

ETSI TS 124 628 V12.3.0 (2014-10)



**Digital cellular telecommunications system (Phase 2+);
Universal Mobile Telecommunications System (UMTS);
LTE;
Common Basic Communication procedures using
IP Multimedia (IM) Core Network (CN) subsystem;
Protocol specification
(3GPP TS 24.628 version 12.3.0 Release 12)**



Reference

RTS/TSGC-0124628vc30

Keywords

GSM,LTE,UMTS

ETSI

650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
Association à but non lucratif enregistrée à la
Sous-Préfecture de Grasse (06) N° 7803/88

Important notice

The present document can be downloaded from:

<http://www.etsi.org>

The present document may be made available in electronic versions and/or in print. The content of any electronic and/or print versions of the present document shall not be modified without the prior written authorization of ETSI. In case of any existing or perceived difference in contents between such versions and/or in print, the only prevailing document is the print of the Portable Document Format (PDF) version kept on a specific network drive within ETSI Secretariat.

Users of the present document should be aware that the document may be subject to revision or change of status. Information on the current status of this and other ETSI documents is available at

<http://portal.etsi.org/tb/status/status.asp>

If you find errors in the present document, please send your comment to one of the following services:

http://portal.etsi.org/chaicor/ETSI_support.asp

Copyright Notification

No part may be reproduced or utilized in any form or by any means, electronic or mechanical, including photocopying and microfilm except as authorized by written permission of ETSI.

The content of the PDF version shall not be modified without the written authorization of ETSI.

The copyright and the foregoing restriction extend to reproduction in all media.

© European Telecommunications Standards Institute 2014.

All rights reserved.

DECT™, **PLUGTESTS™**, **UMTS™** and the ETSI logo are Trade Marks of ETSI registered for the benefit of its Members. **3GPP™** and **LTE™** are Trade Marks of ETSI registered for the benefit of its Members and of the 3GPP Organizational Partners.

GSM® and the GSM logo are Trade Marks registered and owned by the GSM Association.

Intellectual Property Rights

IPRs essential or potentially essential to the present document may have been declared to ETSI. The information pertaining to these essential IPRs, if any, is publicly available for **ETSI members and non-members**, and can be found in ETSI SR 000 314: *"Intellectual Property Rights (IPRs); Essential, or potentially Essential, IPRs notified to ETSI in respect of ETSI standards"*, which is available from the ETSI Secretariat. Latest updates are available on the ETSI Web server (<http://ipr.etsi.org>).

Pursuant to the ETSI IPR Policy, no investigation, including IPR searches, has been carried out by ETSI. No guarantee can be given as to the existence of other IPRs not referenced in ETSI SR 000 314 (or the updates on the ETSI Web server) which are, or may be, or may become, essential to the present document.

Foreword

This Technical Specification (TS) has been produced by ETSI 3rd Generation Partnership Project (3GPP).

The present document may refer to technical specifications or reports using their 3GPP identities, UMTS identities or GSM identities. These should be interpreted as being references to the corresponding ETSI deliverables.

The cross reference between GSM, UMTS, 3GPP and ETSI identities can be found under <http://webapp.etsi.org/key/queryform.asp>.

Modal verbs terminology

In the present document "**shall**", "**shall not**", "**should**", "**should not**", "**may**", "**may not**", "**need**", "**need not**", "**will**", "**will not**", "**can**" and "**cannot**" are to be interpreted as described in clause 3.2 of the [ETSI Drafting Rules](#) (Verbal forms for the expression of provisions).

"**must**" and "**must not**" are **NOT** allowed in ETSI deliverables except when used in direct citation.

Contents

Intellectual Property Rights	2
Foreword.....	2
Modal verbs terminology.....	2
Foreword.....	5
1 Scope	6
2 References	6
3 Definitions and abbreviations.....	7
3.1 Definitions	7
3.2 Abbreviations	7
4 Common basic communication procedures.....	8
4.1 Introduction	8
4.2 Announcement	8
4.2.1 General.....	8
4.2.2 Providing announcements to a user during the establishment of a communication session	8
4.2.3 Providing announcements to a user during an established communication session.....	8
4.2.4 Communication request rejected by AS.....	9
4.2.5 Providing announcements to a user during the release of a communication session	9
4.2.6 Providing announcements to a terminating user just after the call is answered and before establishing direct communication session between end users	9
4.3 Alternative ring tone.....	9
4.3A Voicemail server identification	9
4.4 Network Determined User Busy (NDUB).....	10
4.4a Special REFER request handling procedures	10
4.4b Screening of 200 (OPTIONS) response content.....	11
4.5 Operational requirements	11
4.5.1 Provision/withdrawn.....	11
4.5.2 Requirements on the originating network side.....	11
4.5.3 Requirements on the terminating network side.....	11
4.6 Coding requirements	12
4.7 Signalling procedures	12
4.7.1 Activation, deactivation	12
4.7.1A Registration/erasure	12
4.7.1B Interrogation	12
4.7.2 Invocation and operation	12
4.7.2.1 Actions at the originating UE.....	12
4.7.2.2 Void.....	13
4.7.2.3 Void.....	13
4.7.2.4 Void.....	13
4.7.2.5 Void.....	13
4.7.2.6 Void.....	13
4.7.2.7 Void.....	13
4.7.2.8 Void.....	13
4.7.2.9 Actions at the AS	13
4.7.2.9.1 Providing announcements during an established communication session.....	13
4.7.2.9.2 Providing announcements during the establishment of a communication session	13
4.7.2.9.3 Providing announcements when communication request is rejected by the AS.....	13
4.7.2.9.4 Providing alternative ring tone during the establishment of a communication session	14
4.7.2.9.5 Early dialog procedures at the AS	14
4.7.2.9.6 Providing announcements during the release of a communication session	14
4.7.2.9.7 Starting special REFER handling procedures at the AS of the initiator of the REFER request	14
4.7.2.9.8 Voicemail server.....	16
4.7.2.9.9 Providing announcements to a terminating user just after the call is answered and before establishing direct communication session between end users.....	16

4.7.2.9.10	Screening of 200 (OPTIONS) response content.....	17
4.7.2.10	Action at the terminating UE	17
4.8	Interactions with other networks	17
4.8.1	Void	17
4.8.2	Void	17
4.8.3	Void	17
4.9	Signalling flows.....	17
4.10	Parameter values (timers).....	18
Annex A (informative): Signalling flows for announcements.....		19
A.1	Providing announcements to a user during the establishment of a communication session	19
A.1.1	Providing in-band announcement.....	19
A.1.2	Including Alert-Info header field in the 180 (Ringing) response	22
A.1.3	Announcements provided by the PSTN/ISDN	23
A.1.4	Announcement provided towards a user connected to the PSTN/ISDN	25
A.1.5	Providing in-band announcement after a reliable provisional response has been received by the terminating UE	26
A.2	Providing announcements to a user during an established communication	28
A.2.1	Scenario 1: UE - AS - UE	29
A.2.2	Scenario 2: UE - AS/MRFC/MRFP - UE.....	30
A.3	Communication request rejected.....	31
A.3.1	Sending the announcement as in-band information.....	31
A.3.1.1	Using early media	31
A.3.1.2	Using an established session.....	34
A.3.2	Including an Error-Info header field in a 3xx, 4xx, 5xx and 6xx response	36
A.3.3	Announcements provided by the PSTN/ISDN.....	38
A.3.4	Announcement provided to a user connected to the PSTN/ISDN.....	38
A.4	Providing announcements to a user during the release of a communication session.....	38
A.4.1	Scenario 1: UE - AS - UE	39
A.4.2	Scenario 2: UE - AS/MRFC/MRFP - UE.....	40
A.5	Providing announcements to a terminating user just after the call is answered and before establishing direct communication session between end users	41
A.5.1	Switch media path from (UE-AS/MRFC/MRFP) to (UE-UE) within the same dialog.....	41
A.5.2	Using 180 (Ringing) response towards originating UE.....	43
Annex B (informative): Signalling flows for Network Determined User Busy (NDUB).....		46
B.1	Basic call with UE busy with T-AS involvement (NDUB condition check)	46
B.2	Busy condition (NDUB) detected by terminating AS.....	47
Annex C (normative): Void		49
Annex D (normative): AS establishing multiple dialogs with originating UE.....		50
D.1	General	50
Annex E (informative): Signalling flows for 3rd party call control		51
Annex F (informative): Void		57
Annex G (informative): Change history		58
History		59

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 183 028 [17]. It was transferred to the 3rd Generation Partnership Project (3GPP) 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:

Version x.y.z

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 describes the stage three protocol for basic communication procedures common to several services in the IP Multimedia (IM) Core Network (CN) subsystem when at least one Application Server (AS) is included in the communication. The common procedures are based on stage three specifications for supplementary services.

The present document contains examples of signalling flows for the common basic communication procedures.

The present document is applicable to User Equipment (UE) and Application Servers (AS) which are intended to support the common basic communication procedures.

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: "IP Multimedia Call Control Protocol based on SIP and SDP" ..
- [2] Void.
- [3] Void.
- [4] IETF RFC 3261: "SIP: Session Initiation Protocol".
- [5] IETF RFC 3262: "Reliability of Provisional Responses in the Session Initiation Protocol (SIP)".
- [6] IETF RFC 3960: "Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)".
- [7] ETSI TS 181 005: "Telecommunications and Internet Converged Services and Protocols for Advanced Networking (TISPAN); Service and Capability Requirements".
- [8] Void.
- [9] 3GPP TS 29.163: "Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks".
- [10] Void.
- [11] ITU-T Recommendation I.112: "Vocabulary of terms for ISDNs".
- [12] IETF RFC 5009: "Private Header (P-Header) Extension to the Session Initiation Protocol (SIP) for Authorization of Early Media".
- [13] IETF RFC 3515: "The Session Initiation Protocol (SIP) Refer Method".
- [14] IETF RFC 3725: "Best Current Practices for Third Party Call Control (3pcc) in the Session Initiation Protocol (SIP)".
- [15] 3GPP TS 24.607: "Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR) using IP Multimedia (IM)Core Network (CN) subsystem; Protocol specification".

- [16] Void.
- [17] ETSI TS 183 028 V2.4.0: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Common Basic Communication procedures; Protocol specification".
- [18] IETF RFC 6228 (May 2011): "Response Code for Indication of Terminated Dialog".
- [19] IETF RFC 3840: "Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)"
- [20] IETF RFC 4596: "Guidelines for usage of the Session Initiation Protocol (SIP) Caller Preferences Extension
- [21] IETF RFC 6665 (July 2012): "SIP-Specific Event Notification".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply:

announcement: service related message sent to a user that can be of any type of media e.g. a voice message or a video-clip

communication: transfer of information between two or more users, entities, processes or nodes according to some agreed conventions

NOTE: See ITU-T Recommendation I.112 modified [11].

early media: media sent before a communication is established

in-band announcement: announcement sent by the network using the bearer established for a communication

Originating Application Server (O-AS): controlling application server responsible for the services provided to the originating user

Terminating Application Server (T-AS): controlling application server responsible for the services provided to the terminating user

3.2 Abbreviations

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

3pcc	3 rd party call control
ACR	Automatic Call Rejection
AS	Application Server
B2BUA	Back-to-Back User Agent
IFC	Initial Filter Criteria
IMS	IP Multimedia Subsystem
ISDN	Integrated Services Digital Network
MGCF	Media Gateway Control Function
MGW	Media GateWay
MRFC	Media Resource Function Controller
MRFP	Media Resource Function Processors
NDUB	Network Determined User Busy
O-AS	Originating Application Server
P-CSCF	Proxy Call Session Control Function
PSTN	Public Switched Telephone Network
S-CSCF	Serving Call Session Control Function
SDP	Session Description Protocol

SIP	Session Initiation Protocol
T-AS	Terminating Application Server
T-MGF	Trunking Media Gateway Function
UDUB	User Determined User Busy
UE	User Equipment
URL	Uniform Resource Locator

4 Common basic communication procedures

4.1 Introduction

Services may need to send announcements for example to explain the reason for rejecting a communication request or to report the progress of a communication request. The announcement may be of any type of media e.g. an audio announcement or a video clip. Subclause 4.2 describes the announcement common procedure and annex A shows examples of signalling flows for some announcement scenarios.

Some services are triggered by a user's busy condition e.g. the Communication Forwarding on Busy service. The busy condition may be determined by the network i.e. the Network Determine User Busy (NDUB) condition or by the user i.e. the User Determine User Busy (UDUB) condition. Subclause 4.4 describes the network determine user busy common procedure and the annex B shows examples of signalling flows for some busy scenarios.

