>
P-Preferred-Identity: "User-1" <sip:UE-1@domain1.net>
P-Access-Network-Info: 3GPP2-1X-HRPD; ci-3gpp2=1234123412341234123412341234123411
Privacy: none
Contact: <sip:UE-1@100.200.1.1:1357;comp=sigcomp>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Require: sec-agree
Proxy-Require: sec-agree
Supported: 100rel, precondition
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=9876543; spi-s=3456789;
  port-c=8642; port-s=7531
Content-Type: application/sdp
Content-Length: xxx

v=0
o=- 2987935614 2987935614 IN IP4 100.200.1.1
s=-
c=IN IP4 100.200.1.1
t=0 0
m=audio 10500 RTP/AVP 97 99
b=AS:25.4
a=inactive
a=curr:qos local none
a=curr:qos remote none
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
a=rtpmap:97 EVRC/8000
a=ptime:20
a=rtpmap:99 SMV/8000
m=video 10600 RTP/AVP 34
b=AS:75
a=inactive
a=curr:qos local none
a=curr:qos remote none
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
a=rtpmap:34 H263/90000
a=cif:1
```

3
4
5   6-10. 183 Progress (P-CSCF1 to UE-1)
6   UE-2 accepts both video and audio streams, and chooses EVRC as the codec for the audio stream. An SDP answer is
7   included to assist UE-1 in completing resource reservation as early as possible. The SDP answer includes
8   precondition status.

1 **Table 6.2.4.2-2 - 183 Session Progress (P-CSCF1 to UE-1)**

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 100.200.1.1:1357;comp=sigcomp;branch= z9hG4bK4d29348
From: <sip:UE-1@domain1.net>;tag=a48s
To: <sip:UE-2@domain2.net>;tag=a9f6
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@domain1.net
CSeq: 1 INVITE
Record-Route:<sip:pcscf2.domain2.net;lr>,<sip:scscf2.domain2.net;lr>,
  <sip:scscf1.domain1.net;lr>,<sip:pcscf1.domain1.net:7531;lr;comp=sigcomp>
Contact: <sip:UE-2@100.300.1.2:2345;comp=sigcomp>
P-Asserted-Identity: "User 2" <sip:UE-2@domain2.net>, <tel:+1-972-321-9876>
Privacy: none
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Require: 100rel, precondition
RSeq: 1000
Content-Type: application/sdp
Content-Length: xxx

v=0
o=- 35270718123 35270718123 IN IP4 100.300.1.2
s=-
c=IN IP4 100.300.1.2
t=0 0
m=audio 10700 RTP/AVP 97
b=AS:25.4
a=inactive
a=curr:qos local none
a=curr:qos remote none
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
a=conf:qos remote sendrecv
a=rtpmap:97 EVRC/8000
a=ptime:20
m=video 10702 RTP/AVP 34
b=AS:75
a=inactive
a=curr:qos local none
a=curr:qos remote none
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
a=conf:qos remote sendrecv
a=rtpmap:34 H263/90000
a=cif:1
```

2
3
4 11-15. PRACK (UE-1 to P-CSCF1)
5 UE-1 acknowledges the 183 (Session Progress) response from UE-2 with a PRACK request. Resource reservation at
6 UE-1 is assumed to have completed at some point prior to sending the PRACK request. The local resource status is
7 included in the SDP offer.
8
9

1   **Table 6.2.4.2-3 - PRACK (UE-1 to P-CSCF1)**

```
PRACK sip:UE-2@100.300.1.2:2345;comp=sigcomp SIP/2.0
Via: SIP/2.0/UDP 100.200.1.1:1357;comp=sigcomp;branch=z9hG4bK4d29348
From: <sip:UE-1@domain1.net>;tag=a48s
To: <sip:UE-2@domain2.net>;tag=a9f6
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@domain1.net
P-Access-Network-Info: 3GPP2-1X-HRPD; ci-3gpp2=12341234123412341234123412341234123411
CSeq: 2 PRACK
Max-Forwards: 70
Route: <sip:pcscf1.domain1.net:7531;lr;comp=sigcomp>,<sip:scscf1.domain1.net;lr>,
<sip:scscf2.domain2.net;lr>, <sip:pcscf2.domain2.net;lr>
Require: precondition
RAck: 1000 1 INVITE
Content-Type: application/sdp
Content-Length: xxx


v=0
o=- 2987935614 2987935615 IN IP4 100.200.1.1
s=-
c=IN IP4 100.200.1.1
t=0 0
m=audio 10500 RTP/AVP 97
b=AS:25.4
a=sendrecv
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
a=rtpmap:97 EVRC/8000
a=ptime:20
m=video 10600 RTP/AVP 34
b=AS:75
a=sendrecv
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
a=rtpmap:34 H263/90000
a=cif:1
```

2
3
4   16-20. 200 OK (P-CSCF1 to UE-1)
5   The 200 OK response is generated by UE-2 to acknowledge the reception of the PRACK request. UE-2 has
6   completed resource reservation for both video and audio streams. The response will include an SDP answer to
7   reflect successful resource reservation.

