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Technical Specification

Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); PSTN/ISDN simulation services: Conference (CONF); Protocol specification

Reference

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Keywords

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Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN).

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 and/or edition number or version number) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.

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

[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); Endorsement of "IP Multimedia Call Control Protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) Stage 3 (Release 6)" for NGN Release 1".

[3] draft-ietf-sipping-conference-package-12: "A Session Initiation Protocol (SIP) Event Package for Conference State".

[4] Void.

[5] Void.

[6] Void.

[7] ETSI TS 124 147, V7.0.0: "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 6)".

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

[9] ETSI ES 283 027 "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Interworking SIP-ISUP for TISPAN-IMS".

[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)".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in TS 181 002 [1], 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].

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] shall apply.

4.5.2.1.2 User inviting another user to a conference

Procedures according to TS 124 147 [7] shall apply with the following additions to clause 5.3.1.5.3 of TS 124 147 [7]:

- 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: 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?Replaces=cb03a0s09a2sdfglkj490333%3Bto-tag314159%3Bfrom-tag171828; method=INVITE>

4.5.2.2 Actions at the Conferencing AS

Procedures according to TS 124 147 [7] 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 5.3.2.5.4 of TS 124 147 [7] shall not 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] 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.

4.6 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)

The conference focus shall ensure that privacy policies on identity information that it includes in conference notifications are enforced.

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)

The conference focus shall ensure that privacy policies on identity information that it includes in conference notifications are enforced.

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.

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 Interaction 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 [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

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 Interaction with PSTN/ISDN Emulation

This situation is out of scope of the present document.

4.7.3 Interaction with external IP network

The procedures of ES 283 003 [2] shall apply.

4.8 Parameter values (timers)

Not applicable.

Annex A (informative): Signalling Flows

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.

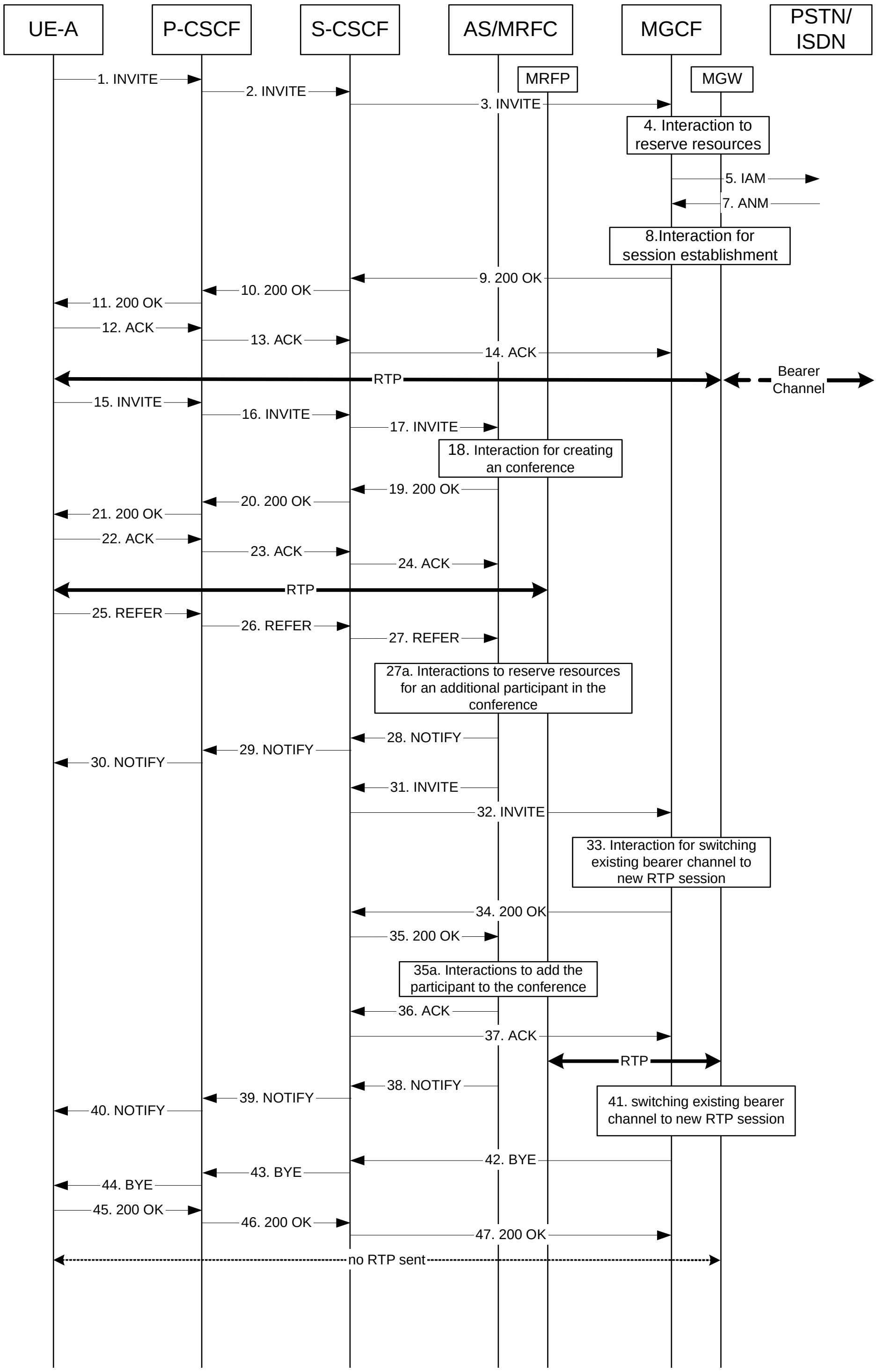

Figure A.1: CONF interworking signalling flow in case of an active communication between NGN and PSTN

- 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: precondition, sec-agree
Proxy-Require: sec-agree
Supported: 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 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: precondition, sec-agree
Proxy-Require: sec-agree
Supported: 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 request includes the Replaces header 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:user2_public1@home2.net>
Call-ID: cb03a0s09a2sdfglkj490444
Cseq: 127 REFER
Require: sec-agree
Refer-To: <sip:mgcf1.home1.net?Replaces=cb03a0s09a2sdfglkj490333%3Bto-tag314159%3Bfrom-
   tag171828 ; method=INVITE>
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
```

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: < tel:+1-212-555-2222 >
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Require: replaces
Replaces: cb03a0s09a2sdfglkj490333;to-tag=314159;from-tag=171828
Supported: 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 conference bridge.

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.

History

Document history		
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