Some services are triggered by sending a REFER request, for example Explicit Communication Transfer. A receiver of the REFER request in some cases might not be able to process the REFER request. Subclause 4.4a describes fallback procedures to 3rd party call control. Annex E provides some examples for signalling flows.

4.2 Announcement

4.2.1 General

Announcements may be sent during the establishment of a communication session, when rejecting a communication request, during an established communication session or during the release of a communication session.

4.2.2 Providing announcements to a user during the establishment of a communication session

A service may provide an announcement during the establishment of a communication. If an announcement is provided the service shall use one of the following methods:

- use an Alert-Info header field in the 180 (Ringing) response to the INVITE request; or
- use early media as defined by IETF RFC 3960 [6] and using the P-Early-Media header field authorizing early media as defined in IETF RFC 5009 [12] for the gateway model; or
- use multiple early dialogs as described in annex D and using the P-Early-Media header field authorizing early media as defined in IETF RFC 5009 [12].

4.2.3 Providing announcements to a user during an established communication session

A service may provide an announcement during an established communication. If an announcement is provided the service shall use one of the following methods:

- use an Call-Info header field in a re-INVITE request; or
- use the existing media stream. The media stream may have to be re-negotiated by the service to a media type suitable for the announcement.

Mixing announcements into an existing media stream requires that the AS use the 3rd party call control procedure as specified by subclause 5.7.5 in 3GPP TS 24.229 [1].

4.2.4 Communication request rejected by AS

A service may provide an announcement when rejecting a communication request e.g. in order to explain the reason for rejecting the communication request in more detail. If an announcement is provided the service shall:

- use an Error-Info header field in the 3xx, 4xx, 5xx or 6xx response to the INVITE request; or
- use early media for sending the announcement in-band as defined by IETF RFC 3960 [6] and using the P-Early-Media header field authorizing early media as defined in IETF RFC 5009 [12] for the gateway model and insert the Reason Header with the proper cause value; or
- use early media for sending the announcement in-band in an early dialog as described in annex D and using the P-Early-Media header field authorizing early media as defined in IETF RFC 5009 [12] and insert the Reason Header with the proper cause value; or
- accept the communication request and use the established session for sending an in-band announcement.

4.2.5 Providing announcements to a user during the release of a communication session

A service may provide an announcement to the UE, who does not end the session, during the release of a communication, in order to, e.g. tell the charge information. If an announcement is provided the service shall:

- use the existing media stream. The media stream may have to be re-negotiated by the service to a media type suitable for the announcement; or
- change to new media for sending the announcement.

4.2.6 Providing announcements to a terminating user just after the call is answered and before establishing direct communication session between end users

When a call is established, a service may, before allowing media to be exchanged between the end points, provide an announcement to the terminating user after the call is answered.

When the session initiation request is sent from the originating UE, the AS will act as a B2BUA and modify the SDP offer so that it represents an MRFP handling the announcement, and a media stream will be established between the MRFP and the terminating UE once the called user answers the call.

When the announcement is completed, the AS will send a new SDP offer, based on the SDP offer initially received in the session initiation request, to the terminating UE, in order to remove the MRFP from the media path and allow media to be exchanged between the end points.

4.3 Alternative ring tone

A service may provide an alternative ring tone using the Alert-Info header field as specified by IETF RFC 3261 [4].

The intention with this alternative ring tone is to override local ring tones provided by the UE. It is recommended that the size of the referenced alternative ring tone is small in order not to delay communication establishment.

4.3A Voicemail server identification

When a voicemail server answers a call:

1. If the voicemail server is able to record a message, the voicemail server shall insert in the SIP 200 (OK) response to the INVITE request the media features tag automata and actor="msg-taker" in the Contact header as described in IETF RFC 3840 [19] and in IETF RFC 4596 [20].
2. If the voicemail server is not able to record a message, the voicemail server shall insert in the SIP 200 (OK) response to the INVITE request the media feature tag automata in the Contact header as described in IETF RFC 3840 [19] and in IETF RFC 4596 [20]. In that case, the media feature actor="msg-taker" shall not be inserted in the SIP 200 (OK) response to the INVITE request.

4.4 Network Determined User Busy (NDUB)

Deployment of some service may require the support of the optional service requirements for "network determined user busy" and "approaching network determined user busy" defined in TS 181 005 [7]. In order to support such requirements it is assumed that a network function/application server is deployed to track a user's busy condition status from the perspective of the network.

The present document is applicable only in cases whereby the network operator has complete knowledge of the applications to which an end user has subscribed and assumes that those applications will furnish the network entity responsible for tracking "busy condition" with appropriate information to enable this determination to be made. This may require appropriate business arrangements between the network operator and the application provider.

NOTE: Tracking bandwidth availability in the customer network is out of scope of the current release. As such it is possible that a communication could be presented based on the network entity determining that the communication can be presented when in fact congestion in the customer network will prevent the communication being presented. This is a limitation of the present document.

Determination of "network determined user busy" by the network may restrict the ability to deploy and support end user devices which perform local services based on "user determined user busy" as part of their base functionality.

4.4a Special REFER request handling procedures

After the reception of a REFER request the AS may start 3pcc procedures under the following conditions:

- the Application Server acts as a B2BUA, so the AS has knowledge about the existing partial dialogs it is involved in, especially of the media user for this communication; and
- the REFER request is routed via this AS.

The 3pcc procedures shall be achieved by sending re-INVITE requests in existing partial dialogs and by sending INVITE requests to establish new partial dialogs.

Tables 1 and 2 give decision criteria when to start 3pcc procedures.

Table 1: Terminating party of a communication sends a REFER request

Content of the Allow header in the initial INVITE request from A->B	Action AS-B on the REFER request from B	Action that the AS-B does on the initial INVITE request
INVITE request with Allow header with no REFER token	Invoke the 3pcc procedure directly	AS-B adds the REFER token to the Allow header
INVITE request with Allow header with a REFER token	Forward the REFER request and if the 403 (Forbidden) or 501 (Not implemented) response is received, fall back to 3pcc procedure	No modification needed in the Allow header
INVITE request without Allow header	Forward the REFER and if the 403 (Forbidden) or 501 (Not implemented) response is received, fall back to 3pcc procedure	No modification needed in the INVITE request

Table 2: Originating party of a communication sends a REFER request

Content of the Allow header in the 200 (OK) response on the initial INVITE request (A->B dialog)	Action AS-A on the REFER request from A	Action that the AS-A does on the 200 (OK) response on A-B dialog
200 (OK) response with Allow header with no REFER token	Invoke the 3pcc procedure directly	AS-A adds the REFER token to the Allow header
200 (OK) response with Allow header with a REFER token	Forward the REFER request and if the 403 (Forbidden) or 501 (Not implemented) response is received, fall back to 3pcc procedure	No modification needed in the Allow header
200 (OK) response without Allow header	Forward the REFER request and if the 403 (Forbidden) or 501 (Not implemented) response is received, fall back to 3pcc procedure	No modification needed in the 200 (OK) response

As a network option, an AS of the initiator of the REFER request that has prior knowledge that the remote party is not allowed to receive or does not support the REFER request, may initiate 3rd party call control procedures directly.

To avoid a longer re-negotiation of the media, the media information of the existing partial dialogs are used for the INVITE requests or the first re-INVITE requests during the 3pcc procedures.

4.4b Screening of 200 (OPTIONS) response content

Some services may use OPTIONS request to discover the UE capabilities. According to RFC 3261 [4], a UE receiving an OPTIONS request generates the same SIP response as if the request was an INVITE request. If a 200 (OK) response is sent, it will contain an SDP description of the UE media capabilities as well as a Contact header field containing the supported media feature tags. This feature may be used by malicious entities to get relevant information about the reachability means and the capabilities of the user and, thus, to maliciously use this information; for spamming for example.

Screening the content of the 200 (OK) response allows to avoid delivering some information on the UE (and therefore on the user) to certain originators of OPTIONS requests.

4.5 Operational requirements

4.5.1 Provision/withdrawn

No special requirements for provision/withdrawn. Any requirements on provision/withdrawn belong to the service using the common basic procedures specified by the present document.

4.5.2 Requirements on the originating network side

There are no service specific requirements on the originating network side defined.

NOTE: If required by local policy the IBCF will remove an Error-Info header field, Call-Info header field or an Alert-Info header field.

4.5.3 Requirements on the terminating network side

There are no service specific requirements on the terminating network side defined.

NOTE: If required by local policy the IBCF will remove an Error-Info header field, Call-Info header field or an Alert-Info header field.

4.6 Coding requirements

The syntax for the relevant headers in the SIP requests and SIP responses shall be as follows:

- The syntax of the Alert-Info header field conforms to the requirements in 3GPP TS 24.229 [1] and IETF RFC 3261 [4].
- The syntax of the Error-Info header field conforms to the requirements in 3GPP TS 24.229 [1] and IETF RFC 3261 [4].
- The syntax of the Call-Info header field conforms to the requirements in 3GPP TS 24.229 [1] and IETF RFC 3261 [4].
- The syntax of the P-Early-Media header field is described in IETF RFC 5009 [12].
- The syntax of the Allow header field conforms to the requirements in 3GPP TS 24.229 [1] and IETF RFC 3261 [4].

4.7 Signalling procedures

4.7.1 Activation, deactivation

There are no procedures for activation or deactivation defined.

4.7.1A Registration/erasure

There are no procedures for registration or erasure defined.

4.7.1B Interrogation

There are no procedures for interrogation defined.

4.7.2 Invocation and operation

4.7.2.1 Actions at the originating UE

Procedures according to 3GPP TS 24.229 [1] shall apply.

Certain services require the usage of the Alert-Info header field, Call-Info header field and Error-Info header field according to procedures specified by IETF RFC 3261 [4].

If the UE detects that in-band information is received from the network as early media, the in-band information received from the network shall override locally generated communication progress information.

The UE shall not generate the locally generated communication progress information if an early dialog exists where the last received P-Early-Media header field as described in IETF RFC 5009 [12] contains "sendrecv" or "sendonly".

NOTE: if an early dialog exists where a SIP 18x response to the SIP INVITE request other than 183 (Session Progress) response was received, no early dialog exists where the last received P-Early-Media header field as described in IETF RFC 5009 [12] contained "sendrecv" or "sendonly" and in-band information is not received from the network, then the UE is expected to render the locally generated communication progress information.

If the UE supports the P-Early-Media header field as defined in IETF RFC 5009 [12], and a P-Early-Media header field has been received, then the UE shall send any available user generated media, e.g. speech or DTMF, on media stream(s) associated with the early dialog for which the most recent P-Early-Media header field, as described in IETF RFC 5009 [12], contained a "sendrecv" or a "recvonly" header field value. If there is more than one such early dialog, the UE shall use the early dialog where the P-Early-Media header field was most recently received.

If the UE receives a re-INVITE request containing no SDP offer, the UE shall send a 200 (OK) response containing an SDP offer according to 3GPP TS 24.229 [1] indicating the directionality used by UE as

- "sendonly" if the re-INVITE request is received on a dialog where the associated communication session has been put on hold by the user; and
- "sendrecv" otherwise.

4.7.2.2 Void

4.7.2.3 Void

4.7.2.4 Void

4.7.2.5 Void

4.7.2.6 Void

4.7.2.7 Void

4.7.2.8 Void

4.7.2.9 Actions at the AS

The procedures in this subclause apply for the AS serving the originating UE and the AS serving the terminating UE.

An AS using the MRFC and MRFP to send in-band media for announcements shall use the 3rd party call control procedure as specified by subclause 5.7.5 in 3GPP TS 24.229 [1] and the media control procedure as specified by subclause 10.2 in 3GPP TS 24.229 [1].

4.7.2.9.1 Providing announcements during an established communication session

Services may use the Call-Info header field according to procedures specified by IETF RFC 3261 [4] to provide an announcement during an established communication session.

Services may send an in-band message or media using an existing media-stream to provide an announcement during an established communication session.

4.7.2.9.2 Providing announcements during the establishment of a communication session

The AS may use the Call-Info header field according to procedures specified by IETF RFC 3261 [4] in order to provide an announcement, or may use the Alert-Info header field to provide an alternative ring tone, as specified in subclause 4.7.2.9.4, during the establishment of a communication session.

The AS may use the MRFC and the MRFP to send an in-band announcement using early media according to the rules and procedures of the IETF RFC 3261 [4], IETF RFC 3262 [5], IETF RFC 3960 [6] and IETF RFC 5009 [12].

4.7.2.9.3 Providing announcements when communication request is rejected by the AS

The AS may use the Error-Info header field according to procedures specified by IETF RFC 3261 [4] in order to provide an announcement when the establishment of a communication session is rejected.

