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TECHNICAL SPECIFICATION

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Universal Mobile Telecommunications System (UMTS);
LTE;
Communication HOLD (HOLD) using IP Multimedia (IM) Core Network (CN) subsystem;
Protocol specification
(3GPP TS 24.610 version 12.6.0 Release 12)

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Foreword

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Contents

Foreword

This Technical Specification (TS) was been produced by ETSI Technical Committee Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN) and originally published as ETSI TS 183 010 [7]. It was transferred to the 3rd Generation Partnership Project (3GPP) in December 2007.

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of the present document, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

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where:

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2 presented to TSG for approval;

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y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.

z the third digit is incremented when editorial only changes have been incorporated in the document.

1 Scope

The present document specifies the stage three Protocol Description of the Communication Hold (HOLD) services, based on stages one and two of the ISDN Hold (HOLD) supplementary services. It provides the protocol details in the IP Multimedia (IM) Core Network (CN) subsystem based on the Session Initiation Protocol (SIP) and the Session Description Protocol (SDP).

The present document is applicable to User Equipment (UE) and Application Servers (AS) which are intended to support the HOLD supplementary service.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present 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 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

[1] 3GPP TS 24.229: "Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3".

[2] Void.

[3] Void

[4] IETF RFC 3264 (2002): "An Offer/Answer Model with the Session Description Protocol (SDP)".

[5] 3GPP TS 22.173: "IP Multimedia Core Network Subsystem (IMS) Multimedia Telephony Service and supplementary services; Stage 1".

[6] 3GPP TS 24.628: "Common Basic Communication procedures; Protocol specification".

[7] ETSI TS 183 010 V1.2.2: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); NGN Signalling Control Protocol; Communication HOLD (HOLD) PSTN/ISDN simulation services; Protocol specification".

[8] IETF RFC 7090 (April 2014): "Public Safety Answering Point (PSAP) Callback".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in 3GPP TS 22.173 [5] apply.

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

ACR/CB Anonymous Communication Rejection and Communication Barring
AS SIP Application Server
CDIV Communication DIVersion

CSCF	Call Session Control Function
ECT	Explicit Communication Transfer
HOLD	communication session HOLD
IMS	IP Multimedia Subsystem
IP	Internet Protocol
ISDN	Integrated Service Digital Network
MCID	Malicious Communication IDentification
OIP	Originating Identification Presentation
OIR	Originating Identification Restriction
P-CSCF	Proxy-CSCF
PSAP	Public Safety Answering Point
PSTN	Public Switched Telephone Network
S-CSCF	Serving-CSCF
SDP	Session Description Protocol
SIP	Session Initiation Protocol
TIP	Terminating Identification Presentation
TIR	Terminating Identification Restriction
UE	User Equipment

4 Communication Hold (HOLD)

4.1 Void

4.2 Description

4.2.1 General description

The Communication Hold supplementary service enables a user to suspend the reception of media stream(s) of an established IP multimedia session, and resume the media stream(s) at a later time.

4.3 Operational requirements

4.3.1 Provision/withdrawal

The HOLD service that includes announcements shall be provided after prior arrangement with the service provider.

4.3.2 Requirements on the originating network side

No specific requirements are needed in the network.

4.3.3 Requirements in the network

No specific requirements are needed in the network.

4.3.4 Requirements on the terminating network side

No specific requirements are needed in the network.

4.4 Coding requirements

No specific coding requirements are needed.

4.5 Signalling requirements

4.5.1 Activation/deactivation

The HOLD service is activated at provisioning and deactivated at withdrawal.

4.5.1A Registration/erasure

The HOLD service requires no registration. Erasure is not applicable.

4.5.1B Interrogation

Interrogation of HOLD is not applicable.

4.5.2 Invocation and operation

4.5.2.1 Actions at the invoking UE

In addition to the application of procedures according to 3GPP TS 24.229 [1], the following procedures shall be applied at the invoking UE in accordance with RFC 3264 [4].

A UE shall not invoke the HOLD service on a dialog associated with an emergency call the UE has initiated.

If not all the media streams are affected, the invoking UE shall generate a new SDP offer where:

1) for each media stream that is to be held, the SDP offer contains:
   - an "inactive" SDP attribute if the stream was previously set to "recvonly"; or
   - a "sendonly" SDP attribute if the stream was previously set to "sendrecv";

NOTE 1: If the directionality attribute of the media stream is currently "sendonly" or "inactive", then that media stream is not put on hold and, in the SDP offer, the directionality for that media stream remains unchanged.

2) for each held media stream that is to be resumed, the SDP offer contains:
   - a "recvonly" SDP attribute if the stream was previously an inactive media stream; or
   - a "sendrecv" SDP attribute if the stream was previously a sendonly media stream, or the attribute may be omitted, since sendrecv is the default; and
3) for each media stream that is unaffected, the media parameters in the SDP offer remain unchanged from the previous SDP.

If all the media streams are to be held:

- if they all have identical directionality, the invoking UE shall generate an SDP offer containing a session level direction attribute, or separate media level direction attributes, in the SDP that is set to:
  1) "inactive" if the streams were previously set to "recvonly"; or
  2) "sendonly" if the streams were previously set to "sendrecv"; and

NOTE 2: If the directionality attribute of all the media streams is currently "sendonly" or "inactive", then all these media streams are not put on hold and, in the SDP offer, the directionality for these media streams will remain unchanged.

- if they all do not have identical directionality, then for each media stream in the session, the invoking UE shall follow the procedure listed above for individual media streams.