1 **Table 6.2.4.2-4 - 200 OK (P-CSCF1 to UE-1)**

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 100.200.1.1:1357;comp=sigcomp;branch=z9hG4bK4d29348
From: <sip:UE-1@domain1.net>;tag=a48s
To: <sip:UE-2@domain2.net>;tag=a9f6
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@domain1.net
CSeq: 2 PRACK
Require: precondition
Content-Length: xxx

v=0
o=- 35270718123 35270718124 IN IP4 100.300.1.2
s=-
c=IN IP4 100.300.1.2
t=0 0
m=audio 10500 RTP/AVP 97
b=AS:25.4
a=sendrecv
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
a=rtpmap:97 EVRC/8000
a=ptime:20
m=video 10600 RTP/AVP 34
b=AS:75
a=sendrecv
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
a=rtpmap:34 H263/90000
a=cif:1
```

2
3
4 21-25. 180 Ringing (P-CSCF1 to UE-1)
5 UE-2 generates a 180 (Ringing) response to UE-1 and sends it reliably.
6
7 **Table 6.2.4.2-5 - 180 Ringing (P-CSCF1 to UE-1)**

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 100.200.1.1:1357;comp=sigcomp;branch= z9hG4bK4d29348
From: <sip:UE-1@domain1.net>;tag=a48s
To: <sip:UE-2@domain2.net>;tag=a9f6
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@domain1.net
CSeq: 1 INVITE
Record-Route:<sip:pcscf2.domain2.net;lr>,<sip:scscf2.domain2.net;lr>,
  <sip:scscf1.domain1.net;lr>,<sip:pcscf1.domain1.net:7531;lr;comp=sigcomp>
Contact: <sip:UE-2@100.300.1.2:2345;comp=sigcomp>
P-Asserted-Identity: "User 2" <sip:UE-2@domain2.net>, <tel:+1-972-321-9876>
Privacy: none
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Require: 100rel
RSeq: 1001
Content-Length: 0
```

8
9
10 26-30. PRACK (UE-1 to P-CSCF1)
11 UE-1 acknowledges the 180 (Ringing) response from UE-2 with a PRACK request.
12

1   **Table 6.2.4.2-6 - PRACK (UE-1 to P-CSCF1)**

```
PRACK sip:UE-2@100.300.1.2:2345;comp=sigcomp SIP/2.0
Via: SIP/2.0/UDP 100.200.1.1:1357;comp=sigcomp;branch=z9hG4bK4d29348
From: <sip:UE-1@domain1.net>;tag=a48s
To: <sip:UE-2@domain2.net>;tag=a9f6
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@domain1.net
P-Access-Network-Info: 3GPP2-1X-HRPD; ci-3gpp2=12341234123412341234123412341234123411
CSeq: 3 PRACK
Max-Forwards: 70
Route: <sip:pcscf1.domain1.net:7531;lr;comp=sigcomp>,<sip:scscf1.domain1.net;lr>,
<sip:scscf2.domain2.net;lr>, <sip:pcscf2.domain2.net;lr>
RAck: 1001 1 INVITE
Content-Length: 0
```

2
3
4   31-35. 200 OK (P-CSCF1 to UE-1)
5   The 200 OK response is generated by UE-2 to acknowledge the reception of the PRACK request.

6   **Table 6.2.4.2-7 - 200 OK (P-CSCF1 to UE-1)**

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 100.200.1.1:1357;comp=sigcomp;branch=z9hG4bK4d29348
From: <sip:UE-1@domain1.net>;tag=a48s
To: <sip:UE-2@domain2.net>;tag=a9f6
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@domain1.net
CSeq: 3 PRACK
Content-Length: 0
```

7
8
9   36-40. 200 OK (P-CSCF1 to UE-1)
10 When the user at UE-2 answers the call, UE-2 generates a 200 OK response towards UE-1 to answer the INVITE
11 request.

12   **Table 6.2.4.2-8 - 200 OK (P-CSCF1 to UE-1)**

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 100.200.1.1:1357;comp=sigcomp;branch=z9hG4bK4d29348
From: <sip:UE-1@domain1.net>;tag=a48s
To: <sip:UE-2@domain2.net>;tag=a9f6
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@domain1.net
CSeq: 1 INVITE
Record-Route:<sip:pcscf2.domain2.net;lr>,<sip:scscf2.domain2.net;lr>,<sip:scscf1.domain1.net;lr>,
<sip:pcscf1.domain1.net:7531;lr;comp=sigcomp>
Contact: <sip:UE-2@100.300.1.2:2345;comp=sigcomp>
Content-Length: 0
```

13
14
15 41-45. ACK (UE-1 to P-CSCF1)
16 UE-1 sends an ACK request in response to the 200 OK response from UE-2.

