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TECHNICAL SPECIFICATION

**Digital cellular telecommunications system (Phase 2+) (GSM);
Universal Mobile Telecommunications System (UMTS);
LTE;
Conference (CONF) using IP Multimedia (IM)
Core Network (CN) subsystem;
Protocol specification
(3GPP TS 24.605 version 15.0.0 Release 15)**

A GLOBAL INITIATIVE

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Contents

Foreword

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1 Scope

The present document specifies the stage three Protocol Description of the Conference (CONF) service based on stage one and two of the ISDN CONF supplementary service. 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 specifies centralized conferencing, using a conference focus, distributed conferencing is out of scope.

The present document does not cover the cases of :

a) cascading conference services; and

b) the support of the PSTN/ISDN conference service hosted in the PSTN.

The present document is applicable to User Equipment (UE) and Application Servers (AS) which are intended to support the CONF 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 22.173: "IP Multimedia Core Network Subsystem (IMS) Multimedia Telephony Service and supplementary services; Stage 1".

[2] 3GPP TS 24.229: "Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3".

[3] Void

[4] Void.

[5] Void.

[6] Void.

[7] 3GPP TS 24.147: "Conferencing using the IP Multimedia (IM) Core Network (CN) subsystem; Stage 3".

[8] IETF RFC 3891: "The SIP Replaces header".

[9] Void.

[10] ETSI TS 183 043 V3.4.1 (2011-04):"Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IMS - based PSTN/ISDN Emulation; Stage 3 specification"

[11] 3GPP TS 24.628: "Common Basic Communication procedures using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification".

[12] 3GPP TS 24.610: "Communication HOLD (HOLD) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification".

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

[14] 3GPP TS 24.166: "3GPP IMS Conferencing Management Object (MO)".

[15] 3GPP TS 23.003: "Numbering, addressing and identification".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in 3GPP TS 22.173 [1] and 3GPP TS 24.147 [7] apply.

3.2 Abbreviations

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

ACR/CB Anonymous Communication Rejection and Communication Barring
AS Application Server
CDIV Communication DIVersion
CONF CONFerence calling
CS Circuit Switch
ECT Explicit Communication Transfer
HOLD communication HOLD
IMS IP Multimedia Subsystem
IP Internet Protocol
ISDN Integrated Service Data Network
MCID Malicious Communication IDentification
MGCF Media Gateway Control Function
OIP Originating Identification Presentation
OIR Originating Identification Restriction
PSAP Public Safety Answering Point
P-CSCF Proxy CSCF
PSTN Public Switched Telephone Network
SIP Session Initiation Protocol
TIP Terminating Identification Presentation
TIR Terminating Identification Restriction
UE User Equipment

4 CONFerence (CONF)

4.1 Introduction

The CONFerence (CONF) service enables a user to participate in and control a simultaneous communication involving a number of users.

4.2 Description

4.2.1 General description

When the CONF service is invoked, conference resources are allocated to the served user.

Once a conference is active, users can join and leave a conference, and remote users can be added to or removed from the conference.

Conference participants can request to be informed of these actions.

The Management Object as defined in 3GPP TS 24.166 [14] provides a mechanism for the UE to discover the Conference Factory URI to be used for the CONF service. If the UE is not configured with the Conference Factory URI within the Management Object then the UE shall derive a Conference Factory URI for MMTEL as described in subclause 13.10 of 3GPP TS 23.003 [15] to be used for the CONF service.

NOTE: Depending on the network operator, various mechanisms can be applied to discover the Conference Factory URI (e.g., include the Conference Factory URI in a letter written to the user or publish it on a website). A derived Conference Factory URI might not be a valid URI.

4.3 Operational requirements

4.3.1 Provision/withdrawal

The CONF service 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

For coding requirements see 3GPP TS 24.147 [7], clause 5.

4.5 Signalling requirements

4.5.1 Activation/deactivation

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

4.5.1a Registration/erasure

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

4.5.1b Interrogation

Interrogation of CONF is not applicable.

4.5.2 Invocation and operation

This subclause describes the usage of and the changes to the procedures of 3GPP TS 24.147 [7] for invoking and operating a conference.

4.5.2.1 Actions at the originating UE

4.5.2.1.1 User joining a conference

Procedures according to 3GPP TS 24.147 [7], subclause 5.3.1.4 shall apply.

4.5.2.1.2 User inviting another user to a conference

Procedures according to 3GPP TS 24.147 [7], subclause 5.3.1.5 shall apply with the following additions to subclause 5.3.1.5.3 of 3GPP TS 24.147 [7]:

- A UE that has initiated an emergency call, shall not perform any procedures to add the remote user in that call to a conference.

- In order to avoid the establishment of a second communication to the invited user, in case of an active session the UE may additionally include the Replaces header in the header portion of the SIP URI of the Refer-to header of the REFER request. The included Replaces header shall refer to the active dialog that is replaced by the ad-hoc conference. The Replaces header shall comply with RFC 3891 [8].