If all the media streams were previously on hold and are to be resumed:

- if they all have identical directionality, the invoking UE shall generate a session level direction attribute, or separate media level direction attributes, in the SDP that is set to:
  1) "recvonly" if the streams were previously inactive media streams; or
  2) "sendrecv" if the streams were previously sendonly media streams, or the attribute may be omitted, since sendrecv is the default; and
- if they all do not have identical directionality, then for each media stream in the session, the invoking UE shall follow the procedure listed above for individual media streams.

If, in the generated SDP offer, there is at least one media stream whose directionality has changed from the previous SDP, the UE shall send the generated SDP offer in a re-INVITE request (or UPDATE request) to the remote UE.

4.5.2.2 Void

4.5.2.3 Void

4.5.2.4 Actions at the AS of the invoking UE

As a network option, for each media stream marked "recvonly" in the SDP answer sent to the invoking UE, the AS of the invoking UE shall lower the bandwidth by setting the "b=AS:" parameter to a small value, e.g. "b=AS:0". The "b=RR:" and "b=RS:" parameters shall be set to values large enough to enable continuation of the RTCP flow, e.g. "b=RR:800" and "b=RS:800".

As a network option, for each media stream marked "inactive" in the SDP answer sent to the invoking UE, the AS of the invoking UE shall lower the bandwidth by setting the "b=AS:" parameter to a small value, e.g. "b=AS:0". The "b=RR:" and "b=RS:" parameters shall be set to values large enough to enable continuation of the RTCP flow, e.g. "b=RR:800" and "b=RS:800".

NOTE 1: A media stream in the SDP answer can be marked as "recvonly" at media description level or/and at session description level (i.e. all the media streams are marked "recvonly").

As a network option, the AS of the invoking UE shall initiate the procedures for the provision of an announcement to the held user in accordance with 3GPP TS 24.628 [6].

NOTE 2: Not providing an announcement allows the service provider to provide the possibility to change from bidirectional to unidirectional media streams.

NOTE 3: A service provider can decide to provide announcements only if a pre-defined set of media streams are put on HOLD.

The AS shall based on local policy on how to handle PSAP callbacks reject any HOLD invocation request from the served UE by sending a 403 (Forbidden) response.

The mechanism to identify an INVITE request as a PSAP callback depends on local policy and can be based on the PSAP callback indicator specified in IETF RFC 7090 [8].

4.5.2.5 Void

4.5.2.6 Void

4.5.2.7 Void

4.5.2.8 Void

4.5.2.9 Actions at the held UE

3GPP TS 24.229 [1] shall apply.

4.6 Interaction with other services

4.6.1 Communication Hold (HOLD)

Not applicable.

4.6.2 Terminating Identification Presentation (TIP)

No impact, i.e. neither service shall affect the operation of the other service.

4.6.3 Terminating Identification Restriction (TIR)

No impact, i.e. neither service shall affect the operation of the other service.

4.6.4 Originating Identification Presentation (OIP)

No impact, i.e. neither service shall affect the operation of the other service.

4.6.5 Originating identification restriction (OIR)

No impact, i.e. neither service shall affect the operation of the other service.

4.6.6 Conference calling (CONF)

If a participant of a conference invokes the HOLD service, it is not desirable to provide an announcement to the conference. If the AS supporting the HOLD supplementary service receives a re-INVITE (or UPDATE) request on a dialog for which the "isfocus" feature parameter was included in the Contact header from the remote end-point, the AS shall not initiate the procedures for the provision of an announcement to the held user(s).

4.6.7 Communication DIVersion services (CDIV)

No impact, i.e. neither service shall affect the operation of the other service.

4.6.8 Malicious Communication IDentification (MCID)

No impact, i.e. neither service shall affect the operation of the other service.

4.6.9 Anonymous Communication Rejection and Communication Barring (ACR/CB)

No impact, i.e. neither service shall affect the operation of the other service.

4.6.10 Explicit Communication Transfer (ECT)

No impact, i.e. neither service shall affect the operation of the other service.

4.7 Interactions with other networks

4.7.1 Void

4.7.2 Void

4.7.3 Void

4.8 Parameter values (timers)

Not applicable.

Annex A (informative): Signalling Flows

A.1 HOLD communication

Assumption is that a session has been established between UE-A and UE-B using basic communication procedures according to 3GPP TS 24.229 [1], therefore the following signalling flows do not apply to the initial INVITE.

A.1.1 HOLD communication without announcement

The following diagram shows a communication session put on hold using a re-INVITE request . The same can be achieved by sending an UPDATE request.

