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Technical Specification

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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 183 005 [12]. It was transferred to the 3rd Generation Partnership Project (3GPP) in in January 2008.

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:

x the first digit:

1 presented to TSG for information;

2 presented to TSG for approval;

3 or greater indicates TSG approved document under change control.

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 Conference (CONF) service based on stage one and two of the ISDN CONF supplementary service. Within the Next Generation Network (NGN) the stage 3 is specified using the IP-Multimedia Call Control Protocol based on Session Initiation Protocol (SIP) and 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.

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*.

Referenced documents which are not found to be publicly available in the expected location might be found at http://docbox.etsi.org/Reference.

For online referenced documents, information sufficient to identify and locate the source shall be provided. Preferably, the primary source of the referenced document should be cited, in order to ensure traceability. Furthermore, the reference should, as far as possible, remain valid for the expected life of the document. The reference shall include the method of access to the referenced document and the full network address, with the same punctuation and use of upper case and lower case letters.

NOTE: While any hyperlinks included in this clause were valid at the time of publication ETSI cannot guarantee their long term validity.

2.1 Normative references

The following referenced documents are indispensable for the application of the present document. For dated references, only the edition cited applies. For non-specific references, the latest edition of the referenced document (including any amendments) applies.

[1] ETSI TS 181 002: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Multimedia Telephony with PSTN/ISDN simulation services".

[2] ETSI ES 283 003: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IP Multimedia Call Control Protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) Stage 3 [3GPP TS 24.229 [Release 7], modified]".

[3] IETF RFC 4575: "A Session Initiation Protocol (SIP) Event Package for Conference State".

[4] Void.

[5] Void.

[6] Void.

[7] ETSI TS 124 147: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Conferencing using the IP Multimedia (IM) Core Network (CN) subsystem; Stage 3 (3GPP TS 24.147 Release 7)".

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

[9] ETSI ES 283 027 " Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Endorsement of the SIP-ISUP Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks [3GPP TS 29.163 (Release 7), modified]".

[10] ETSI TS 129 163: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (3GPP TS 29.163 Release 7)".

[11] ETSI TS 183 028: " Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Common Basic Communication procedures; Protocol specification ".

2.2 Informative references

[12] ETSI TS 183 005 V2.5.0: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); PSTN/ISDN simulation services: Conference (CONF); Protocol specification".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in TS 181 002 [1] and TS 124 147 [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
CPG Call ProGress
CS Circuit Switch
ECT Explicit Communication Transfer
HOLD communication HOLD
IBCF Interworking Border Control Function
I-CSCF Interrogating Call Server Control Function
IMS IP Multimedia Subsystem
IP Internet Protocol
ISDN Integrated Service Data Network
MCID Malicious Communication IDentification
MGCF Media Gateway Control Function
NGN Next Generation Network
OIP Originating Identification Presentation
OIR Originating Identification Restriction
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.

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 TS 124 147 [7], clause 5.

4.5 Signalling requirements

4.5.1 Activation/deactivation/registration

Not applicable.

4.5.2 Invocation and operation

This clause describes the usage of and the changes to the procedures of TS 124 147 [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 TS 124 147 [7], clause 5.3.1.4 shall apply.

4.5.2.1.2 User inviting another user to a conference

Procedures according to TS 124 147 [7], clause 5.3.1.5 shall apply with the following additions to clause 5.3.1.5.3 of TS 124 147 [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 TS 124 147 [7] , clause 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 TS 183 028 [11] sub clause 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 TS 124.147 [7], clause 5.3.1.6 shall apply.

4.5.2.1.4 User creating a conference

Procedures according to TS 124.147 [7], clause 5.3.1.3 shall apply.

4.5.2.1.5 Subscription for the conference event package

Procedures according to TS 124.147 [7], clause 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 TS 124 147 [7] , clauses 5.3.2 and 6.3.2 shall apply with the following additions to clause 5.3.2.5.2 of TS 124 147 [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 REFER request's P-Asserted-Identity.