The AS may use the MRFC and MRFP to send an in-band announcement using early media according to the rules and procedures of the IETF RFC 3261 [4], IETF RFC 3262 [5], IETF RFC 3960 [6] and IETF RFC 5009 [12].

4.7.2.9.4 Providing alternative ring tone during the establishment of a communication session

The AS may use the Alert-Info header field according to procedures specified by IETF RFC 3261 [4] in order to provide an alternative ring tone during the establishment of a communication session.

4.7.2.9.5 Early dialog procedures at the AS

The procedures for dealing with early dialog established between the AS and the originating UE is described in annex D.

4.7.2.9.6 Providing announcements during the release of a communication session

Services may send an in-band message or media using an existing media-stream or changing to new media-stream to provide an announcement during the release of a communication session.

4.7.2.9.7 Starting special REFER handling procedures at the AS of the initiator of the REFER request

4.7.2.9.7.1 REFER is sent inside a dialog

4.7.2.9.7.1.1 Normal procedures

If the AS receives a 403 Forbidden or a 501 Not implemented in response to a REFER request forwarded by the AS, it shall send a 202 Accepted response followed by a NOTIFY request with a 100 (Trying) status line to the originator of the REFER request, according to the procedures of IETF RFC 3515 [13] as updated by IETF RFC 6665 [21].

The AS then shall perform third party call control procedures according to Flow III or Flow IV of IETF RFC 3725 [14], with the following additions and clarifications.

The AS should verify if it is involved in the dialogs between the originator of the REFER request on one side and the REFER target and the Refer-to target on the other side.

Then the AS shall send an INVITE request to the Refer-to target if it **is not** involved in a dialog with the Refer-to target (e.g. Blind ECT), or the AS shall send a re-INVITE request to the Refer-to target if it **is** involved in a dialog with the Refer-to target (e.g. Consultative ECT). The INVITE request shall contain if available a P-Asserted-ID header field with a valid identity of the REFER target and a Referred-by header field matching the P-Asserted-Identity of the REFER request. When including the P-Asserted-Identity the AS shall also include the Privacy header fields obtained from the request or response in which this P-Asserted-Identity was obtained. In addition the AS shall include a P-Served-User header field including a valid identity of the referor.

When the partial dialog with the Refer-to target is acknowledged following a 200 (OK), the AS shall send in the original partial dialog a NOTIFY request with a 100 Trying status line to the originator of the REFER request, according to the procedures of IETF RFC 3515 [13] as updated by IETF RFC 6665 [21]. After that the AS shall send a re-INVITE request to the REFER target. The re-INVITE request shall contain if available a P-Asserted-ID header field with a valid identity of the Refer-to target and a Referred-by header field matching the P-Asserted-Identity of the REFER request.

When the partial dialog with the REFER target is acknowledged following a 200 OK, the AS shall send in the original partial dialog a NOTIFY request with a 200 OK status line to the originator of the REFER request, according to the procedures of IETF RFC 3515 [13] as updated by IETF RFC 6665 [21]. If a Replaces parameter is included in the Refer-To header field of the original REFER request and it refers to the original partial dialog between the referer and the refer-to target, the AS shall send a BYE request in the original partial dialog to the referer.

When the 3rd party call control procedures were successful, continued processing procedures according to clause 7 of IETF RFC 3725 [14] shall be applied.

As a network option, the AS could send a 202 (Accepted) response directly and initiate 3rd party call control procedures without trying to forward the REFER request to the REFER target.

NOTE 1: For example, when UE-A and UE-B establish a session, they will exchange their own capabilities for SIP methods by using "Allow" header. If the AS lies in the signalling path between UE-A and UE-B, it knows whether the two UEs support REFER or not, and can initiate 3rd party call control procedures. Another example is that a network operator does not want to send REFERs to a user because of security reasons.

NOTE 2: The AS can enforce OIR privacy settings on OIR relevant headers carried in the generated INVITE request and/or reINVITE request, as specified 3GPP TS 24.607 [15] for regular INVITE requests originated by the served user.

NOTE 3: The AS can generate charging events for the generated INVITE requests, correlated to the initiator of the REFER request.

4.7.2.9.7.1.2 Exceptional procedures

If the 3rd party call control procedures fail because a media negotiation between REFER target and Refer-to target is not possible (e. g. the codes cannot be negotiated or the offered ports have changed in a subsequent SDP offer), or REFER target or Refer-to target answer the INVITE request with an error response, error handling procedures according to clause 6 of IETF RFC 3725 [14] shall be applied. Additionally the AS shall send a NOTIFY for terminating the original REFER request.

4.7.2.9.7.2 REFER is sent outside a dialog

4.7.2.9.7.2.1 Normal procedures

If the AS receives a 403 (Forbidden) response or a 501 (Not Implemented) response in response to a REFER request forwarded by the AS, it shall send a 202 (Accepted) response followed by a NOTIFY request with a 100 (Trying) status line to the originator of the REFER request, according to the procedures of IETF RFC 3515 [13] as updated by IETF RFC 6665 [21].

The AS then shall perform third party call control procedures according to Flow III or Flow IV of IETF RFC 3725 [14], with the following additions and clarifications:

The AS shall send an INVITE request to the Refer-to target. The INVITE request shall contain if available a P-Asserted-ID header field with a valid identity of the REFER target and a Referred-by header field matching the P-Asserted-Identity of the REFER request. In addition the AS shall include a P-Served-User header field including a valid identity of the referor.

When the dialog with the Refer-to target is acknowledged following a 200 (OK), the AS shall send in the REFER dialog a NOTIFY request with a 100 (Trying) status line to the originator of the REFER request, according to the procedures of IETF RFC 3515 [13] as updated by IETF RFC 6665 [21]. After that the AS shall send an INVITE request to the REFER target. The INVITE request shall contain if available a P-Asserted-ID header field with a valid identity of the Refer-to target and a Referred-by header field matching the P-Asserted-Identity of the REFER request. When including the P-Asserted-Identity the AS shall also include the Privacy header fields obtained from the request or response in which this P-Asserted-Identity was obtained.

When the dialog with the REFER target is acknowledged following a 200 OK, the AS shall send in the REFER dialog a NOTIFY request with a 200 OK status line to the originator of the REFER request, according to the procedures of IETF RFC 3515 [13] as updated by IETF RFC 6665 [21].

When the 3rd party call control procedures were successful, continued processing procedures according to clause 7 of IETF RFC 3725 [14] shall be applied.

As a network option, the AS could send a 202 (Accepted) response directly and initiate 3rd party call control procedures without trying to forward the REFER request to the REFER target.

NOTE 1: For example, when UE-A and UE-B establish a session, they will exchange their own capabilities for SIP methods by using "Allow" header. If the AS lies in the signalling path between UE-A and UE-B, it knows whether the two UEs support REFER or not, and can initiate 3rd party call control procedures. Another example is that a network operator does not want to send REFERs to a user because of security reasons.

NOTE 2: The AS can enforce OIR privacy settings on OIR relevant headers carried in the generated INVITE request and/or reINVITE request, as specified 3GPP TS 24.607 [15] for regular INVITE requests originated by the served user.

NOTE 3: The AS can generate charging events for the generated INVITE requests, correlated to the initiator of the REFER request.

4.7.2.9.7.2.2 Exceptional procedures

If the 3rd party call control procedures fail because a media negotiation between REFER target and Refer-to target is not possible, or REFER target or Refer-to target answer the INVITE request with an error response, error handling procedures according to clause 6 of IETF RFC 3725 [14] shall be applied. Additionally the AS shall send a NOTIFY for terminating the original REFER request.

4.7.2.9.8 Voicemail server

When an AS acts as a voicemail server generates a 200 (OK) response to an INVITE request:

1. If the AS is able to record a message, the AS shall insert in the SIP 200 (OK) response to the INVITE request the media features tag automata and actor="msg-taker" in the Contact header as described in IETF RFC 3840 [19] and in IETF RFC 4596 [20].
2. If the AS is not able to take a message, the AS shall insert in the SIP 200 (OK) response to the INVITE request the media feature tag automata in the Contact header as described in IETF RFC 3840 [19] and in IETF RFC4596 [20]. In that case, the media feature actor="msg-taker" shall not be inserted in the 200 (OK) SIP response to the INVITE request.

4.7.2.9.9 Providing announcements to a terminating user just after the call is answered and before establishing direct communication session between end users

Services can provide an announcement to a terminating user just after the call is answered and before establishing a direct communication session between end users using the procedures in this subclause.

When an INVITE request is received from an originating UE, the AS shall

- act as B2BUA;
- modify the SDP offer received from the originating UE with the media parameters necessary for the announcement;
- indicate that resources at the MRF are available; and
- send an INVITE request to the terminating UE as specified in 3GPP TS 24.229 [4].

Upon receipt of an 18x provisional response from the terminating UE the AS shall indicate the progress of the call towards the originating UE, e.g. send a 180 (Ringing) response or start an announcement.

When the session between the AS and the terminating UE is established the AS shall start the announcement.

When the announcement is completed, the AS shall send a re-INVITE request or an UPDATE request towards the terminating UE as specified in 3GPP TS 24.229 [4]. The SDP offer:

NOTE: An UPDATE request can be used if the required bandwidth is the same or lower for the call as for the announcement towards the terminating UE.

- shall include the SDP offer received from the originating UE;
- if not included in the original offer from the originating UE, all media for providing the announcement shall be removed (i.e. media lines are set to port "0"); and
- shall indicate that the resources at the originating side are not available if preconditions are used.

For the rest of the call the AS shall:

- if needed, modify the sdp sent towards the originating UE so that the content of the sdp aligns with the offer/answer between the originating UE and the AS;
- forward SIP requests and SIP responses received from the terminating UE towards the originating UE without changing the SDP using the SIP dialog created above.

For the rest of the call the AS shall:

- if needed, modify the sdp sent towards the terminating UE so that the content of the sdp aligns with the offer/answer between the terminating UE and the AS; and
- forward SIP requests and SIP responses received from the originating UE towards the terminating UE without changing the SDP.

4.7.2.9.10 Screening of 200 (OPTIONS) response content

When an OPTIONS request is received, the AS shall:

- 1) identify the served user as specified in 3GPP TS 24.229 [1];
- 2) identify the originator of the OPTIONS request by the URI present in the P-Asserted-Identity header field of the request if present and otherwise by the URI present in the From header field of the request; and
- 3) if screening is required, by configuration data, act as a routing B2BUA according to 3GPP TS 24.229 [1]. Then:
 - Upon receipt of any response different from 200 (OK) response, the AS shall generate the same response according to the procedures for a routing B2BUA specified in subclause 5.7.5.1 of 3GPP TS 24.229 [1].
 - Upon receipt of a 200 (OK) response, the AS shall generate a 200 (OK) response according to the procedures for a routing B2BUA specified in subclause 5.7.5.1 of 3GPP TS 24.229 [1] without the information for which screening is required by the configuration data.

NOTE 1: The configuration data can be set the user or by the operator and can include the identity of the served user and/or the identity of the originator, How the user can configure the screening of its destined OPTIONS requests is out of the scope of this specification. Such configuration can for example be performed through a web portal.

NOTE 2: Screening of UE capabilities from the 200 (OK) response to an OPTIONS request can cause the originator not to attempt a session using the screened capabilities.

4.7.2.10 Action at the terminating UE

Certain services require the usage of the Alert-Info header field and Call-Info header field according to procedures specified by IETF RFC 3261 [4].

If the UE receives a re-INVITE request containing no SDP offer, the UE shall send a 200 (OK) response containing an SDP offer according to 3GPP TS 24.229 [1] indicating the directionality used by UE as

- "sendonly" if the re-INVITE request is received on a dialog where the associated communication session has been put on hold by the user; and
- "sendrecv" otherwise.

4.8 Interactions with other networks

4.8.1 Void

4.8.2 Void

4.8.3 Void

4.9 Signalling flows

Signalling flows are documented in annexes A and B.

4.10 Parameter values (timers)

No specific timers are needed.

Annex A (informative): Signalling flows for announcements

This annex shows some example signalling flows for the procedures described in subclause 4.1.

These signalling flows are simplified in that, for in-band announcements, they do not show the AS to MRFC interactions.