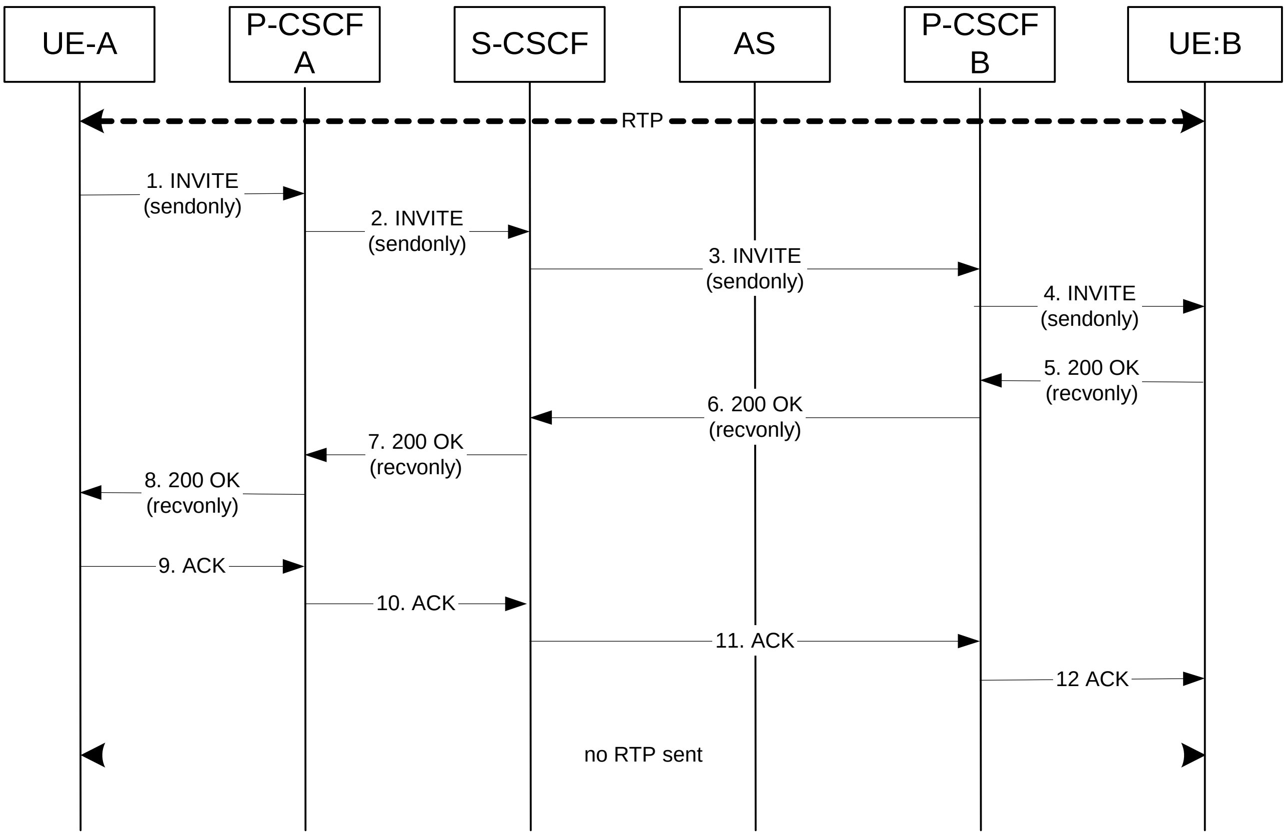

Figure A.1.1.1: HOLD communication without announcement to the held user

1. UE-A sends a re- INVITE to UE-B to hold the session - see example in table A.1.1.1-1. Hold is done by changing the SDP attribute. For each media stream that shall be held:

   - "a=sendonly", if the stream was previously a sendrecv media stream;

   - "a=inactive", if the stream was previously a recvonly media stream.

Table A.1.1.1-1: re-INVITE request (UE to P-CSCF)

```
INVITE user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-ad76cc7fc74
   ;comp=sigcomp SIP/2.0
```

```
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+1-212-555-2222>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Require: sec-agree
Proxy-Require: sec-agree
Supported: precondition, 100rel, gruu, 199
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321;
   port-c=8642; port-s=7531
Contact: <sip:user1_public1@home1.net; gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6
   ;comp=sigcomp>;+g.3gpp.icsi-ref="urn%3Aurn-7%3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, SUBSCRIBE, NOTIFY
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=video 3400 RTP/AVPF 98 99
b=AS:75
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos optional remote sendrecv
a=sendonly
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
m=audio 3456 RTP/AVPF 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos optional remote sendonly
a=sendonly
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

A.1.2 HOLD communication with announcement

The following diagram shows a communication session put on hold using a re-INVITE request with an announcement being played by the AS to the held party. The same can be achieved by sending an UPDATE request.

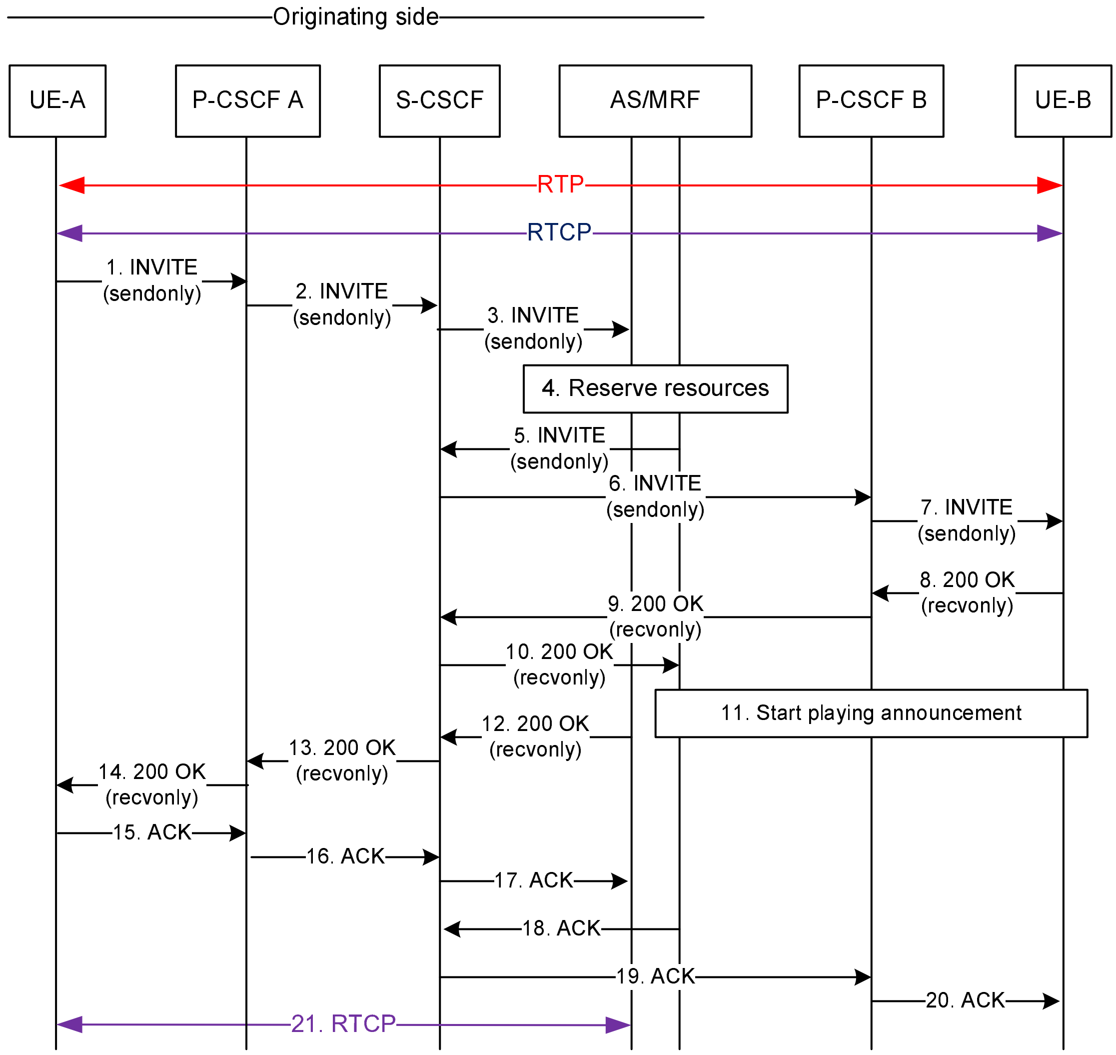

Figure A.1.2.1: HOLD communication with announcement to the held user

1. UE-A sends a SIP INVITE request to UE-B to hold the session - see example in table A.1.2.1-1. Hold is done by changing the SDP attribute. For each media stream that shall be held:

   - "a=sendonly", if the stream was previously a sendrecv media stream;

Table A.1.2.1-1: re-INVITE request (UE to P-CSCF)