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

The procedures described in clause 5.3.2.5.5 of TS 124 147 [7] shall not apply.

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 TS 124.147 [7], clause 5.3.3 shall apply.

4.5.2.3 Actions at the incoming I-CSCF

Basic communication procedures according to ES 283 003 [2] shall apply.

4.5.2.4 Actions at the outgoing IBCF

Basic communication procedures according to ES 283 003 [2] shall apply.

4.5.2.5 Actions at the incoming IBCF

Basic communication procedures according to ES 283 003 [2] shall apply.

4.5.2.6 Actions at the destination P-CSCF

Basic communication procedures according to ES 283 003 [2] shall apply.

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 Actions at the MGCF

Procedures according to TS 124 147 [7] , clause 5.2.4 shall apply.

NOTE: In the case of an interworking a request to a PSTN user to dial into a conference (by means of sending a REFER request to the PSTN user) will result in a 403 Forbidden response from the MGCF.

4.5.2.9 Actions at the S-CSCF

Basic communication procedures according to ES 283 003 [2] shall apply.

4.6 Interaction with other supplementary services

4.6.1 Communication HOLD (HOLD)

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

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 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 Referred-by header field, if it does not exist in the REFER request; or
  b) remove Referred-By header field, if Privacy header field of the REFER request set to "user".
- If an INVITE request with "recipient-list" body is received, and if 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 Interworking with PSTN/ISDN

The procedures of TS 129 163 [10] shall apply with the additions of ES 283 027 [9] and the additions in clause 4.7.1.1.

4.7.1.1 Interworking between the IMS conference status notifications and the notification messages of the PSTN/ISDN CONF supplementary service

In this clause the interworking from the conference event package RFC 4575 [3] to the messages of the PSTN/ISDN CONF supplementary service is described. Note that an interworking from the PSTN/ISDN to the NGN is out of scope.

4.7.1.1.1 Procedures at the MGCF

4.7.1.1.1.1 Subscribing for the conference event package

Based on local policy, the MGCF may subscribe for the conference event package on behalf of the PSTN/ISDN participant after he joins or is added to a conference.

When the conference event package option is implemented, and one of the following events occurs at the MGCF:

- A 200 OK is received as a response to an initial INVITE request originated by the MGCF, where the Contact header field contains an "isfocus" parameter; or

- An ACK message is received which acknowledges a 200 OK response to the initial INVITE request, and the initial INVITE request is originated by the conferencing AS and contains an "isfocus" parameter in the Contact header field.

Then the following steps shall be performed:

1) A SUBSCRIBE request shall be created according to RFC 4575 [3];

2) The request URI is set to the Contact address of the conferencing AS;

3) The P-Asserted-Identity header field, the From header field and the Privacy header field are set with the same value as:

   - the P-Asserted-Identity header field, the From header field and the Privacy header field in the initial INVITE request originated by the MGCF; or

   - the P-Asserted-Identity header field, the To header field and the Privacy header field in a 1xx or 2xx response sent by the MGCF to the initial INVITE request from the conferencing AS.

4.7.1.1.1.2 Interworking the Notification

NOTE: There is a need to differentiate between the procedures of interworking for a full and a partial type of notification.

When a full type of notification is received a check is made of the content. If the changes with respect a previous version of the notification have not been sent on to the PSTN/ISDN for this session, the MGCF shall do an ISUP interaction towards the PSTN. If the changes with respect a previous version of the notification have been sent to the PSTN/ISDN for this session, the MGCF shall not do an ISUP interaction towards the PSTN.

When a partial notification is received then it is assumed that a value of a received notification has changed, so the MGCF shall do an ISUP interaction towards the PSTN.

- Conference established:

Upon the receipt of a conference information document with the <conference-state-type> element *active* is set to "true", the MGCF shall send a CPG message to the CS side with a notification "*conference established*".