A.1 Providing announcements to a user during the establishment of a communication session

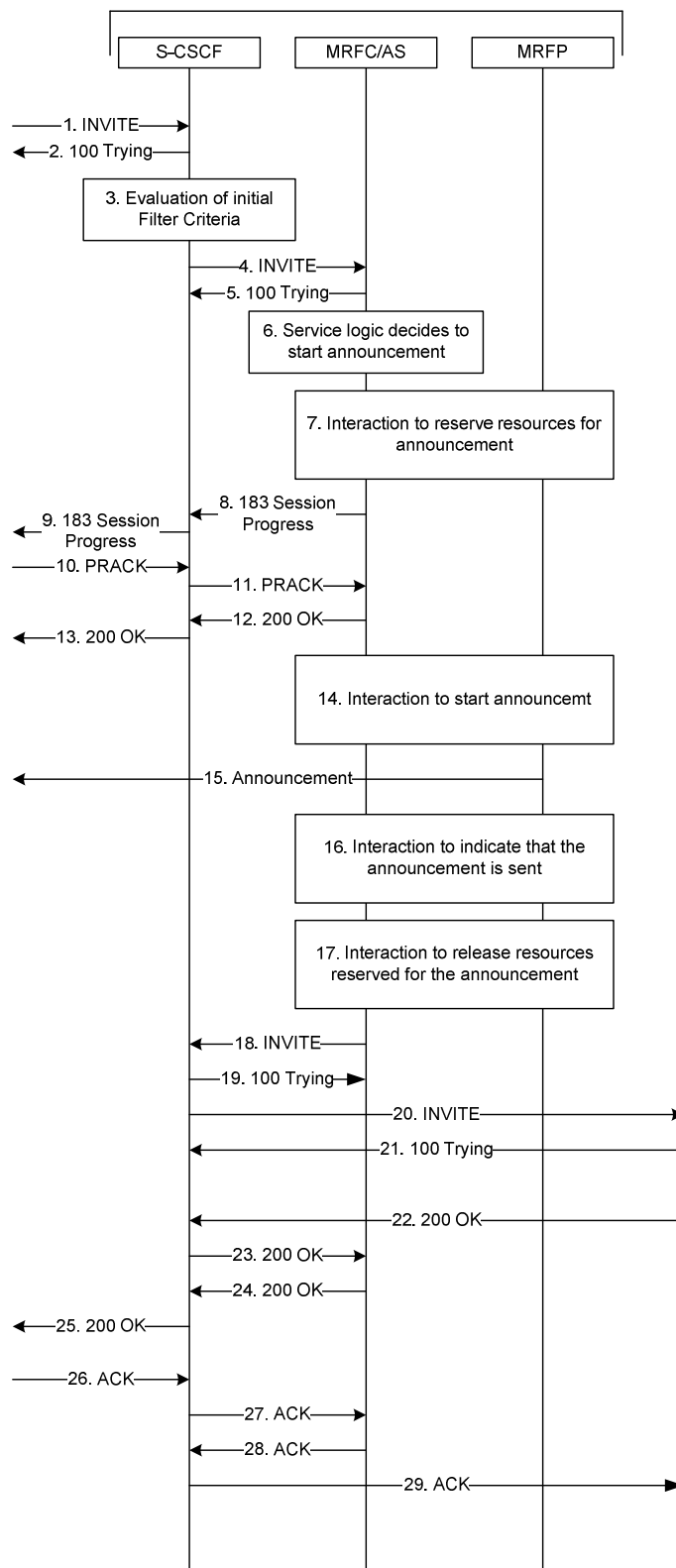
A.1.1 Providing in-band announcement

This subclause shows an example signalling flow of how an AS can send an announcement to the calling user during the establishment of a communication.

Separate dialogs are established between the origination UE and the AS controlling the announcement, and the originating UE and the terminating UE. It is allowed that a different SDP answer is sent in the 200 (OK) response from the terminating UE than the SDP answer that was previously sent from the AS in the 183 (Session progress) response.

The AS can e.g. be the AS serving the calling party or the AS serving a called party and may apply for example when a communication is going to be diverted and the AS serving the diverting user inform the calling party that the communication is going to be diverted.

Figure A.1 shows the signalling flow for the scenario.



NOTE: The called party can return provisional responses to the INVITE request. However, for simplicity those responses are left out.

Figure A.1: Announcement started during the establishment of a communication

The calling party initiates a communication by means of an INVITE request. The INVITE request is forwarded toward the called party.

Along the signalling path, created by the INVITE request, some service logic in an Application Server (AS) wants to send an announcement towards the calling party.

The flow is based on the assumptions that the Supported header field includes the option-tag "100rel".

The steps of the signalling flow are as follows:

- 1) S-CSCF receives an INVITE request.
- 2) S-CSCF sends the 100 (Trying) response towards to sender of the INVITE request.
- 3) S-CSCF evaluates the initial Filter Criteria.
- 4) S-CSCF sends the INVITE request to the AS.
- 5) The AS sends the 100 (Trying) to S-CSCF.
- 6) Service logic in the AS decides to send an announcement to the calling party.
- 7) The MRFC interacts with the MRFP in order to reserve resources for the announcement. As part of the interaction with MRFP the AS receives the necessary media parameters e.g. IP address and port numbers and provide the IP address and port number for the calling party to the MRFP.
- 8) The AS sends a 183 (Session progress) response to S-CSCF. The response includes:
 - a) an answer to the SDP received in the INVITE request;
 - b) a P-Early-Media header field set to "sendonly"; and
 - c) the Require header field set to "100rel".
- 9) S-CSCF sends the 183 (Session progress) response towards the calling party.
- 10) S-CSCF receives a PRACK request.
- 11) S-CSCF sends the PRACK request to the AS.
- 12) The AS sends a 200 (OK) to the PRACK request to S-CSCF.
- 13) S-CSCF sends the 200 (OK) towards the calling party.
- 14) The MRFC interacts with the MRFP in order to start the announcement.
- 15) The MRFP sends the announcement towards the calling party.
- 16) The complete announcement is sent and the MRFP interacts with the AS/MRFC in order to inform that the announcement is terminated.
- 17) The MRFC interacts with the MRFP in order to release the resources used for the announcement.
- 18) The AS sends the INVITE request towards the called party. The INVITE request contains the same information as the INVITE request received in step 4 with the modification done by AS according to rules and procedures of 3GPP TS 24.229 [1].
- 19) S-CSCF sends the 100 (Trying) response to the AS.
- 20) S-CSCF sends the INVITE request towards the called party.
- 21) S-CSCF receives a 100 (Trying) response.
- 22) S-CSCF receives a 200 (OK) response to the INVITE request.
- 23) S-CSCF sends the 200 (OK) response to the INVITE request to the AS.
- 24) The AS sends the 200 (OK) response to the INVITE request to the S-CSCF.
- 25) S-CSCF sends the 200 (OK) response to the INVITE request towards the calling party.

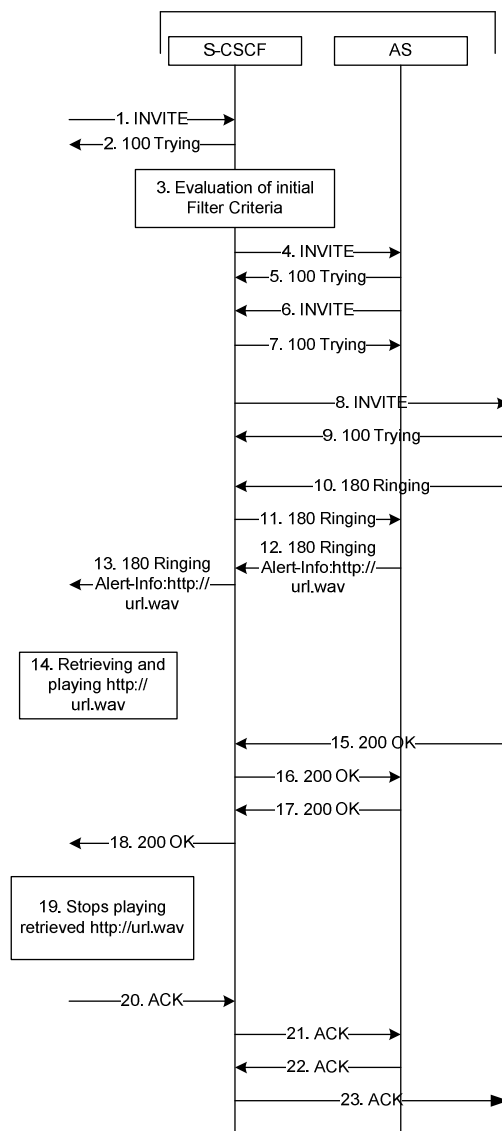
- 26) S-CSCF receives an ACK request.
- 27) S-CSCF sends the ACK request to the AS.
- 28) The AS sends the ACK request to S-CSCF.
- 29) S-CSCF sends the ACK towards the called party.

When the UE of the calling party receives the 200 (OK) response to the INVITE request the UE can regard the early dialog created for the announcement between the UE and the AS terminated.

A.1.2 Including Alert-Info header field in the 180 (Ringing) response

IETF RFC 3261 [4] specifies the Alert-Info header field as a means to indicate a source of media to play an alternative ring tone by an originating endpoint.

An example of this mechanism is shown in figure A.2.



NOTE: In the figure the SDP signalling details to establish media are not shown for simplicity.

Figure A.2: Alert-Info header field in the 180 (Ringing) response to indicate an alternative ring tone

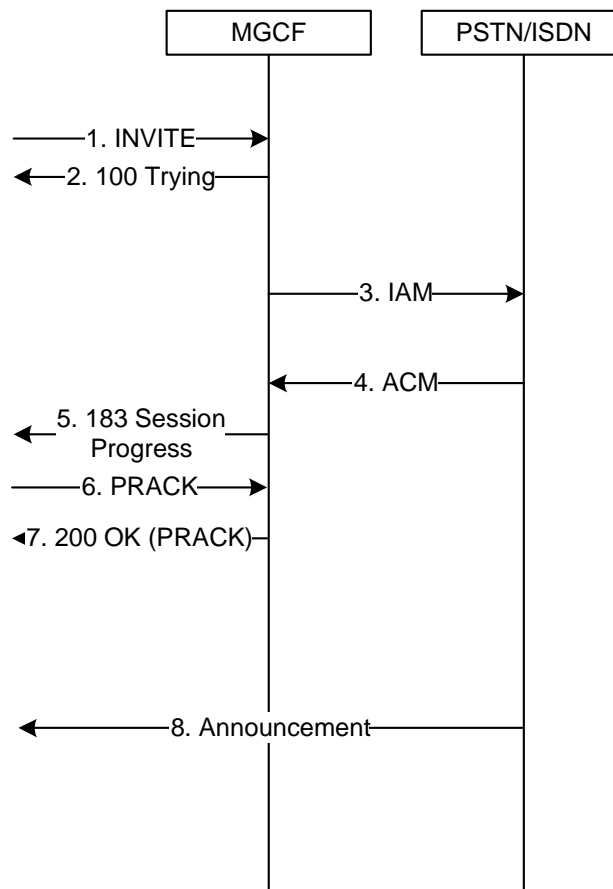
The steps of the flow are as follows:

- 1) S-CSCF receives an INVITE request from the originating user. The originating user may be a user served by this S-CSCF, a user served by another S-CSCF or a user connected to PSTN/ISDN via a MGCF.
 - 2) S-CSCF sends a 100 (Trying) response.
 - 3) S-CSCF evaluates the Initial Filter Criteria.
 - 4) S-CSCF sends the INVITE request to the AS.
 - 5) The AS sends a 100 (Trying) response to S-CSCF.
 - 6) The AS sends the INVITE request to S-CSCF.
 - 7) S-CSCF sends the 100 (Trying) response to the AS.
 - 8) S-CSCF sends the INVITE request towards the called party. The called party may be a user served by another S-CSCF or a user connected to PSTN/ISDN via a MGCF.
 - 9) S-CSCF receives a 100 (Trying) response.
 - 10) S-CSCF receives a 180 (Ringing) response.
 - 11) S-CSCF sends the 180 (Ringing) response to the AS.
 - 12) The AS inserts a valid Alert-Info header field in the 180 (Ringing) including a URL to a media file containing the appropriate tone and sends the 180 (Ringing) response to S-CSCF.
- EXAMPLE: This file <http://operator.net/tone.wav>, in the picture abbreviated to <http://url.wav> is played at the originating UE (step 14).
- 13) S-CSCF sends the 180 (Ringing) response towards the originating user.
 - 14) The <http://url.wav> (for example <http://operator.net/tone.wav>) is retrieved and played at the originating user.
 - 15-18) S-CSCF receives a 200 (OK) response to the INVITE request and forwards it to the originating user via the AS.
 - 19) The originating user stops playing the tone.
 - 20-23) S-CSCF receives an ACK request and forwards it towards the called party via the AS.

A.1.3 Announcements provided by the PSTN/ISDN

This subclause shows the signalling flow for a scenario where a user connected to the IP network establish a communication with a user connected to the PSTN/ISDN. During the establishment of the communication the PSTN/ISDN provides an announcement e.g. "The communication is forwarded" or "The user is not reachable".