1   **Table 6.2.4.2-9 - ACK (UE-1 to P-CSCF1)**

```
ACK sip:UE-2@100.300.1.2:2345;comp=sigcomp SIP/2.0
Via: SIP/2.0/UDP 100.200.1.1:1357;comp=sigcomp;branch=z9hG4bK4d29348
From: <sip:UE-1@domain1.net>;tag=a48s
To: <sip:UE-2@domain2.net>;tag=a9f6
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@domain1.net
CSeq: 1 ACK
Max-Forwards: 70
Route: <sip:pcscf1.domain1.net:7531;lr;comp=sigcomp>,<sip:scscf1.domain1.net;lr>,
<sip:scscf2.domain2.net;lr>, <sip:pcscf2.domain2.net;lr>
Content-Length: 0
```

2
3
4

Appendix A (Informative): VoIP example with QoS reservation, activation, and updating at RAN

In this section, VoIP application is used as an example to show how the QoS reservation, activation, and updating process at RAN and IP-CAN works together with the call flows described in this document. The call flow described in Scenario 1 is used. The following call flows are examples where an HRPD Rev-A (or later) access network is used.
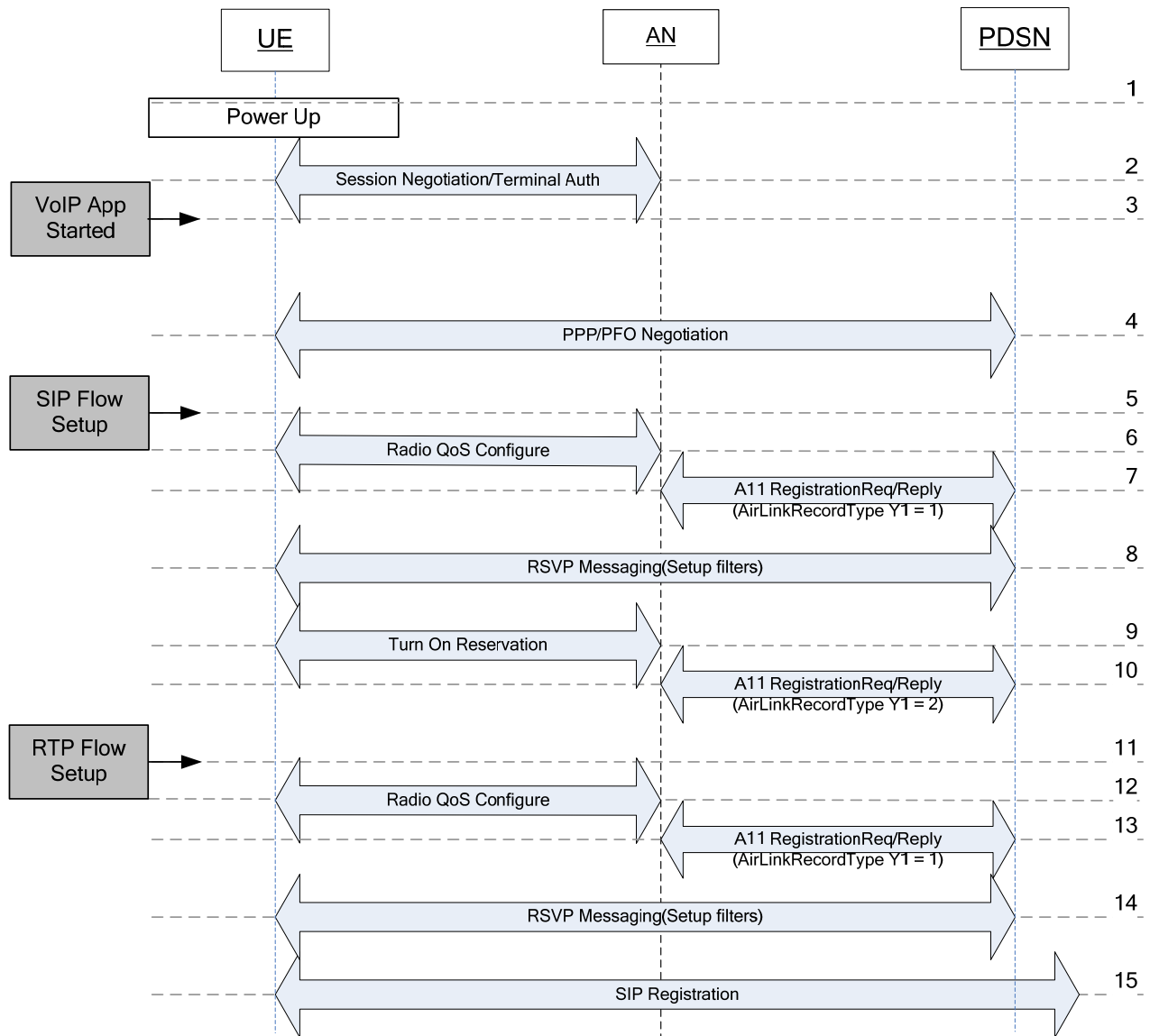
A.1 QoS Configuration for VoIP in advance