```
INVITE user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-ad76cc7fc74
   ;comp=sigcomp SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+1-212-555-2222>
```

```
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Require: sec-agree
Proxy-Require: sec-agree
Supported: precondition, 100rel, gruu, 199
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321;
   port-c=8642; port-s=7531
Contact: <sip:user1_public1@home1.net; gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6
   ;comp=sigcomp>;+g.3gpp.icsi-ref="urn%3Aurn-7%3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, SUBSCRIBE, NOTIFY
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=video 3400 RTP/AVPF 98 99
b=AS:75
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos optional remote sendrecv
a=sendonly
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
m=audio 3456 RTP/AVPF 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos optional remote sendrecv
a=sendonly
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

2. P-CSCF A forwards the SIP INVITE request towards S-CSCF.

3. S-CSCF forwards the SIP INVITE request towards the AS/MRF.

4. AS/MRF decides to configure an announcement towards UE-B. It acts as a B2B UA, inserts MRFP in the media path and reserves resources. Each UA is represented by a separate vertical line in the figure.

5. AS/MRF sends a SIP INVITE request towards UE-B.

6. S-CSCF forwards the SIP INVITE request towards UE-B.

7. P-CSCF B forwards the SIP INVITE request to UE-B.

8. UE-B sends SIP 200 (OK) response towards AS/MRF.

9. P-CSCF forwards the SIP 200 (OK) response towards AS/MRF.

10. S-CSCF forwards the SIP 200 (OK) response towards AS/MRF.

11. AS/MRF starts playing announcement to UE-B, following the procedures in 3GPP TS 24.628 [6].

12. AS/MRF sends a SIP 200 (OK) response towards UE-A.

13. S-CSCF forwards the SIP 200 (OK) response towards UE-A.

14. P-CSCF A forwards the SIP 200 (OK) response towards UE-A.

15. UE-A sends a SIP ACK request towards AS/MRF.

16. P-CSCF forwards the SIP ACK request towards AS/MRF.

17. S-CSCF forwards the SIP ACK request towards AS/MRF.

18. AS/MRF sends a SIP ACK request towards UE-B.

19. S-CSCF forwards the SIP ACK request towards UE-B.

20. P-CSCF B forwards the SIP ACK request towards UE-B.

21. RTCP packets are exchanged between UE-A and AS/MRF. No RTP packets are sent/received by UE-A.

A.1.3 HOLD communication with modification of the SDP answer

The following diagram shows a communication session put on hold using a re-INVITE request with an announcement being played by the AS to the held party. The same can be achieved by sending an UPDATE request.