- Participant added:

Upon the receipt of a conference information document with the <endpoint-type> and the element *status of endpoint-status-type* is set to "connected" and it was not set to "on-hold" before and the Contact URI in the element *entity* is not the address of the served PSTN/ISDN participant, the MGCF shall send a CPG message to the CS side with a notification "*other party added*".

- Served PSTN/ISDN participant isolated:

Upon the receipt of a conference information document with the <endpoint-type> and the element *status of endpoint-status-type* is set to "on-hold" and it was set to "connected" before and the Contact URI in the element *entity* is the address of the served PSTN/ISDN participant, the MGCF shall send a CPG message to the CS side with a notification "*isolated*".

- Other participant isolated:

Upon the receipt of a conference information document with the <endpoint-type> and the element *status of endpoint-status-type* is set to "on-hold" and it was set to "connected" before and the Contact URI in the element *entity* is not the address of the served PSTN/ISDN participant, the MGCF shall send a CPG message to the CS side with a notification "*other party isolated*".

- Served PSTN/ISDN participant reattached:

Upon the receipt of a conference information document with the <endpoint-type> and the element *status of endpoint-status-type* is set to "connected" and it was set to "on-hold" before and the Contact URI in the element *entity* is the address of the served PSTN/ISDN participant, the MGCF shall send a CPG message to the CS side with a notification "*reattached*".

- Other participant reattached:

Upon the receipt of a conference information document with the <endpoint-type> and the element *status of endpoint-status-type* is set to "connected" and it was set to "on-hold" before and the Contact URI in the element *entity* is not the address of the served PSTN/ISDN participant, the MGCF shall send a CPG message to the CS side with a notification "*other party reattached*".

- Other party disconnected:

Upon the receipt of a conference information document with the <endpoint-type> and the element *status of endpoint-status-type* is set to "disconnected" and the element *joining-method of joining-type* is not set to "focus-owner, the MGCF shall send a CPG message to the CS side with a notification "*other party disconnected*".

4.7.2 Interworking with PSTN/ISDN Emulation

The interworking with PSTN/ISDN Emulation is for further study.

4.7.3 Interaction with external IP network

For SIP based networks the-procedures used shall be compliant with ES 283 003 [1].

The interworking with non SIP networks is for further study.

4.8 Parameter values (timers)

Not applicable.

Annex A (informative): Signalling flows

A.1 Call flow for CONF interworking between NGN 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.