In order to reduce the call setup delay, VoIP application on the AT sets up two independent QoS flows in advance (e.g. right after the AT powers up):

> 1) SIP flow for SIP signaling messages;
> 2) RTP flow for VoIP RTP traffic.

The steps for configuration of the QoS are the same for these two flows, except for the actual activation part of the SIP flow. The SIP flow reservation always remains in the "on" state while the VoIP RTP flow needs to be explicitly turned on when actual VoIP call is being set up.

UE   AN   PDSN

Power Up

Session Negotiation/Terminal Auth

VoIP App Started

PPP/PFO Negotiation

SIP Flow Setup

Radio QoS Configure

A11 RegistrationReq/Reply
(AirLinkRecordType Y1 = 1)

RSVP Messaging(Setup filters)

Turn On Reservation

A11 RegistrationReq/Reply
(AirLinkRecordType Y1 = 2)

RTP Flow Setup

Radio QoS Configure

A11 RegistrationReq/Reply
(AirLinkRecordType Y1 = 1)

RSVP Messaging(Setup filters)

SIP Registration

1
2

Figure 5 QoS configuration for VoIP

Step 1: The UE powers up for the first time.

Step 2: UE performs session negotiation. The AN provides the list of QoS User Profiles that the AN can handle as part of the parameter negotiation for MPA/EMPA. This step also includes the terminal authentication. UE gets the RANSupportedQoSProfiles as part of the session negotiation if AN plans to send one to the UE.

Step 3: VoIP Application is started.

Step 4: The PPP/PFO negotiation with the PDSN is initiated. The user is authenticated with the AAA and the user's allowed QoS User Profiles is updated in the AN and PDSN.

Step 5: The UE starts SIP flow setup.

Step 6: The UE negotiates with the AN to configure the radio protocols requires to carry the SIP flow and once the PDSN is also successfully configured requests the AN to turn the reservation on.

Step 7: AN performs A11 signaling to establish the required A10 connections.

Step 8: The TFT filters and the channel treatment information are configured at the PDSN using RSVP messaging.

1   Step 9: The SIP flow is activated.
2   Step 10: AN informs the PDSN that the QoS flow is activated.
3   Step 11: The UE starts RTP flow setup. This may happen in parallel to step 5~10.
4   Step 12: The UE negotiates with the AN to configure the radio protocols required to carry the RTP flow.
5   Step 13: AN performs A11 signaling to establish the required A10 connections.
6   Step 14: The TFT filters and the channel treatment information are configured at the PDSN. The RTP flow is now
7   successfully configured and reservation is left in the off state.
8   Step 15: Once step 10 and step 14 are completed, The UE performs SIP registration.
9
10