NOTE 1: In case of an interworking to the PSTN the routing of the INVITE request from the conference focus to the MGCF that handles the Replaces information is not deterministic and the replacement of the active dialog might fail.

EXAMPLE: Refer-To: <sip:mgcf1.home1.net; method=INVITE?Replaces=cb03a0s09a2sdfglkj490333%3Bto-tag%3D 314159%3Bfrom-tag%3D171828&Requrie=replaces >.

NOTE 2: If a conference participant invites another user to a conference by using a REFER request targeted at the other user (following 3GPP TS 24.147 [7] , subclause 5.3.1.5.2), there can be cases where such REFER request is intercepted by an AS serving the requesting user which applies special REFER handling procedures according to 3GPP TS 24.628 [11] subclause 4.7.2.9.7.2. The consequence of this is that the conference focus AS will receive an INVITE from the referrers AS and not from the targeted user. This however does not affect the conference focus procedures in any way.

4.5.2.1.3 User leaving a conference

Procedures according to 3GPP TS 24.147 [7], subclause 5.3.1.6 shall apply.

4.5.2.1.4 User creating a conference

Procedures according to 3GPP TS 24.147 [7], subclause 5.3.1.3 shall apply.

4.5.2.1.5 Subscription for the conference event package

Procedures according to 3GPP TS 24.147 [7], subclause 5.3.1.2 shall apply.

4.5.2.2 Actions at the conferencing AS

4.5.2.2.1 Conference focus

Procedures according to 3GPP TS 24.147 [7] , subclauses 5.3.2 and 6.3.2 shall apply with the following additions to subclause 5.3.2.5.2 of 3GPP TS 24.147 [7]:

- If a Referred-By header is available in the REFER request, the AS shall verify if the provided Referred-By header contains a valid identity of the requesting user. If not, the AS shall replace the Referred-By header with a valid value matching the P-Asserted-Identity header in the REFER request.

If no Referred-By header is available in the request, the AS shall add a Referred-By header that matches the P-Asserted-Identity header in the REFER request.

The procedures described in subclause 5.3.2.5.5 of 3GPP TS 24.147 [7] shall not apply.

4.5.2.2.1A Void

4.5.2.2.2 Conference notification service

In case of the subscription of a conference participant to the conference notification service, procedures according to 3GPP TS 24.147 [7], subclause 5.3.3 shall apply.

4.5.2.2A Procedures at the AS serving the originating user

Upon receiving a REFER request in an existing dialog or outside of an existing dialog containing a Target-Dialog header field identifying an existing dialog, from a user involved also in a conference, the AS serving the originating user shall first check that the REFER request is valid:

- the Request-URI in the REFER request is targeted to the same UE instance (remote UE) that is involved in the dialog; and

- the Refer-To header in the REFER request contains an URI so that the method constructed from the URI is equal to an INVITE request to the Conference focus.

Otherwise, the AS may, depending on operator policy:

- reject the REFER request; or
- handle the REFER request with another service; or
- proxy the REFER request on.

If any of the following is true:

- the dialog on which the REFER request is received or the dialog which is identified by the Target-Dialog header field in the REFER request, was that established by an initial INVITE request that was identified as a PSAP callback request; or
- the Refer-To header field in the REFER request contains a URI with which the referor is involved in a dialog where the initial INVITE request was identified as a PSAP callback request.

the AS shall based on local policy on how to handle PSAP callbacks reject the REFER request.

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 [13].

If the AS determines that the REFER request shall not be sent to the remote UE and the AS decides to apply 3pcc the AS shall follow the special REFER handling procedures in 3GPP TS 24.628 [11] with the following addition:

- the AS shall include the "isfocus" header field parameter in the Contact header field in the INVITE request sent towards the remote UE.

4.5.2.3 Void

4.5.2.4 Void

4.5.2.5 Void

4.5.2.6 Void

4.5.2.7 Actions at the destination UE

Upon receipt of an INVITE request that includes a Replaces header, the UE shall apply the procedures described in RFC 3891 [8] to the INVITE request.

4.5.2.8 Void

4.5.2.9 Void

4.6 Interaction with other services

4.6.1 Communication HOLD (HOLD)

The AS supporting the CONF service shall support the procedures for the held UE as specified in 3GPP TS 24.610 [12]

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)

For the conferencing AS implementing the conference focus, the following applies:

- If a participant is added to the conference and if TIR is active for the terminating party of this session, then the identity information of that participant shall not be included in conference notifications to other participants.