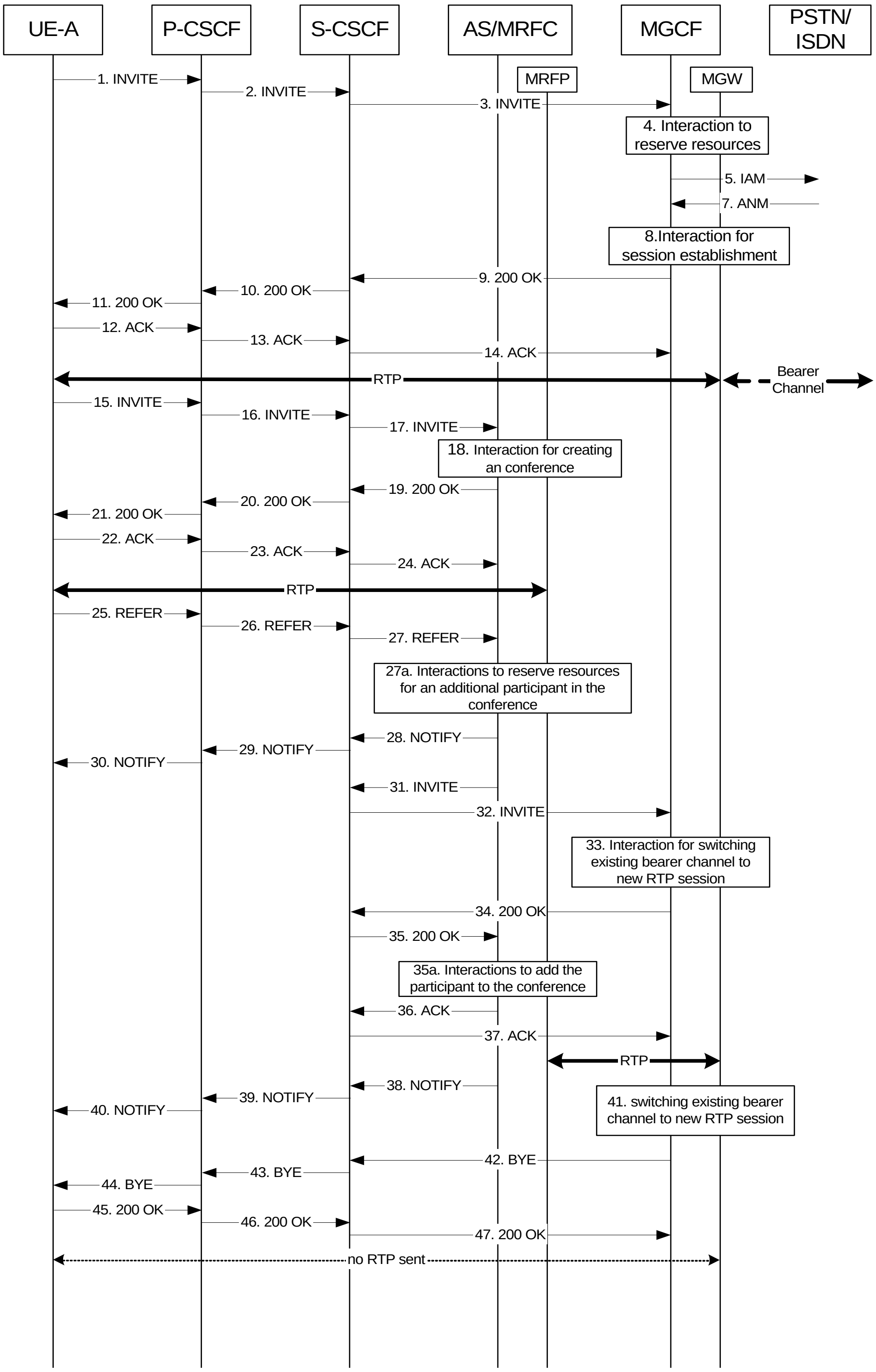

Figure A.1: CONF interworking signalling flow in case of an active communication between NGN and PSTN

- 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 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
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:[5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
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 99
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=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99 MP4V-ES
m=audio 3456 RTP/AVP 97 96
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=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
Contact: <sip:mgcf1.home1.net>
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/AVP 97 96
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=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 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
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:[5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp>
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 99
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=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
a=rtpmap:99:MPVMP4V-ES
m=audio 3456 RTP/AVP 97 96
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=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:[5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp>
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=z9hG4bK232738 46
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
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=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
m=audio 6544 RTP/AVP 97 96
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=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.