Figure A.3 shows the signalling flow for the scenario.



NOTE: The flow assumes the use of the option-tag "100rel" defined in RFC 3262 [5] other scenarios may also apply. T-MGF is left out of the figure for simplicity.

Figure A.3: Announcement provided by PSTN/ISDN during the establishment of a communication

The steps of the flow are as follows:

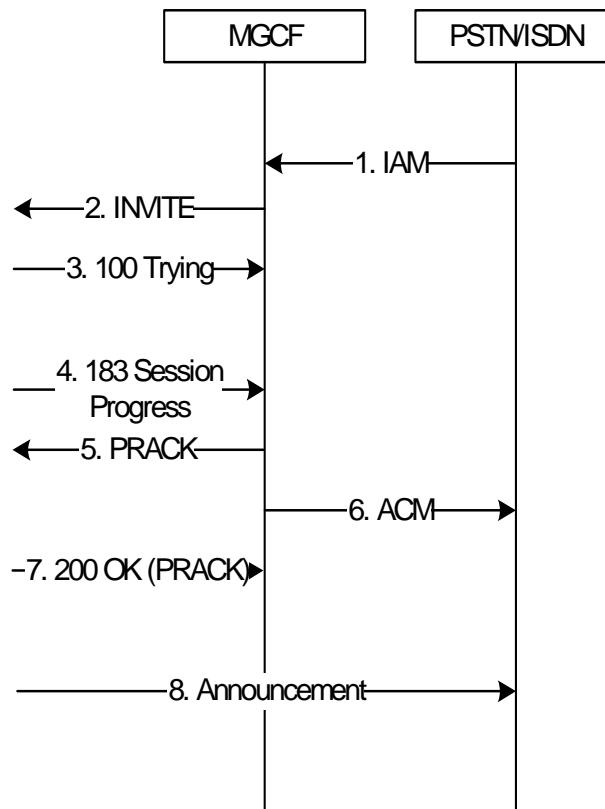
- 1) The MGCF receives an INVITE request from the IP network. The request includes an SDP offer.
- 2) The MGCF sends a 100 (Trying) response to the IP network.
- 3) The MGCF sends an IAM towards PSTN.
- 4) The MGCF receives an early ACM from the PSTN/ISDN with an indication that "In-band information may be available".
- 5) The MGCF sends a 183 (Session Progress) response to the IP network. The response includes:
 - a) the answer to the SDP offer received in the INVITE request;
 - b) A P-Early-Media header field set to "sendonly"; and
 - c) The option-tag "100rel" in the Require header.
- 6) The MGCF receives the PRACK request.
- 7) The MGCF sends a 200 (OK) response to the PRACK request.
- 8) The T-MGF sends the in-band announcement received from the PSTN/ISDN to the IP network.

Depending on the reason for the announcement the establishment of the communication continues or the establishment of the communication is aborted.

A.1.4 Announcement provided towards a user connected to the PSTN/ISDN

This subclause shows an example signalling flow for a scenario where a user in PSTN/ISDN establishes a communication with a user connected to IMS. During the establishment an AS in the IP network provides an announcement, e.g. "The communication is forwarded" or "The user is not reachable".

Figure A.4 shows the signalling flow for the scenario.



NOTE: The flow assumes the use of the option-tag "100rel" defined in RFC 3262 [5] other scenarios can also apply. T-MGF is left out of the figure for simplicity.

Figure A.4: Announcement provided towards a user connected to PSTN/ISDN during establishment of a communication

The steps of the flow are as follows:

- 1) The MGCF receives an IAM from the PSTN/ISDN.
- 2) The MGCF sends an INVITE request to the IP network. The request includes a SDP offer.
- 3) The MGCF receives a 100 (Trying) response from the IP network.
- 4) The MGCF receives a 183 (Session Progress) response from the IP network. The response includes:
 - a) the answer to the SDP offer sent in the INVITE request;
 - b) a P-Early-Media header field set to "sendonly"; and
 - c) the option-tag "100rel" in the Require header field.
- 5) The MGCF sends a PRACK request towards the IP network.
- 6) The MGCF sends an early ACM to the PSTN/ISDN. The early ACM contains the "in-band information may be available" indication.

- 7) The MGCF receives a 200 (OK) response to the PRACK request.
- 8) The T-MGF receives the in-band announcement from the IP network and forwards the announcement to the PSTN/ISDN network.

Depending on the reason for the announcement the establishment of the communication continues or the establishment of the communication is aborted.

A.1.5 Providing in-band announcement after a reliable provisional response has been received by the terminating UE

This subclause shows an example signalling flow of how an AS can send an announcement to the calling user during the establishment of a communication, in the case the terminating UE has already sent SDP within a reliable provisional response.

Separate dialogs are established between the origination UE and the AS controlling the announcement, in accordance with annex A.1.1.

Figure A.5 shows the signalling flow for the scenario.

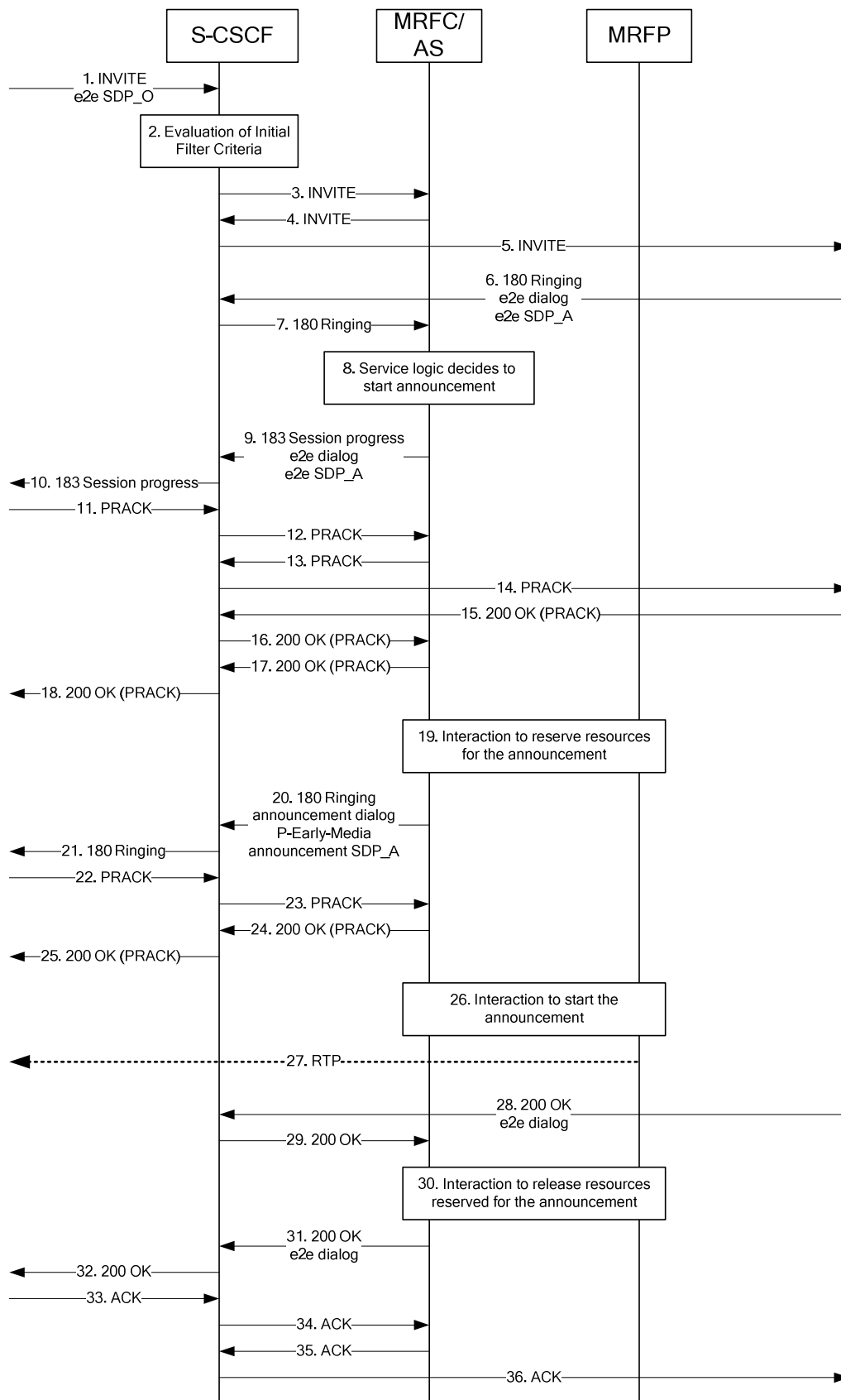


Figure A.5: Originating announcement during communication establishment considering terminating early media

The calling party initiates a communication by means of an INVITE request. The INVITE request is forwarded toward the called party.

Along the signalling path, created by the INVITE request, some service logic in an AS wants to send an announcement towards the calling party.

The flow is based on the assumptions that the Supported header field includes the option-tag "100rel".

The steps of the signalling flow are as follows:

- 1) S-CSCF receives an INVITE request with the initial SDP offer
- 2) S-CSCF evaluates the initial Filter Criteria
- 3) S-CSCF sends the INVITE request to the AS
- 4), 5) the AS forwards the INVITE to the terminating UE
- 6), 7) the terminating UE answers with a reliable 180 (Ringing) response including a SDP answer.
- 8) based on the received response the service logic in the AS decides to send an announcement to the calling party
- 9), 10) the AS forwards the received SDP answer within a reliable 183 (Session progress) response, in order to establish an e2e dialog between the originating and terminating UEs
- 11) to 18) the originating UE confirms the reliable response to the terminating UE
- 19) The MRFC interacts with the MRFP in order to reserve resources for the announcement. As part of the interaction with MRFP the AS receives the necessary media parameters e.g. IP address and port numbers and provide the IP address and port number for the calling party to the MRFP.
- 20) to 25) the AS sends a reliable 180 (Ringing) response to the originating UE and creates a new dialog between the AS and the originating UE. The response includes:
 - a) an answer to the SDP received in the INVITE request;
 - b) a P-Early-Media header field set to "sendonly"; and
 - c) the Require header field set to "100rel".
- 26) The MRFC interacts with the MRFP in order to start the announcement.
- 27) The MRFP sends the announcement towards the calling party.
- 28), 29) the terminating UE answers with a 200 OK response without SDP
- 30) The MRFC interacts with the MRFP in order to release the resources used for the announcement.
- 31), 32) the AS forwards the 200 OK response in the e2e dialog to the originating UE
- 33) to 36) the communication is confirmed

A.2 Providing announcements to a user during an established communication

The way an announcement is sent to a user during an established communication depends on the scenario and the importance of the announcement.

The following scenarios exist:

- scenario 1: two users are communicating with (at least) one AS in the signalling path (UE AS -UE); or
- scenario 2: two (or more) users communicating with (at least) one AS in the signalling and media path (UE-AS/MRFC-UE); or