11 # A.2 Resource activation on originating side

12 The call flow in this section show how the pre-reserved resources are turned on before the UE-1 makes a VoIP call.
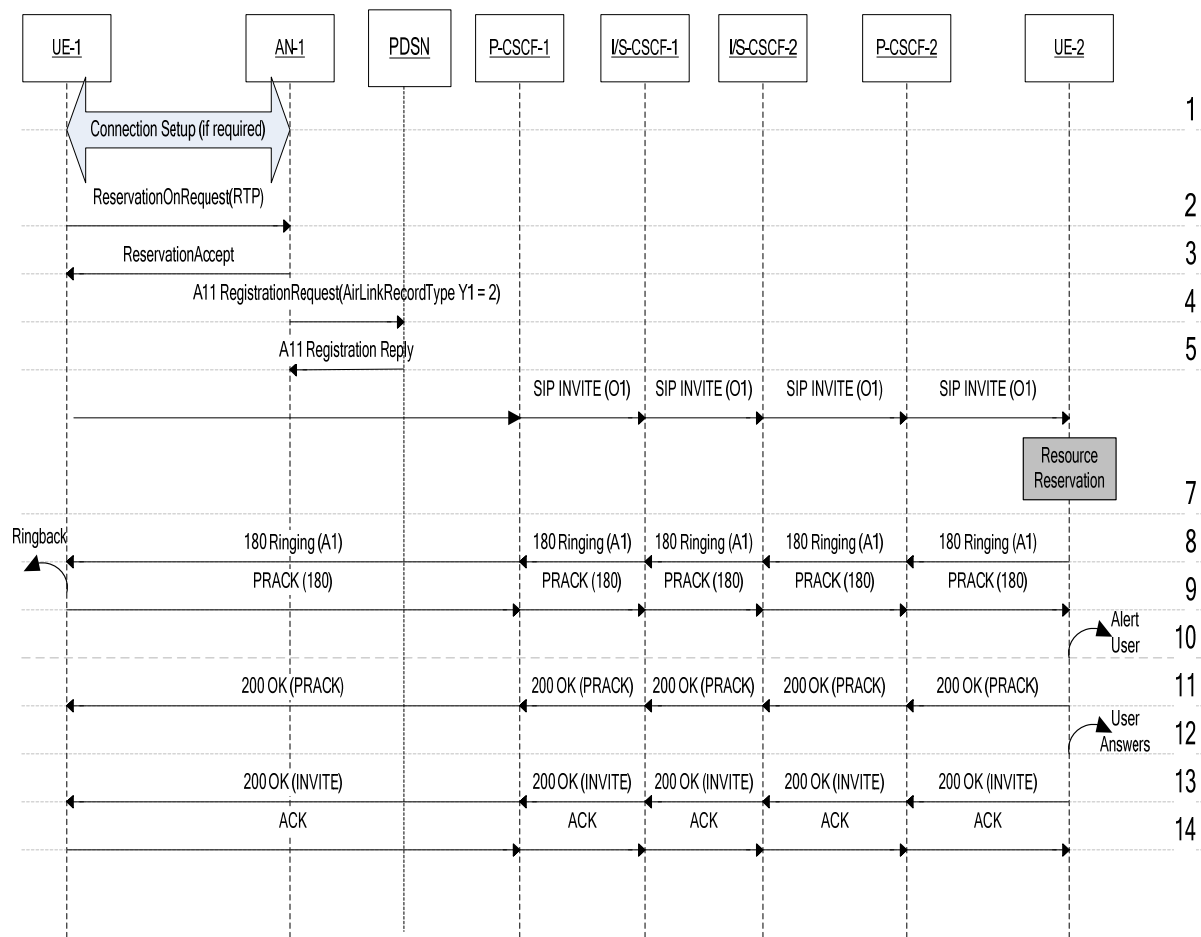
13
14

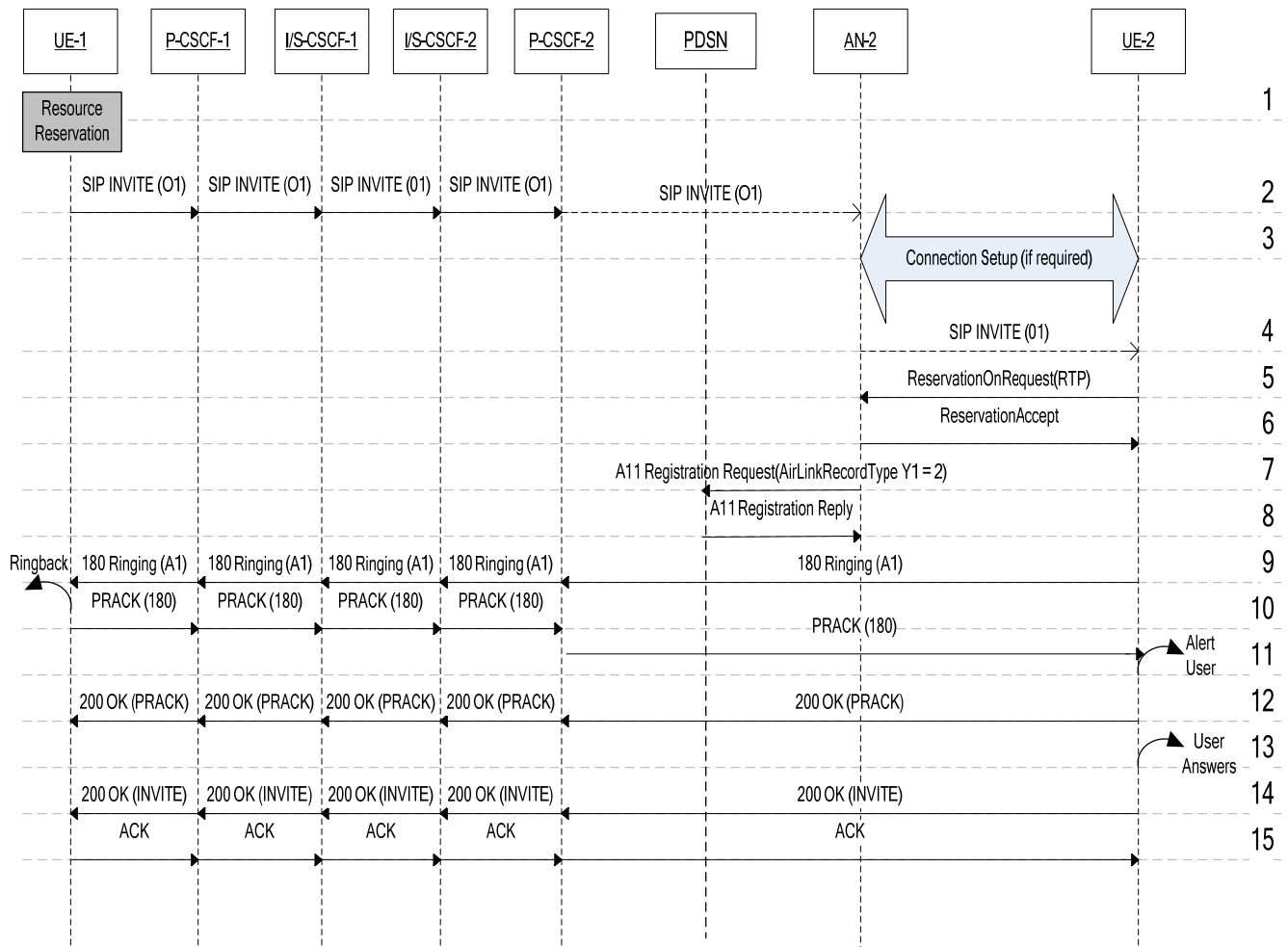15   **Figure 6 QoS Activation on Originating UE**