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)

For the conferencing AS implementing the conference focus, the following applies:

- If a participant joins the conference and if OIR is active for the originating party of this session, then the identity information of that participant shall not be included in conference notifications to other participants.
- If a REFER request is received and if the Privacy header field is set to "header" or "user", then for the INVITE request to the refer-to target, the conference AS shall:

  a) not insert the Referred-by header field, if it does not exist in the REFER request; or

  b) remove the Referred-By header field, if the Privacy header field of the REFER request is set to "user".
- If an INVITE request with "recipient-list" body is received, and if the Privacy header field is set to "user", then the conference AS shall anonymize the From header field of resulting reINVITE request, if there is established dialog between the conference controller and the target of the reINVITE request.

4.6.6 CONFerence calling (CONF)

Not applicable.

NOTE: Cascading conference services are out of scope of the present specification.

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)

The focus AS shall not accept REFER requests with a refer-to target that is barred by the conference creators Outgoing Communication Barring (OCB) rules.

The focus AS shall remove the URI that is barred by the conference creator Outgoing Communication Barring (OCB) rules from the list of URIs in the "recipient-list" body of INVITE request.

4.6.10 Explicit Communication Transfer (ECT)

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

4.7 Interworking 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.0 Scope of signalling flows

This annex gives examples of signalling flows related to the Conference (CONF) service.

These signalling flows are simplified in that they do not show the AS to MRFC interactions nor the AS and MRFC functional split.

A.1 CONF interworking signalling flow in case of an active communication between IMS and PSTN

Figure A.1 depictures a flow where two UEs are engaged in a call, and one of the users is located in the PSTN. At some point in time, UE A decides to activate the CONF service and move the call to a centralized conference. UE A creates the conference, and provides instructions to the conference server to contact UE B and replace the initial communication with a communication to the conference server.