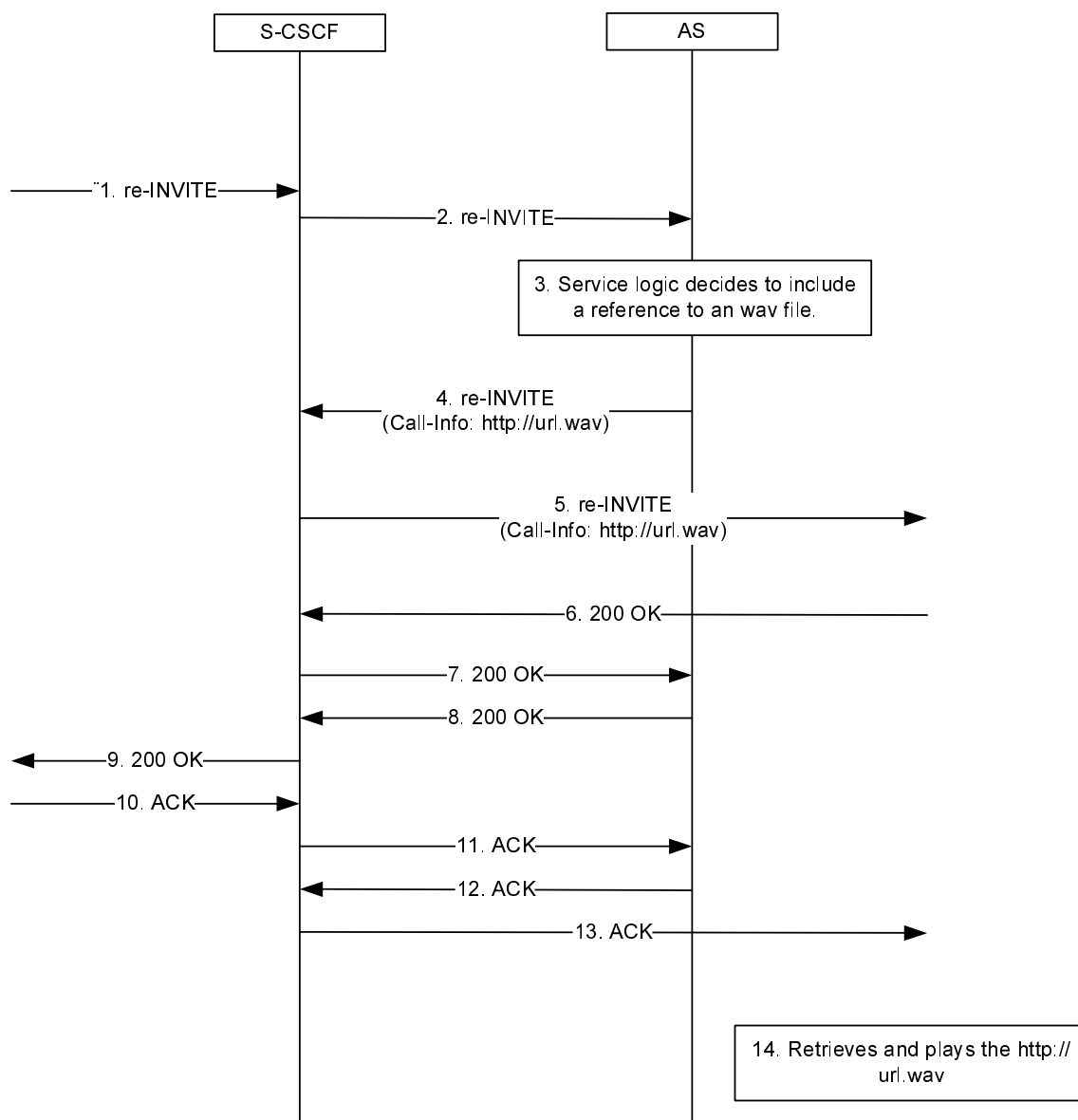
- scenario 3: two users communicate and one of the users is connected to PSTN/ISDN (UE-MGCF). This scenario can be seen as part of basic communication and requires no SIP signalling; or
- scenario 4: two users communicate directly with each other without involving an AS in the signalling path and without involving an AS in the media path (UE-UE). This scenario is out of scope of the present document.

A.2.1 Scenario 1: UE - AS - UE

Two users are communicating with (at least) one AS in the signalling path. In this scenario the AS is connected to the S-CSCF over the ISC interface acting as a SIP proxy or an AS performing 3rd party call control.

IETF RFC 3261 [4] specifies the Call-Info header field as a means to indicate a source of media to be played by the receiving endpoint.

Figure A.6 shows an example of the use of this mechanism in the INVITE request.



NOTE: Some signalling details are left out of the figure for simplicity.

Figure A.6: Call-Info header field in a re-INVITE request to indicate media

A user wants to place a communication session on hold and sends a re-INVITE request towards the remote user involved in the communication.

The steps of the flow are as follows:

- 1) S-CSCF receives a reINVITE request from a user. The user can be a user served by this S-CSCF, a user served by another S-CSCF or a user connected to the PSTN via MGCF.
- 2) S-CSCF sends the reINVITE request along the signalling path to the AS using the route set received in the reINVITE request.
- 3) Service logic in the AS decides to include a reference to a wav file with an announcement or music.
- 4) The AS sends the reINVITE request to the S-CSCF. Including in the Call-Info header a URL to a media file containing the appropriate announcement or music, for example <http://operator.net/announcement.wav> (in the picture abbreviated to <http://url.wav>).
- 5) The S-CSCF sends the reINVITE request along the signalling path towards the remote user. The remote user can be a user served by this S-CSCF, a user served by another S-CSCF or a user connected to the PSTN via MGCF.
- 6-9) The 200 (OK) response from the remote user is forwarded via the S-CSCF and the AS towards the originating user.
- 10-13) The ACK request from the originating user is forwarded via the S-CSCF and the AS towards the remote user.
- 14) The <http://url.wav> file is retrieved and played to the user. In the case the user is connected to the PSTN via a MGCF, the T-MGF retrieves and plays the announcement towards the user. In case the user is connected to IMS the UE retrieves and plays the announcement.

A.2.2 Scenario 2: UE - AS/MRFC/MRFP - UE

This subclause describes the scenario when two (or more) users are communicating with (at least) one AS controlling the media path. The MRFP is in the media path. In this scenario the AS acts as a B2BUA.

Figure A.7 shows the signalling flow for the scenario.

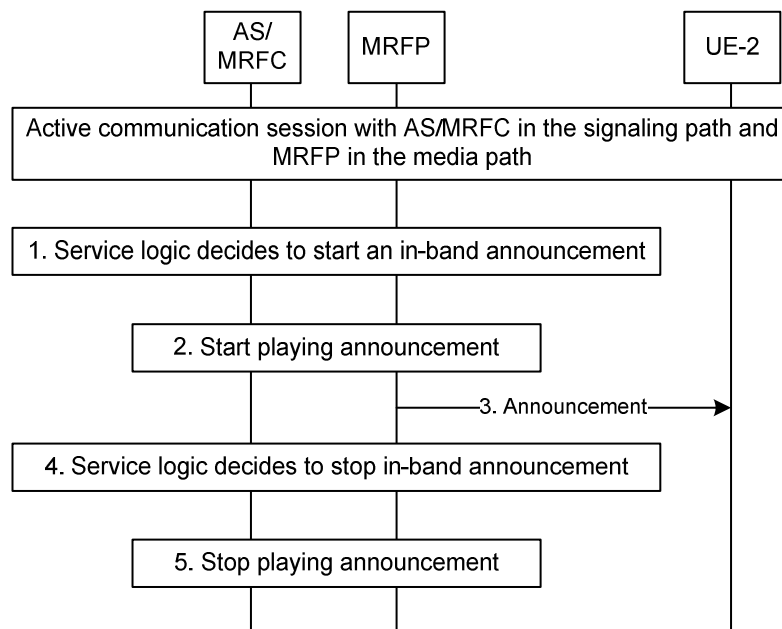


Figure A.7: In-band announcement during an established communication

An AS, acting as a B2BUA, is involved in a communication session. The AS controls the media path via a co-located MRFC controlling a MRFP.

The steps of the flow are as follows:

- 1) Service logic in the AS decides to start an in-band announcement towards a user e.g. "Music on hold".

- 2) The MRFC co-located with the AS interacts with the MRFP in order to start the announcement.
- 3) The MRFP sends the announcement towards the remote user.
- 4) The MRFC co-located with the AS interacts with the MRFP to stop the announcement.
- 5) MRFP stops the announcement provided to UE-2.

A.3 Communication request rejected

Service logic in an AS, e.g. the ACR service, can decide to reject a communication request and provide an announcement to explain the reason for the rejection to the originating user. The AS can:

- 1) Send the announcement as in-band information.
- 2) Include a reference to the announcement in a 3xx, 4xx, 5xx and 6xx response.

A.3.1 Sending the announcement as in-band information

The network can generate announcement using one of the following procedures:

- 1) using early media i.e. the AS establish an early session and uses that early session to send the in-band announcement; or
- 2) using an established session i.e. the AS accepts the INVITE request and uses the established session to send the in-band announcement.

A.3.1.1 Using early media

This subclause explains how an AS can use an early media session to send the in-band announcement and when the announcement is sent reject the communication request with an appropriate reject code.

Figure A.8 shows the signalling flow for the scenario.

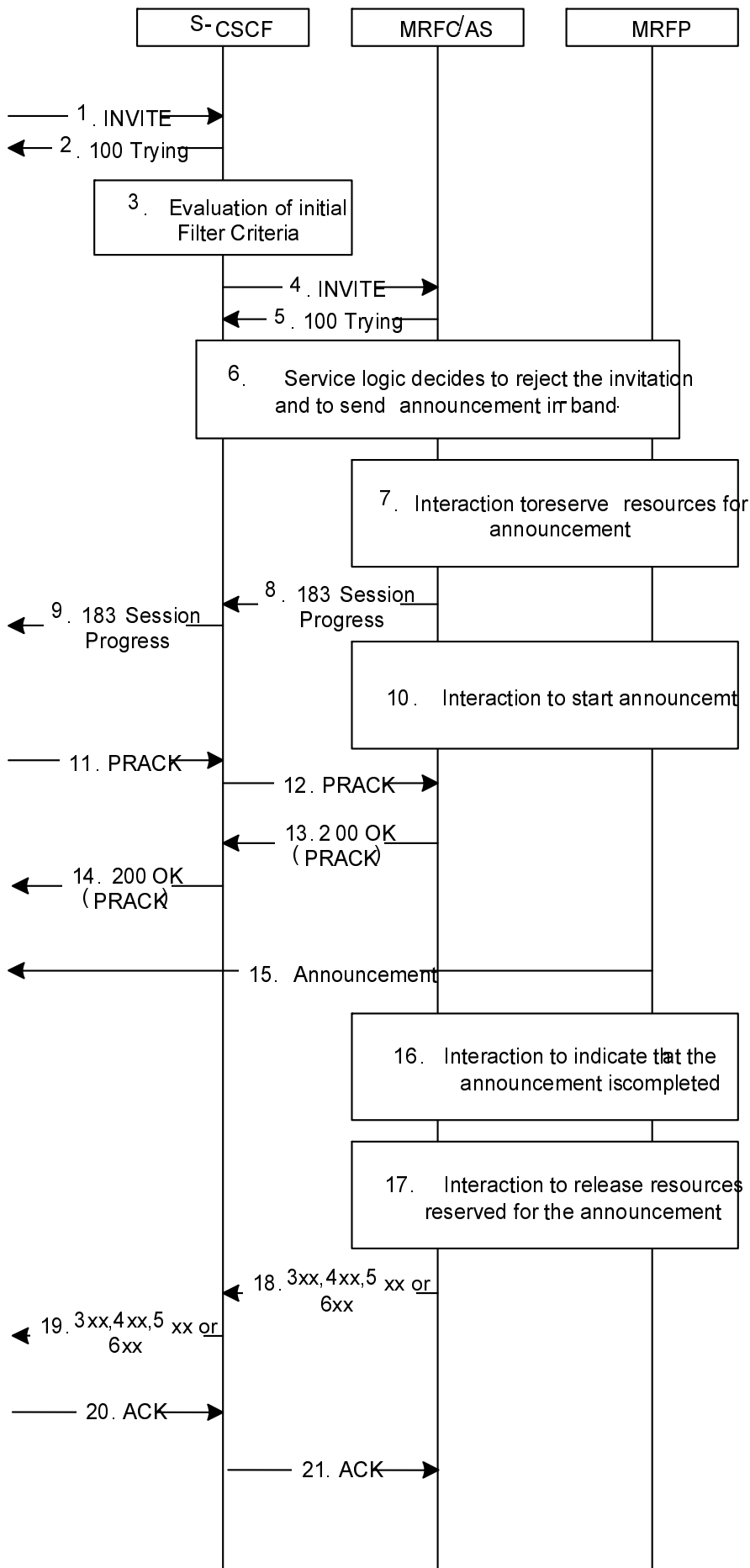


Figure A.8: Using early media to send in-band announcement

The originating user initiates communication by means of an INVITE request. Along the path towards the terminating user an AS determines that the INVITE request cannot be forwarded to the terminating user. The steps of the flow are as follows:

- 1) S-CSCF receives an INVITE request from the originating user. The originating user can be a user served by this S-CSCF, a user served by another S-CSCF or a user connected to PSTN/ISDN via a MGCF.
- 2) S-CSCF sends a 100 (Trying) response.
- 3) S-CSCF evaluates the Initial Filter Criteria.
- 4) S-CSCF sends the INVITE request to the AS.
- 5) The AS sends a 100 (Trying) response to S-CSCF.
- 6) Service logic in the AS decides to reject the communication request and to send an announcement in-band in order to give a detailed reason to the originating user.
- 7) The MRFC collocated with the AS interacts with the MRFP and reserves resources for the announcement.
- 8) The AS sends a 183 (Session progress) response to S-CSCF. The response includes:
 - a) the Require header field with the option-tag "100rel";
 - b) an answer to the SDP received in the INVITE request; and
 - c) an answer to the SDP received in the INVITE request.
- 9) S-CSCF sends the 183 (Session Progress) response towards the originating user.
- 10) The MRFC collocated with the AS interacts with the MRFP in order to start the announcement.
- 11) S-CSCF receives a PRACK request from the originating user.
- 12) S-CSCF sends the PRACK request to the AS.
- 13) The AS sends the 200 (OK) response to the PRACK request to S-CSCF.
- 14) S-CSCF sends the 200 (OK) response to the PRACK request to the originating user.
- 15) MRFP sends the announcement towards the UE.
- 16) The MRFP interacts with the MRFC collocated with the AS to indicate that the announcement is sent.
- 17) The MRFC collocated with the AS interacts with the MRFP in order to release resources reserved for the announcement.
- 18) The AS sends a 3xx, 4xx, 5xx or 6xx response to the INVITE request to S-CSCF.
- 19) S-CSCF sends a 3xx, 4xx, 5xx or 6xx response to the INVITE request to the originating user.
- 20) S-CSCF receives an ACK request from the originating user.
- 21) S-CSCF sends an ACK request to the AS.