16 Step 1. After the mobile user initiates the VoIP call, UE-1 performs connection setup for both SIP and RTP flows as
17 described in Section A.1 if the flows are not set up in advance.
18 Step 2. UE-1 sends ReservationOnRequest to turn on the QoS for the RTP flow.
19 Step 3. AN-1 sends ReservationAccept to UE-1 indicating that the QoS for the RTP flow is changed to "on" state.
20 Step 4~5. AN-1 informs the PDSN that the QoS flow for RTP is now activated.

1   The rest of the call flow follows the procedures described in Scenario 1 where the resources at UE-1 are ready.
2
3

A.3 Resource activation on terminating side

5   This section shows how the terminating side, UE-2, quickly turns on its pre-configured QoS flow for RTP after it
6   receives the INVITE and before it sends 180 (Ringing) to indicate resource ready.
7

8
9

10   **Figure 7 QoS Activation on Terminating UE**

11   Step 1. UE-1 activates its QoS upon VoIP call initiation.
12   Step 2. UE-1 sends INVITE to UE-2 and the INVITE request is eventually forwarded to AN-2.
13   Step 3. AN-2 optionally performs connection setup (e.g., traffic channel setup etc) with UE-2.
14   Step 4. AN-2 sends the INVITE to UE-2.
15   Step 5. UE-2 sends ReservationOnRequest to turn on the QoS for the RTP flow.
16   Step 6. AN-2 sends ReservationAccept to UE-2 indicating that the QoS for the RTP flow is changed to "on" state.
17   Step 7~8. AN-2 informs the PDSN that the QoS flow for RTP is now activated.
18   The rest of the call flow follows the procedures described in Scenario 1 where the resources at UE-2 are ready

1   before the first provisional response is sent.
2
3

# A.4 Updating of resource reservation

5   In some cases, the UE may need to adjust the pre-configured QoS based on the actual codec negotiated. In this
6   section, VoIP application with multiple codecs in the initial offer is used as an example to show how QoS resources
7   can be updated without affecting the call setup delay.
8
9   For example, in the initial INVITE sent from UE-1 to UE-2, two different codecs may be offered, one is supported
10   by RS1 with FlowProfileID 256 (e.g. EVRC), while the other may be supported by RS2 with FlowProfileID 257
11   (e.g. AMR). Typically, the VoIP calls will be RS1 with RS2 calls occurring occasionally. In order to expedite the call
12   setup delay using the approach described in Sections A.1 through A.3, FlowProfileID 256 is always granted initially
13   to the UE for the pre-configured RTP QoS flow. The reservation is turned to the open state with FlowProfileID 256
14   when the UE initiates VoIP calls. If both ends actually agrees on RS2 Vocoder, the QoS flow may be switched to
15   FlowProfileID=257.
16
17