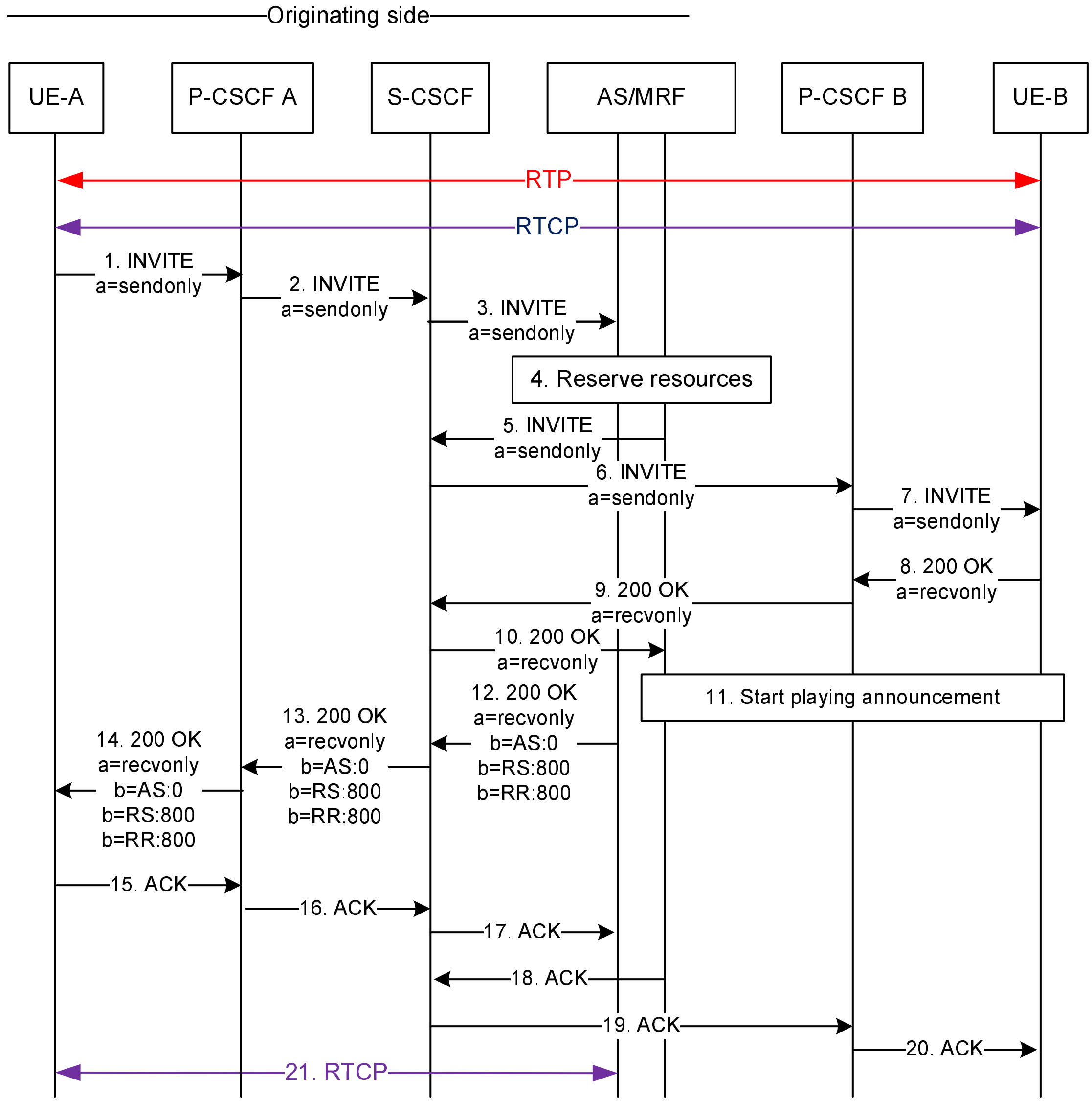

Figure A.1.3.1: HOLD communication with modification of the SDP answer

1. UE-A sends a SIP INVITE request to UE-B to hold the session by changing the direction attribute to "a=sendonly".

Table A.1.3-1: re-INVITE request (UE to P-CSCF)

```
INVITE user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-ad76cc7fc74
   ;comp=sigcomp SIP/2.0
```

```
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+1-212-555-2222>; tag=24615

Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Require: sec-agree
Proxy-Require: sec-agree
Supported: precondition, 100rel, gruu, 199
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321;
   port-c=8642; port-s=7531
Contact: <sip:user1_public1@home1.net; gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6
   ;comp=sigcomp>;+g.3gpp.icsi-ref="urn%3Aurn-7%3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, SUBSCRIBE, NOTIFY
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=video 3400 RTP/AVPF 98 99
b=AS:75
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos optional remote sendrecv
a=sendonly
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
m=audio 3456 RTP/AVPF 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos optional remote sendrecv
a=sendonly
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

2. P-CSCF A forwards the SIP INVITE request towards S-CSCF.

3. S-CSCF forwards the SIP INVITE request towards the AS/MRF.

4. AS/MRF decides to configure an announcement towards UE-B. It acts as a B2B UA, inserts MRFP in the media path and reserves resources. Each UA is represented by a separate vertical line in the figure.

5. AS/MRF sends a SIP INVITE request towards UE-B.

6. S-CSCF forwards the SIP INVITE request towards UE-B.

7. P-CSCF B forwards the SIP INVITE request to UE-B.

8. UE-B sends a SIP 200 (OK) response towards AS/MRF.

9. P-CSCF forwards the SIP 200 (OK) response towards AS/MRF.

10. S-CSCF forwards the SIP 200 (OK) response towards AS/MRF.

11. AS/MRF starts playing announcement to UE-B, following the procedures in 3GPP TS 24.628 [6].

12. The AS/MRF modifies the bandwidth attribute in the SIP 200 (OK) response.

Table A.1.3-2: 200 (OK) response (AS to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net;branch=z9hG4bk120f34.1
Via: SIP/2.0/UDP 1.2.3.4:1357;branch=z9hG4bKnashds7
From: <sip:user1_public1@home1.net>;tag=171828
To: <tel:+1-212-555-2222>; tag=24615
Contact: <sip:user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-
   ad76cc7fc74>;+g.3gpp.icsi-ref="urn%3Aurn-7%3gpp-service.ims.icsi.mmtel"
Call-ID: cb03a0s09a2sdfglkj490333
CSeq: 127 INVITE
Content-Length: (…)

v=0
o=- 2987933817 2987933817 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=video 3400 RTP/AVPF 98 99
b=AS:0
b=RS:800
b=RR:2400
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=recvonly