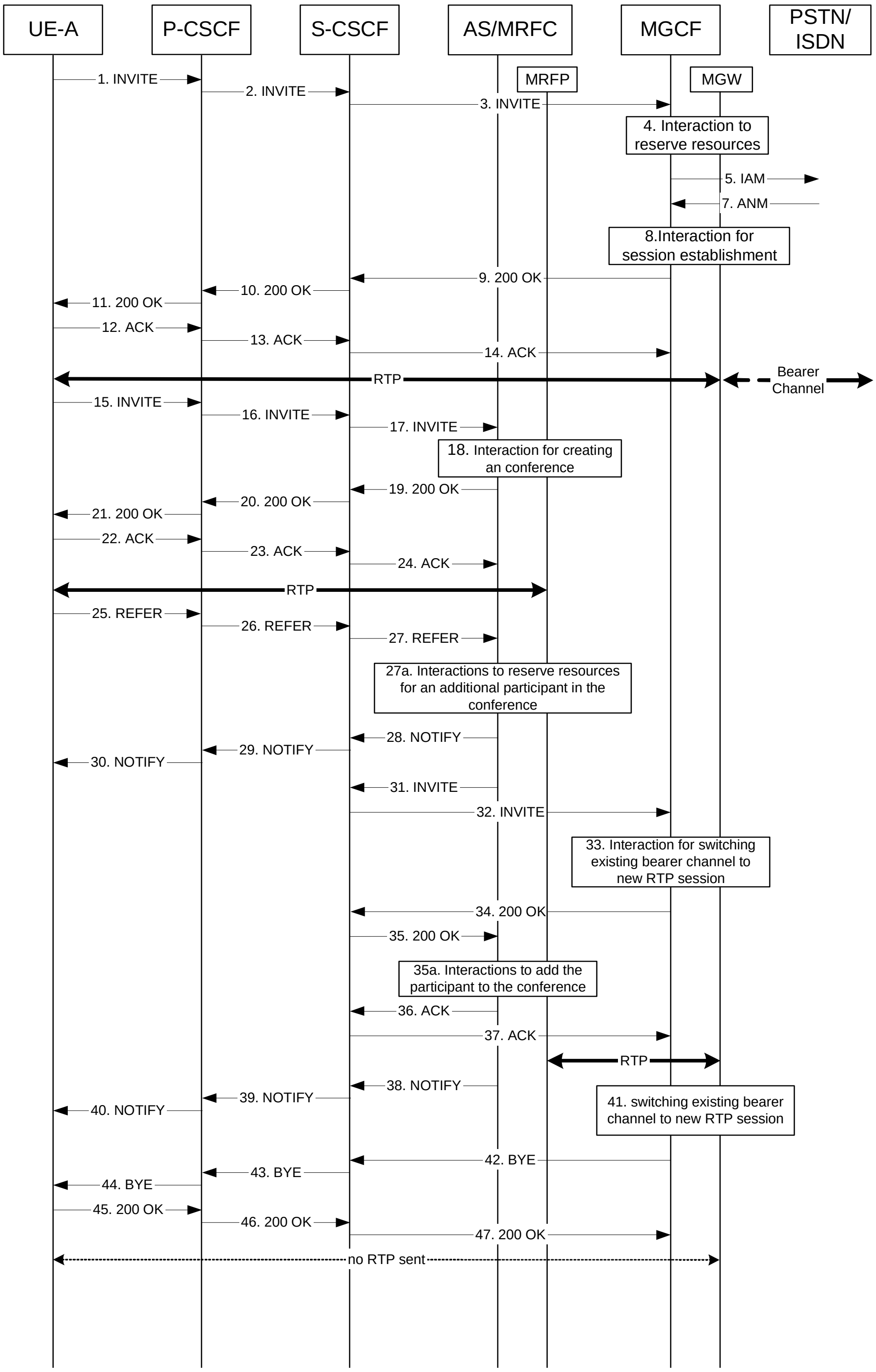

Figure A.1: CONF interworking signalling flow in case of an active communication between IMS and PSTN

- Description figure A.1

NOTE: Only the most relevant messages are shown in figure A.1

UE-A is in an active voice session with a PSTN/ISDN TE (SIP dialog with Call-ID, to-tag and from-tag between UE-A and MGCF). It then creates a conference and invites the PSTN/ISDN TE to the conference by sending a REFER to the conference focus, which invites the PSTN/ISDN TE to the conference by sending an INVITE which includes the Replaces header to the MGCF. The MGCF confirms the session, switches the existing bearer channel to the new RTP session, and terminates the session which is replaced.

1. to 3. UE-A initiates a voice session with a PSTN/ISDN TE by sending an INVITE request to the MGCF.

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

```
INVITE tel:+1-212-555-2222 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: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
Accept:application/sdp,.application/3gpp-ims+xml
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3gpp-service.ims.icsi.mmtel"
P-Preferred-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel
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/AVP 98 99a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=inactive
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
m=audio 3456 RTP/AVP 97 96a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=inactive
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

4: Interaction to reserve resources.

5: SS7: IAM.

7: SS7: ANM.

8: Interaction for session establishment.

9 to 11: The MGCF sends a final response back to the session originator.

Table A.2: 9. 200 OK (MGCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP bgcf1.home1.net;branch=z9hG4bK6546q2.1, SIP/2.0/UDP
scscf1.home1.net;branch=z9hG4bK332b23.1, SIP/2.0/UDP pcscf1.home1.net;branch=z9hG4bK431h23.1,
SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Record-Route: <sip:scscf1.home1.net;lr>, <sip:pcscf1.home1.net;lr>
P-Asserted-Identity: <tel:+1-212-555-2222>
P-Charging-Vector:
Privacy: none
```

```
From:
To: <tel:+1-212-555-2222>;tag=314159
Call-ID:
CSeq:
Require: 100rel, precondition
Contact: <sip:mgcf1.home1.net;gr>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE
RSeq: 9021
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933623 2987933623 IN IP6 5555::eee:fff:aaa:bbb
s=-
c=IN IP6 5555::eee:fff:aaa:bbb
t=0 0
m=video 0 RTP/AVP 98 99
m=audio 6544 RTP/AVPF 97 96a=acfg:1 t=1
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=inactive
a=conf:qos remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

12 to 14: The Calling party acknowledges the final response with an ACK request.

15 to 24: UE-A creates a conference by sending an INVITE request to the Conference Factory URI and connects to the conference.

Table A.3: 15. INVITE request (UE-A to P-CSCF)

```
INVITE sip:conference-factory1@mrfc1.home1.net 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=171829
To: <sip:conference-factory1@mrfc1.home1.net>
Call-ID: cb03a0s09a2sdfglkj490444
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"
Accept:application/sdp,.application/3gpp-ims+xml
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3gpp-service.ims.icsi.mmtel"
P-Preferred-Service: urn:urn-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/AVP 98 99a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=inactive
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
```

```
a=rtpmap:99:MPVMP4V-ES
m=audio 3456 RTP/AVP 97 96a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=inactive
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

25 to 27: UE-A invites the PSTN/ISDN TE to the conference by sending a REFER request to the conference focus, the "method" parameter set to "INVITE". The Refer-To header of the REFER request includes the Replaces parameter with Call-ID, to-tag and from-tag from the existing SIP dialog.

Table A.4: 25. REFER request (UE-A to P-CSCF)

```
REFER sip: conference1@mrfc1.home1.net 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=171829
To: <sip:conference1@mrfc1.home1.net>
Call-ID: cb03a0s09a2sdfglkj490555
Cseq: 127 REFER
Require: sec-agree
Refer-To: <sip:mgcf1.home1.net; method=INVITE?Replaces=cb03a0s09a2sdfglkj490333%3Bto-
tag%3D314159%3Bfrom-tag%3D171828&Requrie=replaces >
Referred-By: <sip:user1_public1@home1.net>
Proxy-Require: sec-agree
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"
Content-Length: 0
```

27a: Interactions to reserve resources for an additional participant in the conference.