A.4.1 Resource updating on originating side

19   Call flow in this section shows how UE-1 can update its QoS to support RS2 even though QoS flow for RS1 is
20   pre-configured.
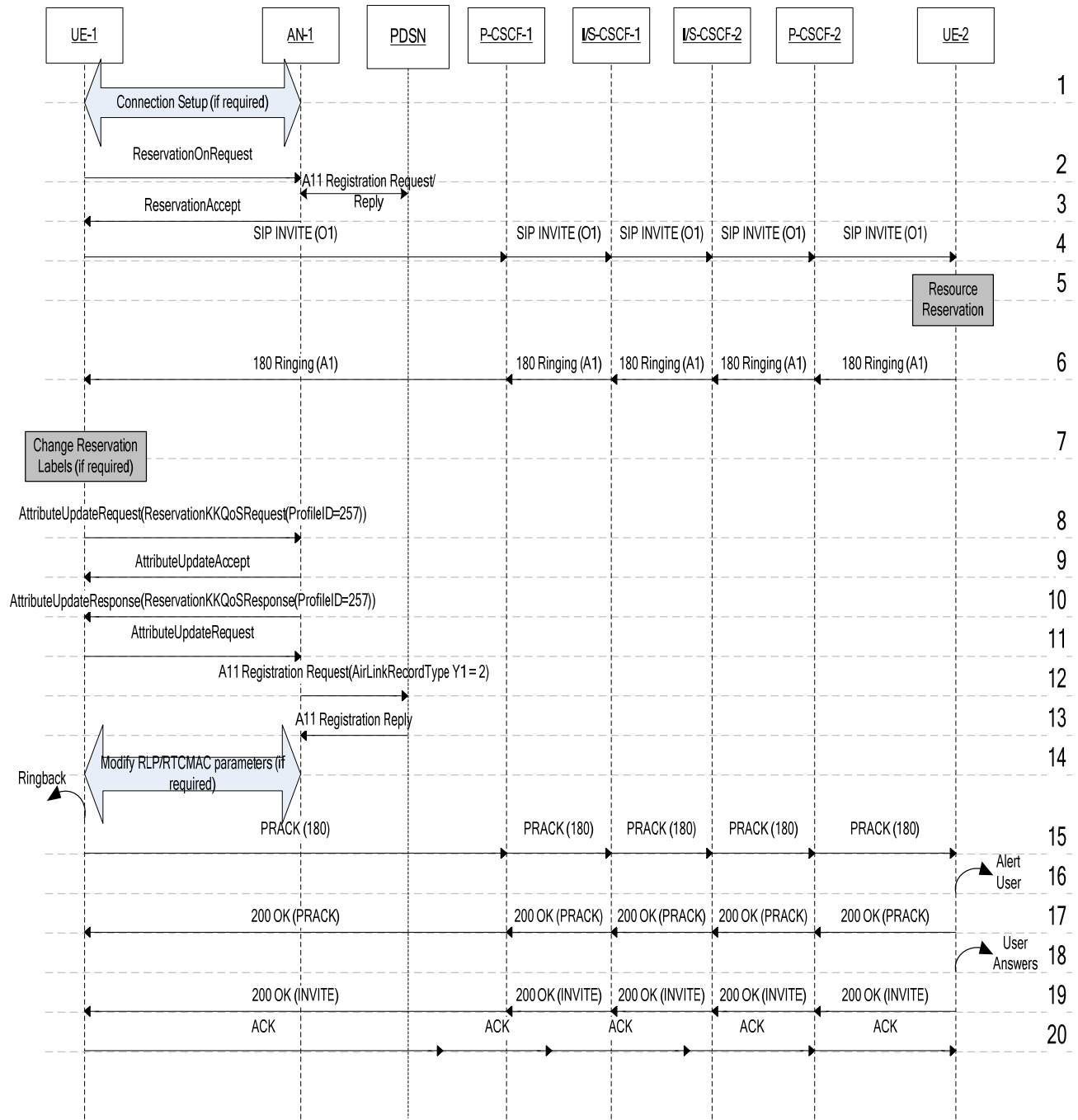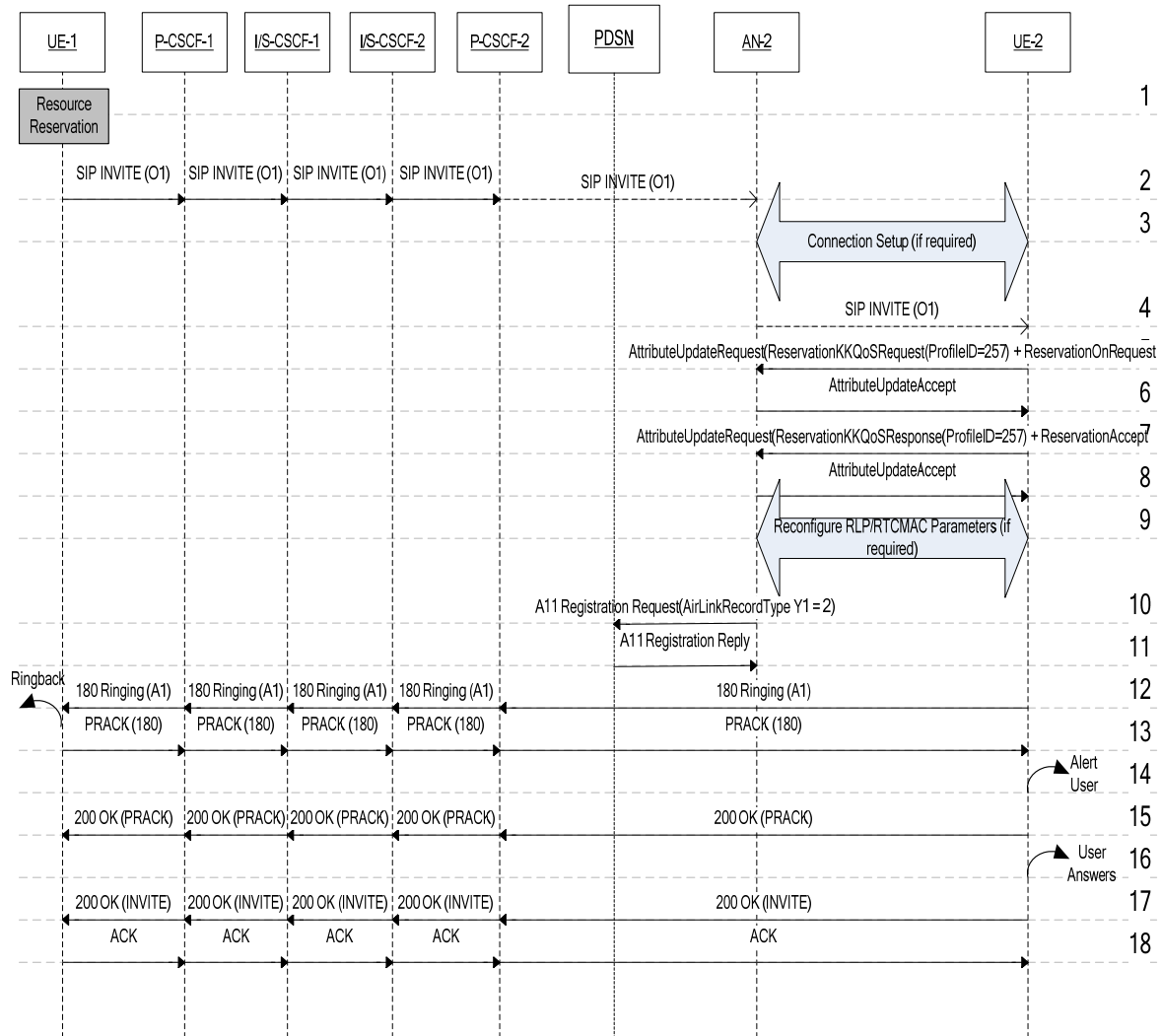
Figure 8 Resource Updating on Originating UE