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
m=audio 3456 RTP/AVPF 97 96
b=AS:0
b=RS:800
b=RR:800
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=recvonly
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

13. S-CSCF forwards the SIP 200 (OK) response towards UE-A.

14. P-CSCF A forwards the SIP 200 (OK) response towards UE-A.

15. UE-A sends a SIP ACK request towards AS/MRF.

16. P-CSCF forwards the SIP ACK request towards AS/MRF.

17. S-CSCF forwards the SIP ACK request towards AS/MRF.

18. AS/MRF sends a SIP ACK request towards UE-B.

19. S-CSCF forwards the SIP ACK request towards UE-B.

20. P-CSCF B forwards the SIP ACK request towards UE-B.

21. RTCP packets are exchanged between UE-A and AS/MRF. No RTP packets are sent/received by UE-A.

A.2 RESUME Communication

A.2.1 RESUME communication without announcement

The following diagram shows how a communication session is resumed using a re-INVITE request; The same can be achieved by sending an UPDATE request.

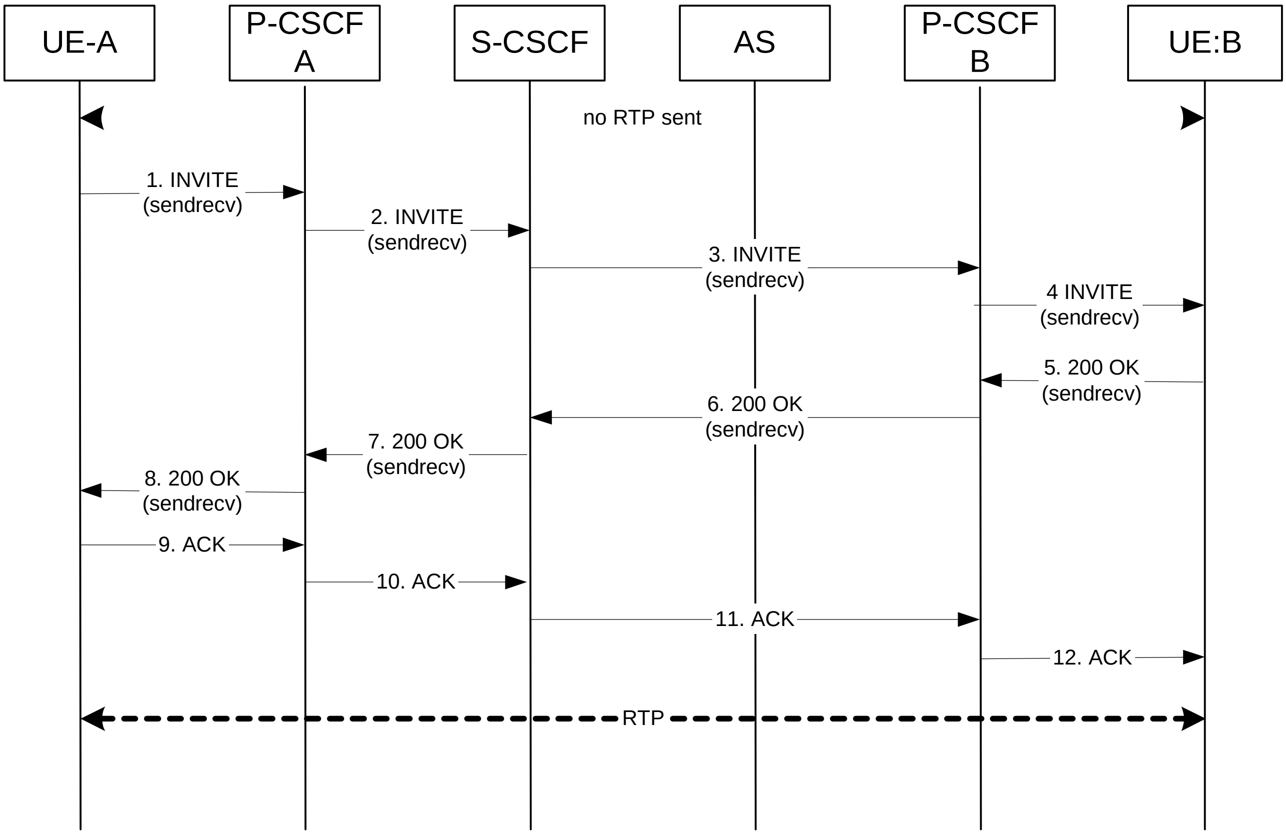

Figure A.2.1.1: RESUME communication without announcement to the held user

1. UE-A sends an INVITE to UE-B to resume the session - see example in table A.2.1.1-1. Resume is done by changing the SDP attribute. For each media stream that shall be resumed:

   - "a=sendrecv", if the stream was previously a sendonly media stream, or the attribute can be omitted, since sendrecv is the default;

   - "a=recvonly", if the stream was previously an inactive media stream.

Table A.2.1.1-1: re-INVITE request (UE to P-CSCF)

```
INVITE user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-ad76cc7fc74
   ;comp=sigcomp SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+1-212-555-2222>
```

```
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Require: sec-agree
Proxy-Require: sec-agree
Supported: precondition, 100rel, gruu, 199
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321;
   port-c=8642; port-s=7531
Contact: <sip:user1_public1@home1.net; gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6
   ;comp=sigcomp>;+g.3gpp.icsi-ref="urn%3Aurn-7%3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, SUBSCRIBE, NOTIFY
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=video 3400 RTP/AVPF 98 99
b=AS:75
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos optional remote sendrecv
a=sendrecv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
m=audio 3456 RTP/AVPF 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos optional remote sendrecv
a=sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

A.2.2 RESUME communication with announcement

The following diagram shows how a communication session is resumed using a re-INVITE request after it was held with an announcement being played by the AS to the held party. The same can be achieved by sending an UPDATE request.