28 to 30: The conference focus sends a NOTIFY request containing information about the progress of the REFER request processing. The Subscription-State is set to "active".

31 to 32: The conference focus invites the PSTN/ISDN TE by sending an INVITE request to the MGCF. The INVITE request includes the Replaces header with Call-ID, to-tag and from-tag from the existing SIP dialog.

Table A.5: 31. INVITE request (MRFC/AS to S-CSCF)

```
INVITE sip:mgcf1.home1.net SIP/2.0
Via: SIP/2.0/UDP mrfc1.home1.net;branch=z9hG4bK23273846
Max-Forwards: 70
P-Asserted-Identity: <sip:conference1@mrfc1.home1.net>
P-Charging-Vector: icid-value="AyretyU0dm+6O2IrT5tAFrbHLso=023551024"; orig-ioi=home1.net
Privacy: none
From: <sip:conference1@mrfc1.home1.net>;tag=171123
To: <sip:mgcf1.home1.net>
Call-ID: bc03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Require: replaces
Replaces: cb03a0s09a2sdfglkj490333;to-tag=314159;from-tag=171828
Supported: precondition, 100rel
Referred-By: <sip:user1_public1@home1.net>
Contact: <sip:conference1@mrfc1.home1.net>;isfocus
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE, SUBSCRIBE, NOTIFY
Allow-Events: conference
Content-Type: application/sdp
Content-Length: (…)

v=0
```

```
o=- 2987933615 2987933615 IN IP6 5555::abc:def:abc:abc
s=-
c=IN IP6 5555::abc:def:abc:def
t=0 0
m=video 10001 RTP/AVP 98
a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=inactive
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
m=audio 6544 RTP/AVP 97 96a=tcap:1 RTP/AVPF
a=pcfg:1 t=1
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=inactive
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

33: Interaction for switching existing bearer channel to new RTP.

34 to 35: The MGCF sends a final response back to the session originator.

35a: Interaction to add the participant to the conference.

36 to 37: The Calling party acknowledges the final response with an ACK request.

38 to 40: The conference focus sends a NOTIFY request containing information about the progress of the REFER request processing. The Subscription-State is set to "terminated".