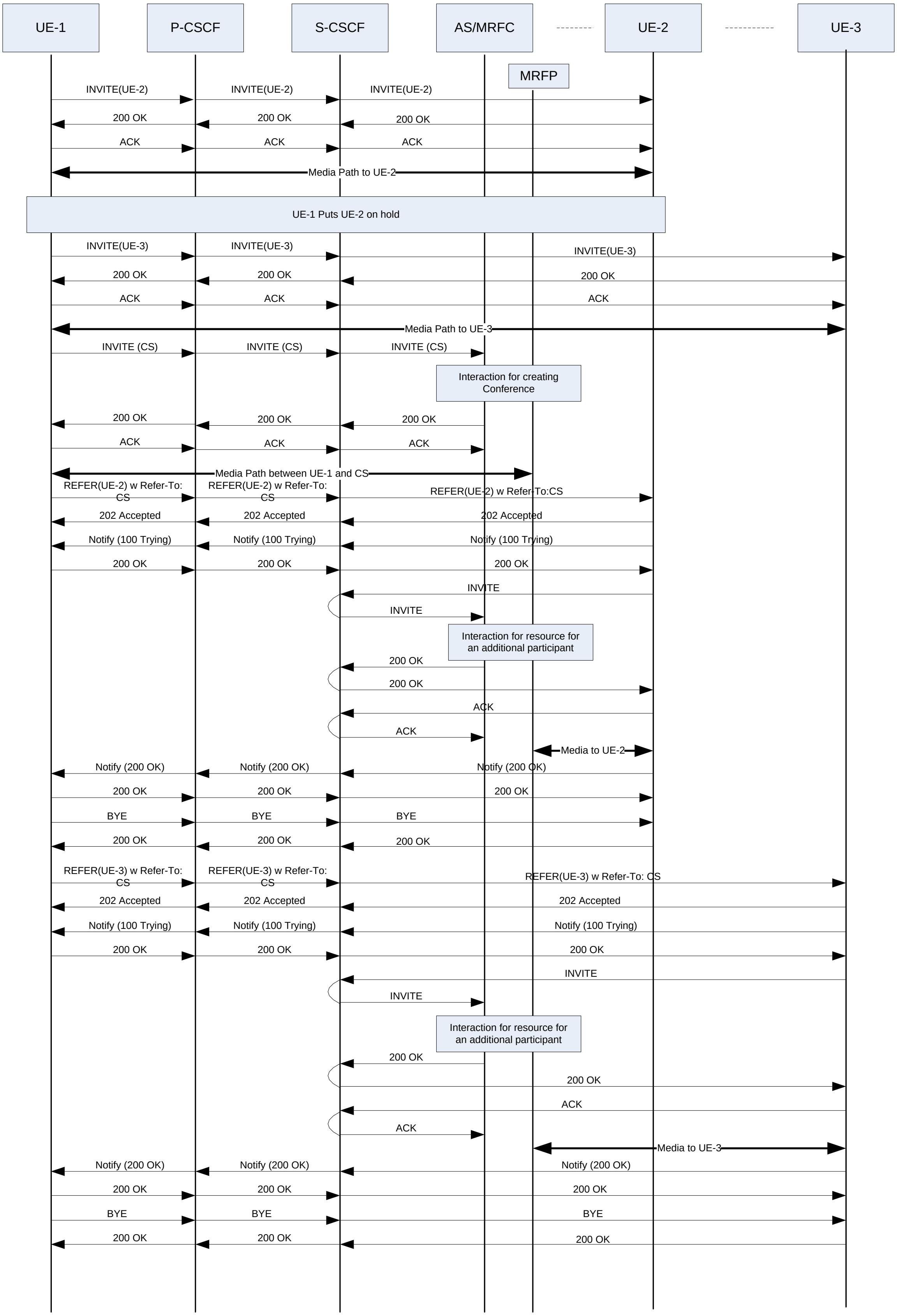

Figure A.2: Call flow for 3PTY conference

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 INVITE to the AS to create the conference and indicates that certain dialogs shall 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