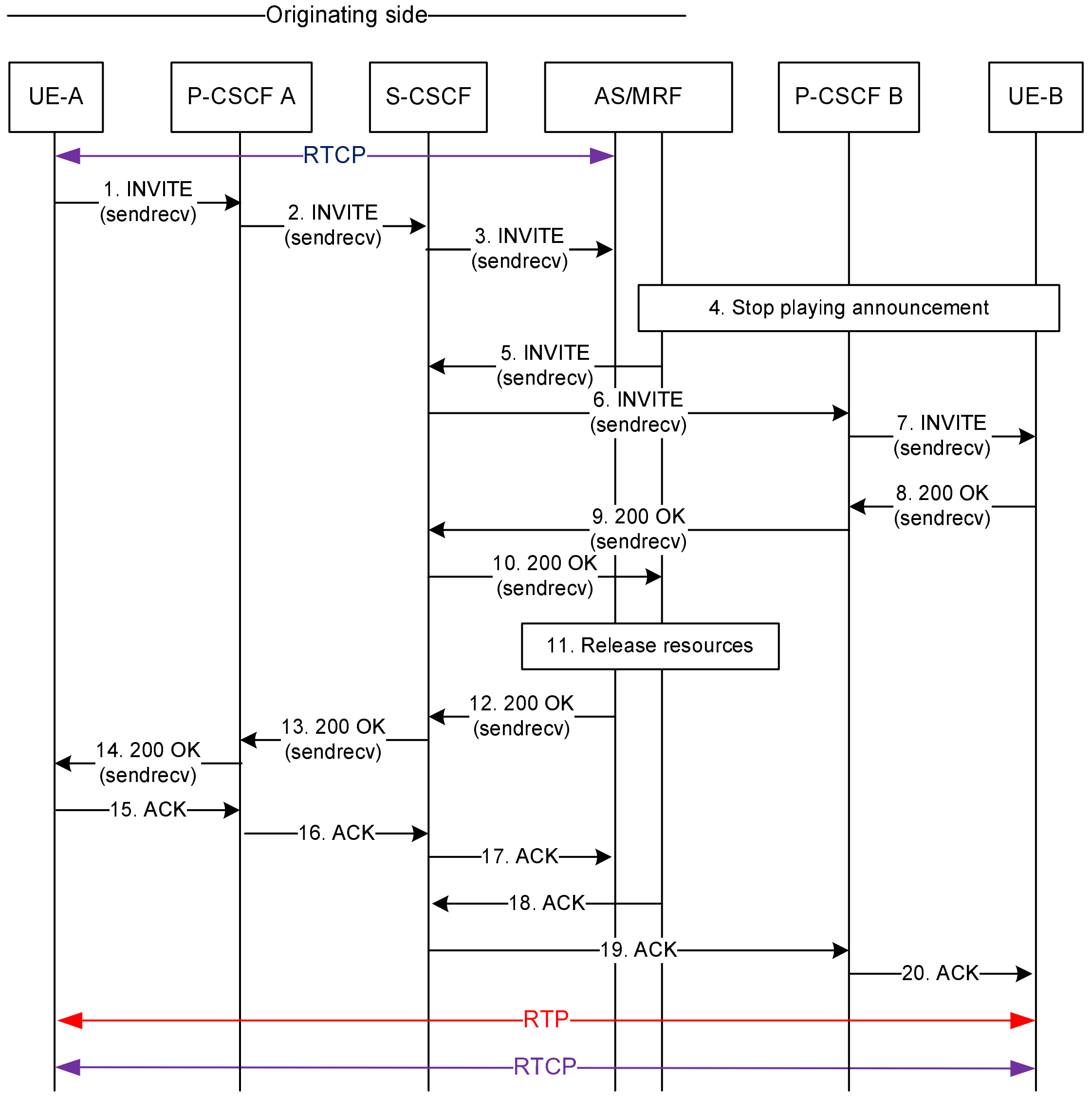

Figure A.2.2.1: RESUME communication with announcement to the held user

1. UE-A sends a SIP INVITE request to UE-B to resume the session - see example in table A.2.2.1-1. Resume is done by changing the SDP attribute. For each media stream that shall be resumed:

   - "a=sendrecv", if the stream was previously a sendonly media stream, or the attribute can be omitted, since sendrecv is the default;