41: The MGCF replaces the existing RTP stream to UE-A with the new RTP stream to the conferencefocus.

42 to 44: The MGCF releases the session with UE-A by sending a BYE request to UE-A.

45 to 47: UE-A responds with a 200 OK response.

A.2 Call flow for 3PTY CONF

A.2.1 Invite other user to 3PTY CONF by sending REFER request

Figure A.2 depictures a flow where two UEs, UE-1 and UE-2, are engaged in a call. At some point in time, UE-1 decides to involve UE-3 into the communication and activate the 3PTY CONF service. UE-1 puts UE-2 on hold, initiates a session toward UE-3 to get the user's permission to start 3PTY call, creates the conference, and moves the original communication with both UE-2 and UE-3 to the conference server.

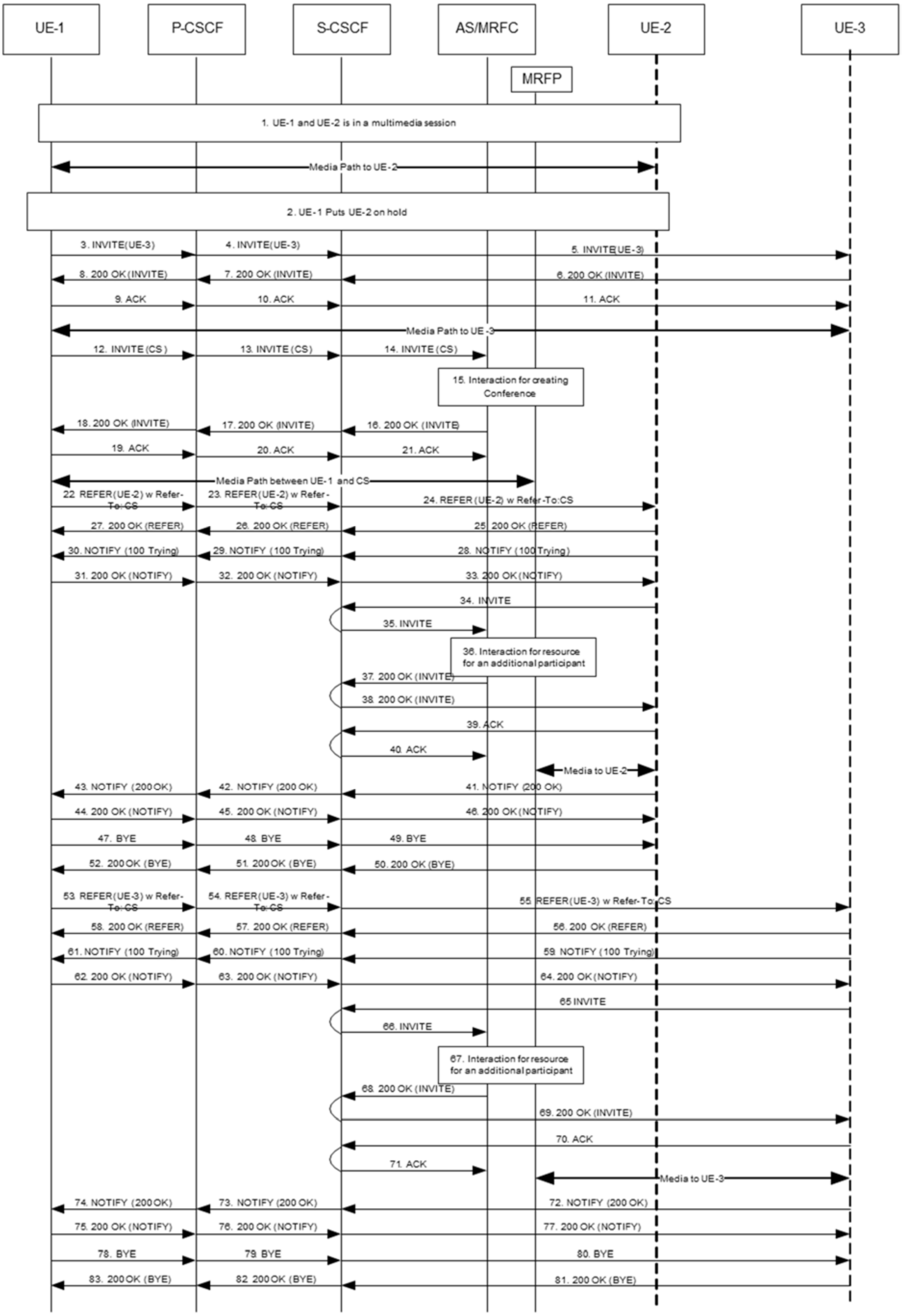

Figure A.2: Call flow for 3PTY conference

UE-1 and UE-2 are in an active call. UE-1 decides to add UE-3 to make it a 3-way conferencing call.

1. UE-1 and UE-2 are in an active call.

2. UE-1 puts UE-2 on hold before invoking the 3-Way Calling with UE-3.

3~11. UE-1 establishes a call with UE-3 following normal call setup procedure and gets UE-3's permission to start the 3-Way Calling.

12~14. UE-1 sends an INVITE request to the conference server to establish a conference session.

15. The CS coordinates with MRFP to allocate conference resources.

16~21. The CS sends a 200 (OK) response and receives an ACK request from UE-1.

22~27. UE-1 sends a REFER request to UE-2 with the Refer-To header set to the address of the CS; UE-2 accepts the REFER request.

28~33. UE-2 sends a NOTIFY request to UE-1 to indicate that UE-2 is acting on the REFER request.

34~35. UE-2 sends an INVITE request to the CS to join the conference.

36. The CS coordinates with MRFP to allocate more resources.

37~40. The CS sends a 200 (OK) response to UE-2 and receives an ACK request.

41~46. UE-2 sends a NOTIFY request to UE-1 to indicate that it has finished action required by the REFER request.

47~52. UE-1 sends a BYE request to terminate the call between itself and UE-2.