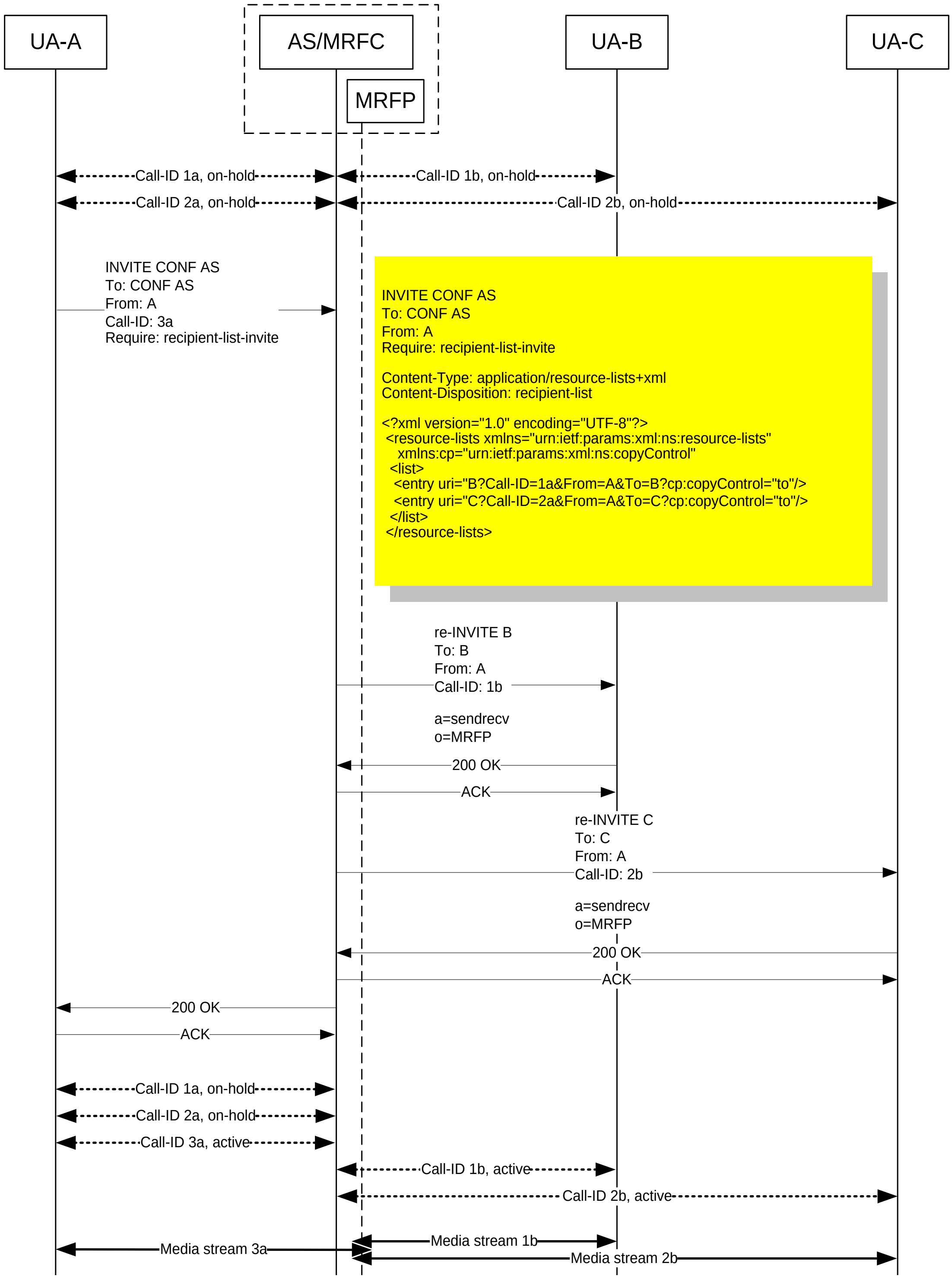

Figure A.3: Call flow for 3PTY conference

1: UA-A creates a conference and invites user B and user C to the conference by sending an INVITE to the Conference Factory URI and including URI list in the INVITE request, UA-A indicates the certain dialogs which be re-used for this conference in the uri list by ? mechanism.

```
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&From=A%3Btag%3Da&To=B%3Btag%3Db" cp:copyControl="to"/>
   <entry uri="C?Call-ID=2a&From=A%3Btag%3Da&To=C%3Btag%3Dc" 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.

Annex B (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
2008-03	CT#39	CP-080100			Based on the decision in CT#39 version 8.0.0 created by MCC	2.5.1	8.0.0
2012-12	CT#58	CP-120778	0001	1	Emergency call CONF suppression	8.0.0	8.1.0