A.3.1.2 Using an established session

This subclause explains how an AS can use an established session to send the in-band announcement and when the announcement is sent, release the communication and include an appropriate reject code in the BYE request.

Figure A.9 shows the signalling flow for the scenario.

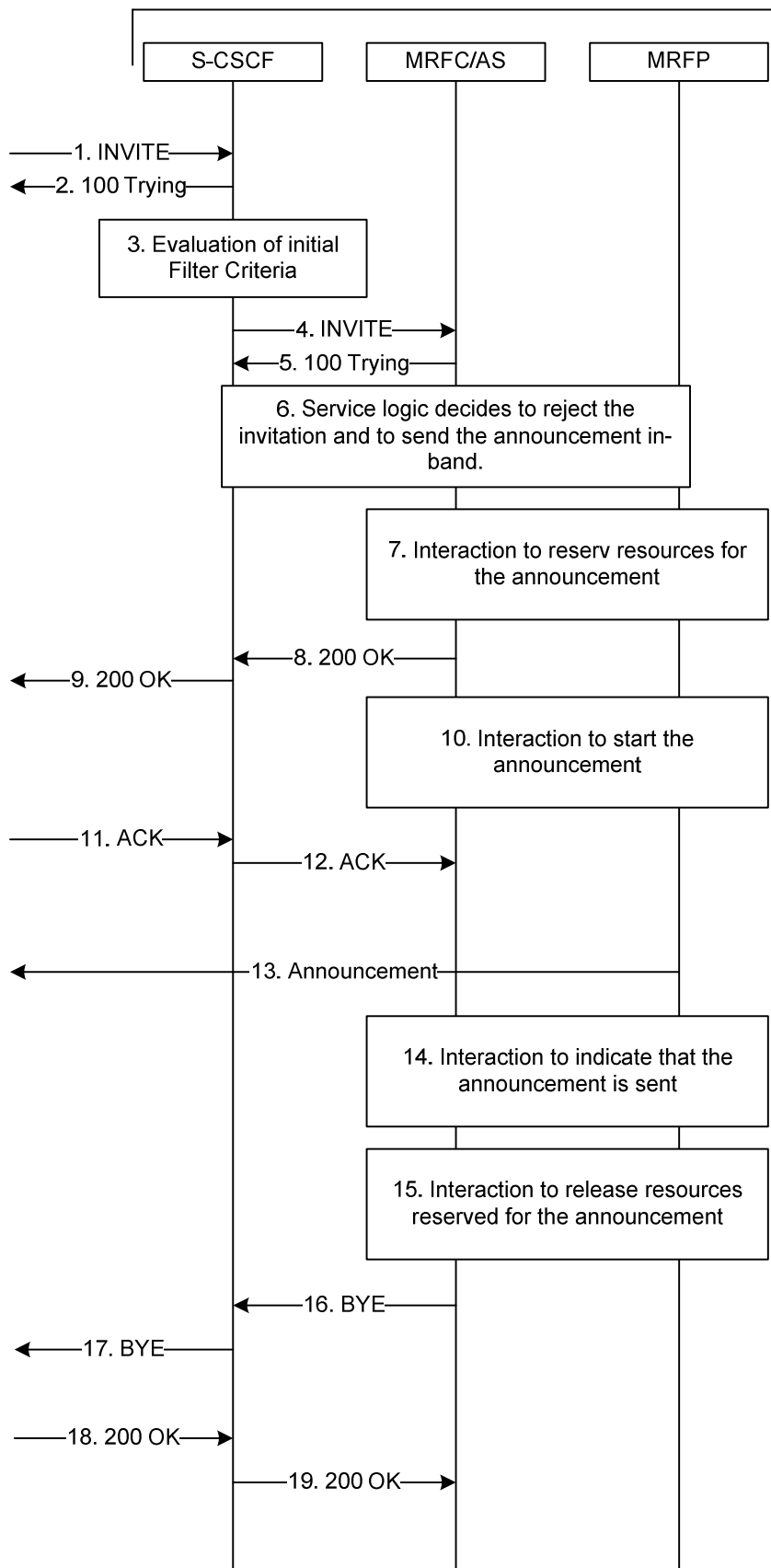


Figure A.9: In-band information generated by network when an invitation to a communication is rejected

The originating user initiates communication by means of an INVITE request. Along the path towards the terminating user an AS determines that the INVITE request cannot be forwarded to the terminating user.

The steps are as follows:

- 1) S-CSCF receives an INVITE request from the originating user. The originating user can be a user served by this S-CSCF, a user served by another S-CSCF or a user connected to PSTN/ISDN via a MGCF.
- 2) S-CSCF sends a 100 (Trying) response.
- 3) S-CSCF evaluates the Initial Filter Criteria.
- 4) S-CSCF sends the INVITE request to the AS.
- 5) The AS sends a 100 (Trying) response to S-CSCF.
- 6) The AS decides to reject the communication request and to send an announcement in-band in order to give a detailed reason to the originating user.
- 7) The MRFC collocated with the AS interact with the MRFP and reserves resources for the announcement.
- 8) The AS sends a 200 (OK) response to the INVITE request to S-CSCF.
- 9) S-CSCF sends the 200 (OK) response to the INVITE request towards the originating user.
- 10) The MRFC collocated with the AS interact with the MRFP in order to start the announcement.
- 11) S-CSCF receives an ACK request from the originating user.
- 12) S-CSCF sends the ACK request to the AS.
- 13) MRFP sends the announcement towards the originating user.
- 14) The MRFP interacts with the MRFC collocated with the AS to indicate that the announcement is sent.
- 15) The MRFC collocated with the AS interact with the MRFP in order to release resources reserved for the announcement.
- 16) The AS sends a BYE request to S-CSCF. The BYE request can include an appropriate reject reason.
- 17) S-CSCF sends the BYE request towards the originating user.
- 18) S-CSCF receives a 200 (OK) response to the BYE request from the originating user.
- 19) S-CSCF sends the 200 (OK) response to the BYE request to the AS.

A.3.2 Including an Error-Info header field in a 3xx, 4xx, 5xx and 6xx response

This subclause explains how an AS can include a reference to an announcement stored in the network.

IETF defines an Error-Info header field for use in 3xx, 4xx, 5xx and 6xx responses to the INVITE request. The Error-Info header field transports a reference to a file e.g. a file containing an announcement.

When the originating UE receives the reference the UE retrieves the announcement and plays it for the user.

Figure A.10 shows the message flow for the scenario.

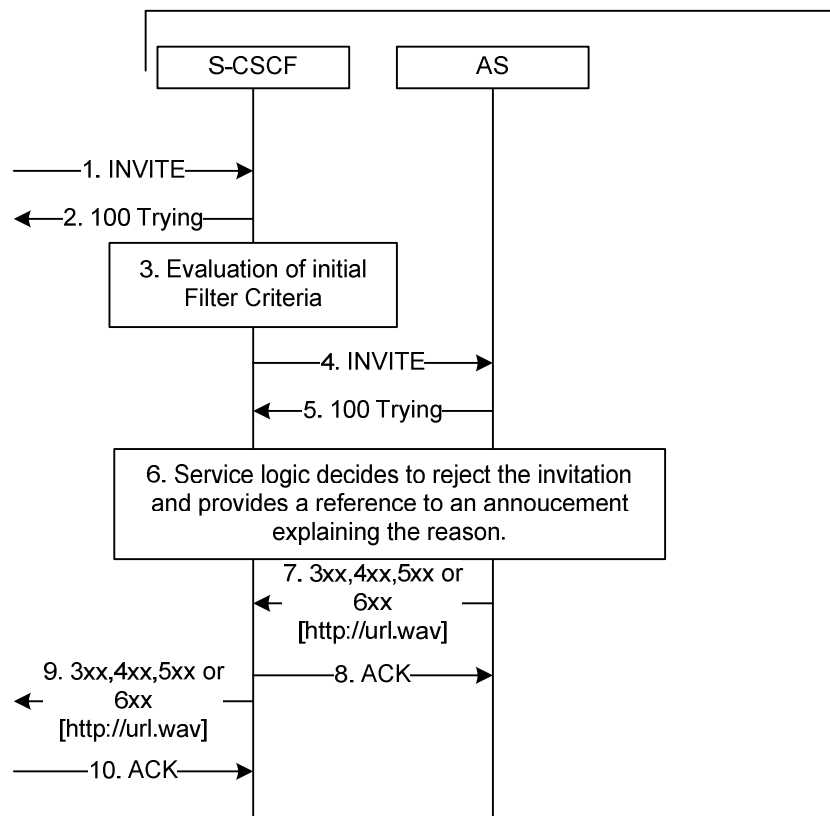


Figure A.10: Error-Info header in 3xx, 4xx, 5xx and 6xx responses

The originating user initiates communication by means of an INVITE request. Along the path towards the terminating user an AS determines that the INVITE request cannot be forwarded to the terminating user.

The steps are as follows:

- 1) S-CSCF receives an INVITE request from the originating user (in the case the AS is an O-AS) or the originating network (in the case the AS is a T-AS).
- 2) S-CSCF sends a 100 (Trying) response.
- 3) S-CSCF evaluates the Initial Filter Criteria.
- 4) S-CSCF sends the INVITE request to the AS.
- 5) The AS sends a 100 (Trying) response to S-CSCF.

6) The AS decides to reject the invitation to communication and to provide a reference to an announcement explaining the reason in more detail.

7) The AS sends a 3xx, 4xx, 5xx or 6xx response to S-CSCF. The application server inserts a valid Error-Info header field in either a 3xx, 4xx, 5xx or 6xx response to the INVITE request, including a URL to a media file containing the appropriate tone, announcement or music.

EXAMPLE: <http://operator.net/announcement.wav>, in the picture abbreviated to <http://url.wav>, is played at the originating UE (after step 10).

8) S-CSCF sends the ACK request to the AS.

9) S-CSCF sends the 3xx, 4xx, 5xx or 6xx response towards the originating user.

10) S-CSCF receives the ACK request.

A.3.3 Announcements provided by the PSTN/ISDN

The signalling flow for this scenario is the same as the signalling flow example given in subclause A.1.3.

A.3.4 Announcement provided to a user connected to the PSTN/ISDN

The signalling flow for this scenario is the same as the signalling flow example given in subclause A.1.4.

A.4 Providing announcements to a user during the release of a communication session

The way an announcement is sent to a user during the release of a communication depends on the scenario.

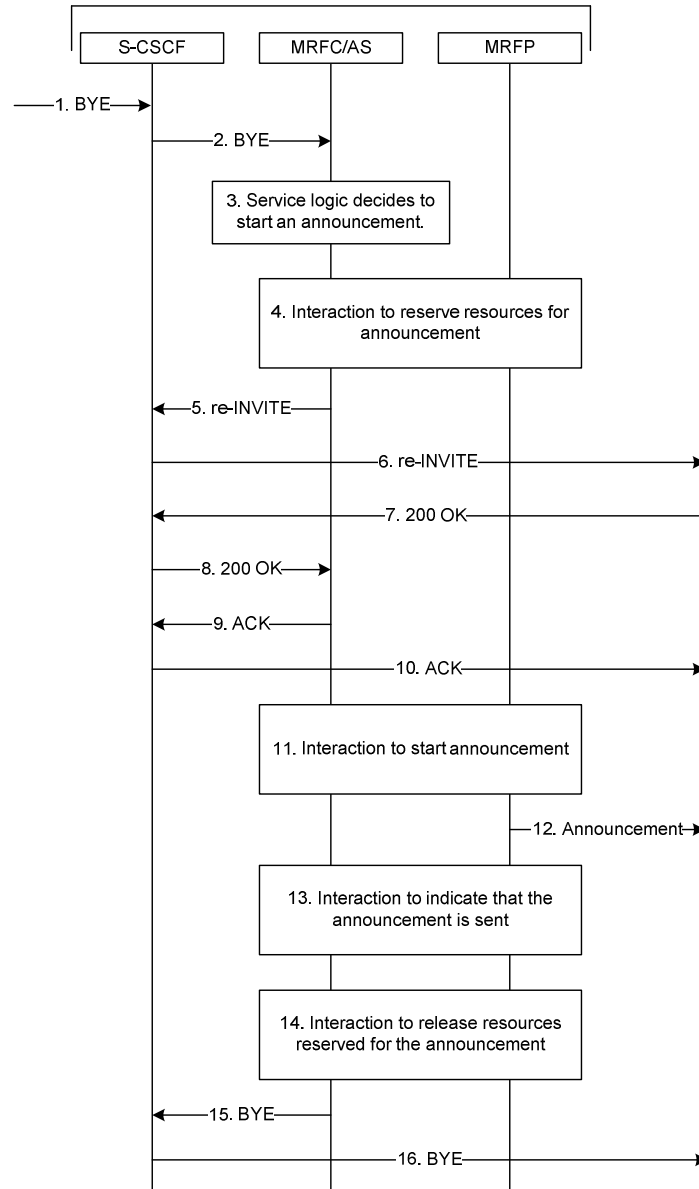
The following scenarios exist:

- scenario 1: two users are communicating with (at least) one AS in the signalling path (UE-AS-UE); or
- scenario 2: two (or more) users communicating with (at least) one AS in the signalling path and MRFP in the media path (UE-AS/MRFC/MRFP-UE).