53~58. In parallel to step 22~52, UE-1 sends a REFER request to UE-3 with the Refer-To header set to the address of the CS; UE-3 accepts the REFER request.

59~64. UE-3 sends a NOTIFY request to UE-1 to indicate that UE-3 is acting on the REFER request.

65~66. UE-3 sends an INVITE request to the CS to join the conference.

67. The CS coordinates with MRFP to allocate more resources.

68~71. The CS sends a 200 (OK) response to UE-3 and receives an ACK request.

72~77. UE-3 sends a NOTIFY request to UE-1 to indicate that it has finished action required by the REFER request.

78~83. UE-1 sends a BYE request to terminate the call between itself and UE-3.

A.2.2 Invite other user to 3PTY CONF by sending INVITE request with URI list

Figure A.3 depictures a flow where UA-A is involved in 2 communications with UA-B and UA-C, both 2 communications are on-hold. The AS is involved in both 2 communications as a B2BUA.

When user A intends to start the 3PTY conference, UA-A sends an INVITE request to the AS to create the conference and indicates that certain dialogs will be re-used for this conference, The AS sends re-INVITEs in the indicated dialogs and connects the media to the conference bridge. The dialogs can be indicated by adding the Call-ID header field, the From header field and the To header field to the entries in the URI list of the initial INVITE request.

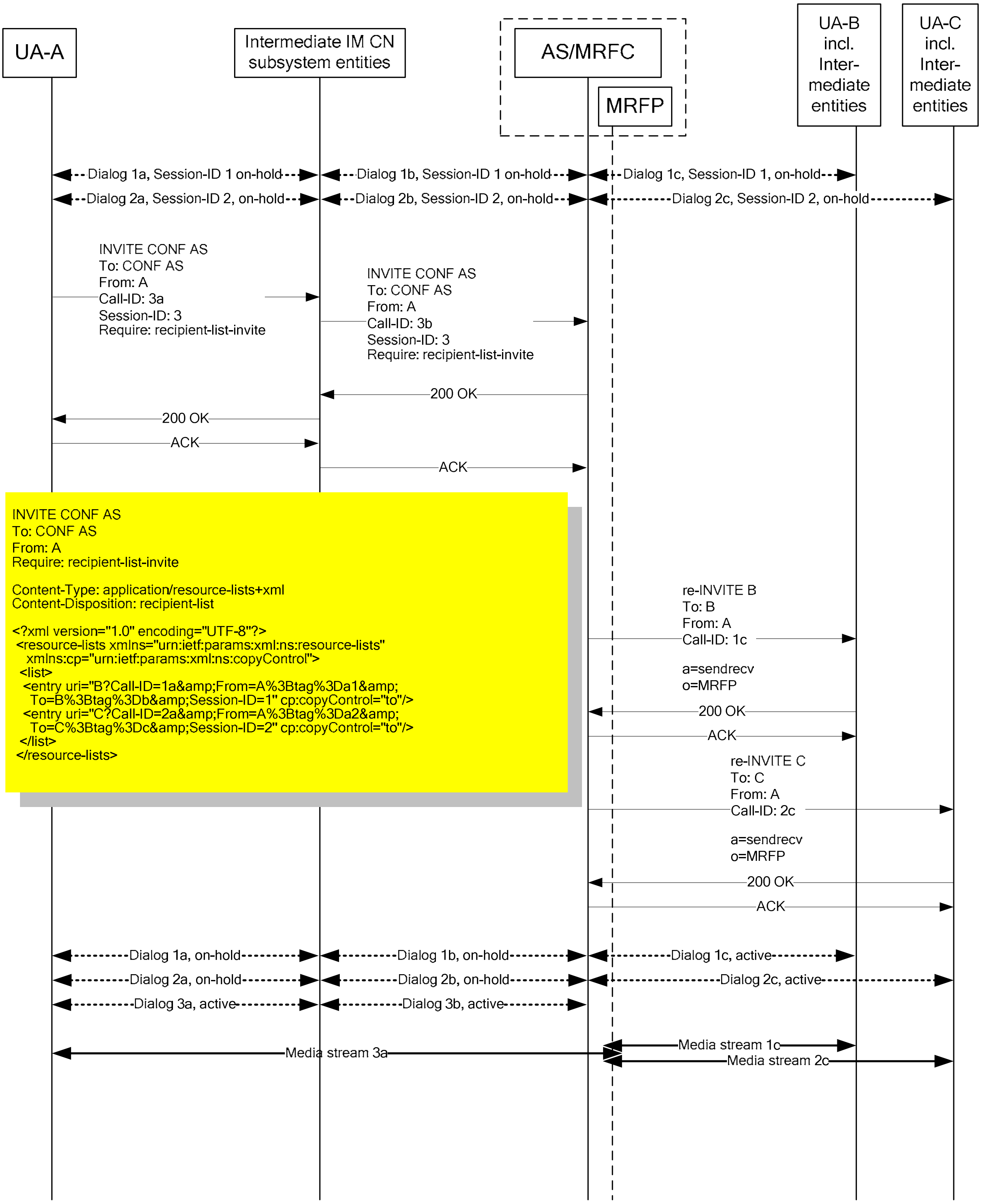

Figure A.3: Call flow for 3PTY conference

1: Both dialogs are in the held condition. This assumes that the AS has the correct interactions between the services call HOLD and CONF i.e. when the conference is finished the origin dialogs 1c and 2c with Session-ID 1 and Session-ID 2 will be resumed. This is needed to allow resuming the held dialogs.

2: UA-A creates a conference and invites user B and user C to the conference by sending an INVITE request to the Conference Factory URI and including URI list in the INVITE request, UA-A indicates the dialogs which be re-used for this conference in the uri list by the mechanism described in 3GPP TS 24.147 [7].