Table A.2.2.1-1: re-INVITE request (UE to P-CSCF)

```
INVITE user2_public1@home2.net;gr=urn:uuid:2ad8950e-48a5-4a74-8d99-ad76cc7fc74
   ;comp=sigcomp SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>, <sip:orig@scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <sip:user1_public1@home1.net>; tag=171828
To: <tel:+1-212-555-2222>
```

```
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Require: sec-agree
Proxy-Require: sec-agree
Supported: precondition, 100rel, gruu, 199
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321;
   port-c=8642; port-s=7531
Contact: <sip:user1_public1@home1.net; gr=urn:uuid:f81d4fae-7dec-11d0-a765-00a0c91e6bf6
   ;comp=sigcomp>;+g.3gpp.icsi-ref="urn%3Aurn-7%3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, SUBSCRIBE, NOTIFY
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=video 3400 RTP/AVPF 98 99
b=AS:75
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos optional remote sendrecv
a=sendrecv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
m=audio 3456 RTP/AVPF 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos optional remote sendrecv
a=sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

2. P-CSCF A forwards the SIP INVITE request towards S-CSCF.

3. S-CSCF forwards the SIP INVITE request towards the AS/MRF.

4. AS/MRF stops playing announcement to UE-B, following the procedures in 3GPP TS 24.628 [6].

5. AS/MRF forwards the SIP INVITE request towards UE-B, resulting in the removal of MRFP from the media path.

6. S-CSCF forwards the SIP INVITE request towards UE-B.

7. P-CSCF B forwards the SIP INVITE request to UE-B.

8. UE-B sends a SIP 200 (OK) response towards AS/MRF.

9. P-CSCF forwards the SIP 200 (OK) response towards AS/MRF.

10. S-CSCF forwards the SIP 200 (OK) response towards AS/MRF.

11. AS/MRF releases the resources allocated previously for the announcemnt

12. AS/MRF forwards the SIP 200 (OK) response towards UE-A.

13. S-CSCF forwards the SIP 200 (OK) response towards UE-A.

14. P-CSCF A forwards the SIP 200 (OK) response towards UE-A.

15. UE-A sends SIP ACK request towards AS/MRF.

16. P-CSCF forwards the SIP ACK request towards AS/MRF.

17. S-CSCF forwards SIP ACK request towards AS/MRF.

18. AS/MRF forwards the SIP ACK request towards UE-B.

19. S-CSCF forwards the SIP ACK request towards UE-B.

20. P-CSCF B forwards the SIP ACK request towards UE-B.

Annex B (informative): Example of filter criteria

An example of an IFC Trigger Point configuration where the S-CSCF invokes the HOLD AS:

- Method="INVITE".

An example of an IFC Trigger Point configuration where the S-CSCF does not invoke the HOLD AS for a PSAP callback:

- Method="INVITE" and not Priority header field with a "psap-callback" header field value.

NOTE: Not invoking the HOLD AS assumes that the HOLD invocation request can be handled elsewhere in the network, e.g. in the PSAP itself.

Annex C (informative): Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
2005-08					Publication as **ETSI TS 183 010**		1.1.1
2006-04					Publication as ETSI TS 183 010		1.2.1
2007-04					Publication as ETSI TS 183 010		1.2.2
2007-12					Conversion to **3GPP TS 24.410**		1.2.3
2008-01					Technically identical copy as **3GPP TS 24.610** as basis for further development.		1.2.4
2008-02					Implemented C1-080101		1.3.0
2008-04					Implemented C1-080886, C1-080887, C1-081090, C1-081091, C1-081113		1.4.0
2008-05					Implemented C1-081831, C1-081913		1.5.0
2008-05					Editorial changes done by MCC	1.5.0	1.5.1
2008-06	CT#40	CP-080330			CP-080330 was approved by CT#40 and version 8.0.0 is created by MCC for publishing	1.5.1	8.0.0
2008-09	CT#41	CP-080533	0001		Correction of Activation/deactivation of HOLD	8.0.0	8.1.0
2008-09	CT#41	CP-080533	0002	1	Miscellaneous clean-up corrections	8.0.0	8.1.0
2008-09	CT#41	CP-080533	0003		Applicability statement in scope	8.0.0	8.1.0
2008-09	CT#41	CP-080533	0004		Interaction of HOLD and CONF	8.0.0	8.1.0
2008-12	CT#42	CP-080865	0005	1	Holding or resuming all media streams	8.1.0	8.2.0
2008-12	CT#42	CP-080865	0006	2	Contents of SDP offer in HOLD	8.1.0	8.2.0
2008-12	CT#42	CP-080865	0007	1	Fixed the flows	8.1.0	8.2.0
2009-03	CT#43	CP-090121	0008		Correction of URN-value for Service Identifiers	8.2.0	8.3.0
2009-12	CT#46	CP-090923	0009	1	Correction of icsi-ref feature tag	8.3.0	9.0.0
2010-09	CT#49	CP-100526	0014		HOLD corrections	9.0.0	9.1.0
2011-03	CT#51				Upgrade to Rel-10	9.1.0	10.0.0
2012-03	CT#55	CP-120124	0015	1	HOLD-CONF interaction	10.0.0	11.0.0
2012-03	CT#55	CP-120197	0016	2	Clarification of HOLD procedures	10.0.0	11.0.0
2012-06	CT#56	CP-120307	0017	1	Usage of SDP direction attributes correction	11.0.0	11.1.0
2012-12	CT#58	CP-120778	0021	2	Emergency call HOLD suppression	11.1.0	11.2.0
2013-06	CT#60	CP-130414	0022	8	Prevent HOLD for PSAP callback	11.2.0	12.0.0
2013-09	CT#61	CP-130485	0027	2	Call Hold Bandwidth Management	12.0.0	12.1.0
2013-09	CT#61	CP-130507	0028		draft-ietf-ecrit-psap-callback reference update	12.0.0	12.1.0
2013-12	CT#62	CP-130758	0029	2	Reference update: draft-ietf-ecrit-psap-callback	12.1.0	12.2.0
2013-12	CT#62	CP-130763	0030	1	RTCP flow during hold with announcement	12.1.0	12.2.0
2014-03	CT#63	CP-140143	0032		Correction the direction attribute of media stream(s) in the Communication Hold supplementary service	12.2.0	12.3.0
2014-06	CT#64	CP-140330	0034		Clean-up of HOLD flows	12.3.0	12.4.0
2014-06	CT#64	CP-140330	0036		Minor corrections 24.610	12.3.0	12.4.0
2014-09	CT#65	CP-140665	0033	4	Media directionality when resuming a session established with one-way media	12.4.0	12.5.0
2014-12	CT#66	CP-140833	0037	1	Reference update: RFC 7090 (draft-ietf-ecrit-psap-callback)	12.5.0	12.6.0

History

Document history		
V12.5.0	October 2014	Publication
V12.6.0	January 2015	Publication