Step 1. UE-1 performs connection setup if no flows have been pre-configured.

Step 2~4. UE-1 sends INVITE to UE-2 and also turns on the QoS flow for RTP stream with FlowProfileID 256.

Step 5~6. UE-2 completes resource reservation and returns 180 (Ringing) with the answer included.

Step 7. UE-1 learns that RS2 codec is chosen for the session instead of the RS1 codec. It changes reservation label if different labels are used for the two reservations.

1   Step 8~9. UE-1 requests the AN-1 to update the QoS flow profile to 257.
2   Step 10~11. AN-1 indicates to UE-1 that the flow profile has been updated to 257.
3   Step 12~13. AN-1 informs the PDSN that the updated flow has been activated.
4   Step 14. UE-1 and AN-1 updates RLP/RTCMAC parameters to match the updated flow if necessary.
5   The rest of the SIP messages are exchanged following the same procedure described in Scenario 1.
6

7   A.4.2 Resource updating on terminating side

8   Call flow in this section shows how UE-2 can update its QoS to support RS2 even though QoS flow for RS1 is
9   pre-configured.

10

11   **Figure 9 Resource Updating on Terminating UE**

12   Step 1~2. UE-1 completes resource reservation and sends INVITE to UE-2 with QoS ready indication in the offer.
13   The INVITE arrives at AN-2 first.
14   Step 3. AN-2 performs connection setup if no flows have been pre-configured.
15   Step 4. AN-2 forwards the INVITE to UE-2.
16   Step 5~6. UE-2 determines that RS2 codec will be used instead of the RS1 codec. It requests AN-2 to update the
17   QoS flow for RTP to profile 257 and at the same time requests AN-2 to turn on the reservation.

37

1   Step 7~8. AN-2 indicates to UE-2 that the flow profile has been updated to 257 and the flow is turned on.
2   Step 9. UE-2 and AN-2 updates RLP/RTCMAC parameters to match the updated flow if necessary.
3   Step 10~11. AN-2 informs the PDSN that the updated flow has been activated.
4   The rest of the SIP messages are exchanged following the same procedure described in Scenario 1.