```
INVITE CONF AS
To: CONF AS
From: A
Require: recipient-list-invite

Content-Type: application/resource-lists+xml
Content-Disposition: recipient-list

<?xml version="1.0" encoding="UTF-8"?>
 <resource-lists xmlns="urn:ietf:params:xml:ns:resource-lists"
    xmlns:cp="urn:ietf:params:xml:ns:copyControl">
  <list>
   <entry uri="B?Call-ID=1a&amp;From=A%3Btag%3Da1&amp;To=B%3Btag%3Db&amp;Session-ID=1"
cp:copyControl="to"/>
   <entry uri="C?Call-ID=2a&amp;From=A%3Btag%3Da2&amp;To=C%3Btag%3Dc&amp;Session-ID=2"
cp:copyControl="to"/>
  </list>
 </resource-lists>
```

2: AS verifies if the dialogs in URI list matches to a partial dialog which AS already involved, In the case of a match the AS use this dialog ID information to send re-INVITE request to UA-B and UA-C in the partial dialogs between the AS and the invited users in order to connect the media of the invited users to the MRFP.

3: After the establishment of the conference a couple of possibilities exist:

- Release the two held and the active dialogs.
- Release the active dialog between UA-A and the AS and resume one of the held dialogs to active. And let the other dialog either to user B or C in status HOLD.
- Release one of the connections either to user B or user C. Which includes the release of the UA-A to AS dialog and resume the held connection.

NOTE: There are different behaviours between mobile and fixed access devices. While mobile devices are not able to provide more than two dialogs (e.g. one held one active) while the fixed line access has no restrictions. TISPAN defined in ETSI TS 183 043 [10] the IMS-based PSTN/ISDN emulation where a couple of examples are given.

A.3 Void

Annex B (informative): Example of filter criteria

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

- Method="INVITE".

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

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

NOTE: Not invoking the MMTel AS assumes that the CONF 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**
2008-01					Publication as **ETSI TS 183 005**		2.5.0
2008-01					Conversion to **3GPP TS 24.505**		2.5.1
2008-01					Technically identical copy as **3GPP TS 24.605** as basis for further development.		2.5.1
2008-02					Implemented C1-080097, C1-080424, C1-080426		2.6.0
2008-04					Implemented C1-080878, C1-081082, C1-081083, C1-081245.		2.7.0
2008-05					Implemented C1-081550, C1-081906, C1-081909.		2.8.0
2008-05					Editorial changes done by MCC	2.8.0	2.8.1
2008-06	CT#40	CP-080326			CP-080326 was approved by CT#40 and version 8.0.0 is created by MCC for publishing	2.8.1	8.0.0
2008-09	CT#41	CP-080533	0001		Correction of reference	8.0.0	8.1.0
2008-09	CT#41	CP-080533	0002		Applicability statement in scope	8.0.0	8.1.0
2008-09	CT#41	CP-080533	0003		Interaction of HOLD and CONF	8.0.0	8.1.0
2008-12	CT#42	CP-080854	0004		Note on conference examples	8.1.0	8.2.0
2008-12	CT#42	CP-080865	0005	1	Fixed the flows	8.1.0	8.2.0
2008-12	CT#42				Editorial cleanup by MCC	8.1.0	8.2.0
2009-03	CT#43	CP-090121	0006		Correction of URN-value for Service Identifiers	8.2.0	8.3.0
2009-09	CT#45	CP-090682	0007	1	Correction of signalling flow	8.3.0	9.0.0
2009-12	CT#46	CP-090923	0009	1	Correction of icsi-ref feature tag	9.0.0	9.1.0
2010-12	CT#50	CP-100864	0010		CONF clean-up of 3pcc procedures	9.1.0	10.0.0
2012-09	CT#57				Upgrade to Rel-11	10.0.0	11.0.0
2012-12	CT#58	CP-120778	0014	2	Emergency call CONF suppression	11.0.0	11.1.0
2013-06	CT#60	CP-130413	0015	8	PSAP callback CONF suppression	11.1.0	12.0.0
2013-09	CT#61	CP-130507	0016		draft-ietf-ecrit-psap-callback reference update	12.0.0	12.1.0
2013-12	CT#62	CP-130758	0017	2	Reference update: draft-ietf-ecrit-psap-callback	12.1.0	12.2.0
2014-09	CT#65	CP-140665	0020	1	Release of held sessions after conference establishment using URI list	12.2.0	12.3.0
2014-09	CT#65	CP-140665	0021	1	Include "isfocus" for 3pcc conf setup	12.2.0	12.3.0
2014-09	CT#65	CP-140689	0022	3	Conference Factory URI for IMS	12.2.0	12.3.0
2014-12	CT#66	CP-140833	0023	1	Reference update: RFC 7090 (draft-ietf-ecrit-psap-callback)	12.3.0	12.4.0
2015-03	CT#67	CP-150067	0024	4	Update REFER to reflect RFC 6665	12.4.0	12.5.0
2015-12	CT#70				Upgrade to Rel-13	12.5.0	13.0.0
2016-06	CT#72	CP-160332	0027	1	Correcting Call Flow following RFC 7647 publication	13.0.0	14.0.0

Change history							
Date	**Meeting**	**TDoc**	**CR**	**Rev**	**Cat**	**Subject/Comment**	**New version**
2018-06	SA#80					Upgrade to Rel-15	15.0.0

History

Document history		
V15.0.0	June 2018	Publication