A.4.1 Scenario 1: UE - AS - UE

Two users are communicating with (at least) one AS in the signalling path.

Figure A.11 shows the signalling flow for the scenario.



NOTE: The party still in the communication can return provisional responses to the INVITE request. However, for simplicity those responses are left out.

Figure A.11: Play announcement using new media during the release of a communication

The steps of the signalling flow are as follows:

- 1) S-CSCF receives a BYE request.
- 2) S-CSCF sends the BYE request to the AS.
- 3) Service logic in the AS decides to send an announcement to the party still in the communication.
- 4) The MRFC interacts with the MRFP in order to reserve resources for the announcement. As part of the interaction with MRFP, the AS retrieves the media parameters from MRFP e.g. IP address and port numbers, and provide the IP address and port numbers of media that original dialog used to the party still in the communication.

- 5) The AS sends a reINVITE request to S-CSCF.
- 6) S-CSCF sends the reINVITE request towards the party still in the communication.
- 7) S-CSCF receives a 200 (OK) response.
- 8) S-CSCF sends the 200 (OK) response to the AS.
- 9) The AS sends an ACK request to S-CSCF.
- 10) S-CSCF sends the ACK request to the party still in the communication.
- 11) The MRFC interacts with the MRFP in order to start the alternative ring tone.
- 12) The MRFP sends the announcement towards the party still in the communication.
- 13) The complete announcement is sent and the MRFP interacts with the AS/MRFC in order to inform that the announcement is terminated.
- 14) The MRFC interacts with the MRFP in order to release the resources used for the announcement.
- 15) The AS sends the BYE request to S-CSCF. The BYE request contains the same information as the BYE request received in step 2 with the modification done by AS according to rules and procedures of 3GPP TS 24.229 [1].
- 16) S-CSCF sends the BYE request towards the party still in the communication.

A.4.2 Scenario 2: UE - AS/MRFC/MRFP - UE

This subclause describes the scenario when two (or more) users are communicating with (at least) one AS controlling the media path. The MRFP is in the media path. In this scenario the AS acts as a B2BUA.

Figure A.12 shows the signalling flow for the scenario.

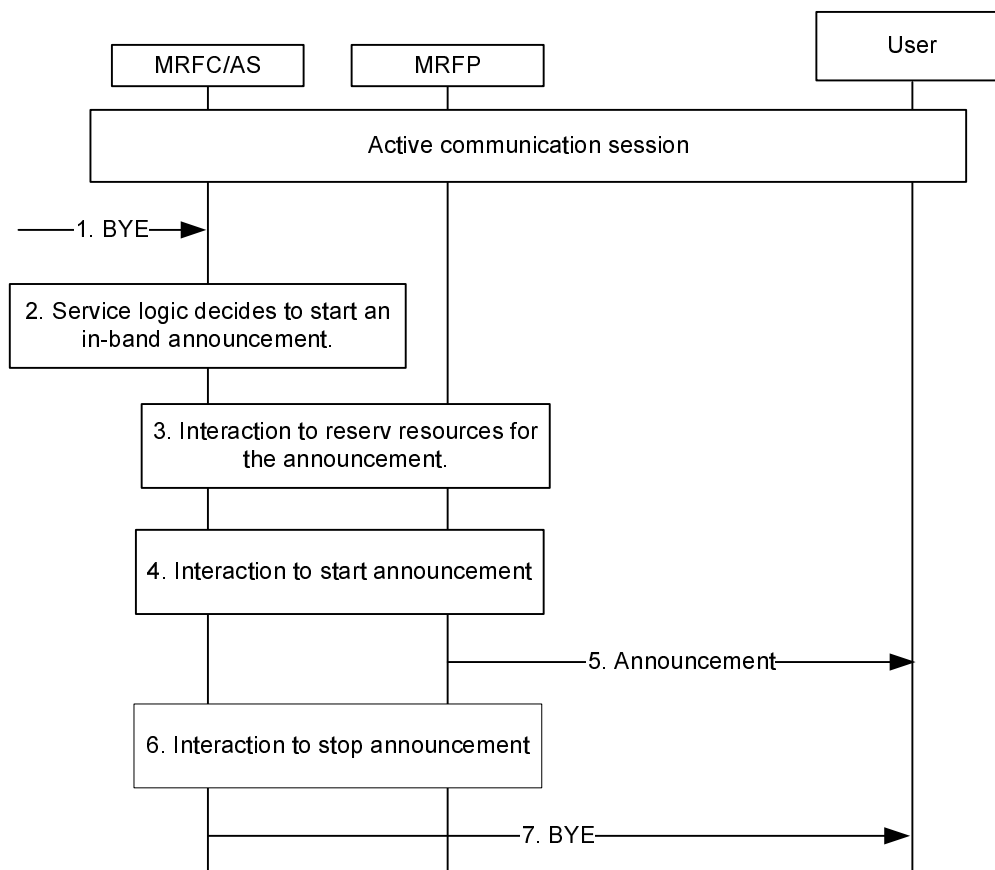


Figure A.12: Play announcement using exist media during the release of a communication

The steps of the signalling flow are as follows:

- 1) The AS receives a BYE request.
- 2) Service logic in the AS decides to start an in-band announcement towards a user.
- 3) The AS using the co-located MRFC interacts with the MRP in order to reserve resources for the announcement.
- 4) The MRFC co-located with the AS interacts with the MRFP in order to start the announcement.
- 5) The MRFP sends the announcement towards the remote user.
- 6) The MRFC co-located with the AS interacts with the MRFP to stop the announcement.

The AS sends the BYE request towards the remote user. The BYE request contains the same information as the BYE request received in step 1 with the modification done by AS according to rules and procedures of 3GPP TS 24.229 [1].

A.5 Providing announcements to a terminating user just after the call is answered and before establishing direct communication session between end users

A.5.1 Switch media path from (UE-AS/MRFC/MRFP) to (UE-UE) within the same dialog

In order to provide announcement to the terminating UE when the called user answers the call, the AS associated with MRFC providing announcement, acting as a B2BUA, will intercept all SIP messages exchanged between the end UEs prior to establishing media stream between end UEs. The AS will replace SDP indicated by the originating UE to establish media stream with MRFC and provide announcement from MRFC once the call is answered, which later on can be replaced with SDP indicated originally by the originating UE.

Figure A.13 shows the message flow for the scenario.

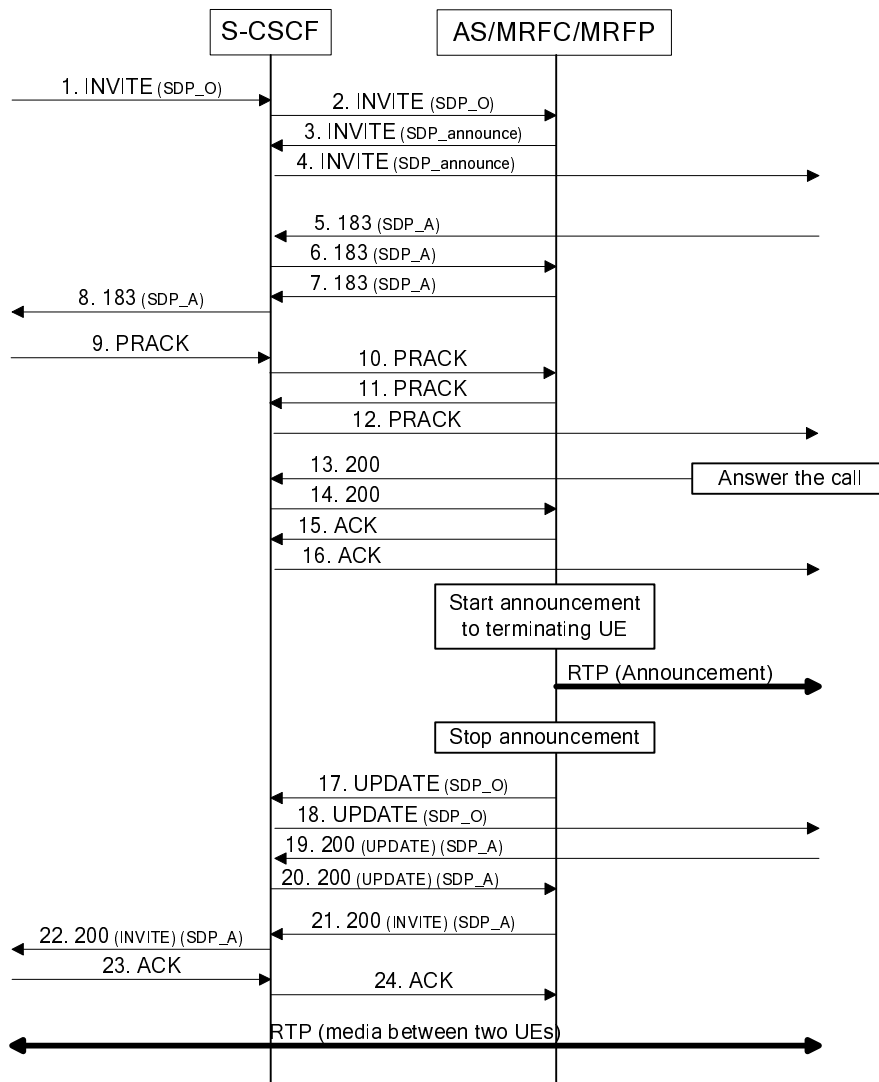


Figure A.13: Announcement provided to the terminating UE when the called user answers the call and prior to establishing media stream between users

The steps of the flow are as follows:

- 1-2) The originating UE sends INVITE request to the intermediate IM CN subsystem.
- 3-4) The AS modifies the SDP in the received INVITE request so that the media stream will be established to the MRFP associated with the AS.
- 5-8) The called party is alerted. The terminating UE sends SIP 183 (Session Progress) provisional response for the INVITE request.
- 9-12) PRACK request is sent towards the terminating UE.
- 13-14) The called party answers the call. The terminating UE sends SIP 200 (OK) response for the INVITE request. The AS providing the announcement terminates this response.
- 15-16) The AS sends ACK request to the terminating UE. The MRFP associated with the AS starts the announcement to the terminating UE, and the called user will receive the announcement.
- 17-18) When the announcement finishes, the AS sends UPDATE request containing SDP received in the INVITE request from the originating UE to the terminating UE.
- 19-20) The terminating UE updates the media stream based on the information received in the UPDATE request. The terminating UE sends SIP 200 (OK) response to the UPDATE request. The AS providing the announcement terminates this response.

NOTE: SDP answer conveyed in this 200 (OK) response from the terminating UE is assumed to be the same as the SDP answer conveyed in 183 (Session Progress) response in step 5-8.

21-22) The AS sends 200 OK response to the originating UE.

23-24) The originating UE sends ACK request. The AS providing the announcement terminates this request. The media between the two UEs are established.

A.5.2 Using 180 (Ringing) response towards originating UE

This subclause shows an example signalling flow of how an AS can send an announcement to the called user just after the call is answered and before establishing direct communication session to the calling user.

Separate early dialogs are established between the originating UE and the AS controlling the announcement, and the originating UE and the terminating UE. The dialog between the originating UE and AS is in this example only used for sending a 180 (Ringing) response.

Figure A.14 shows the message flow for the scenario.

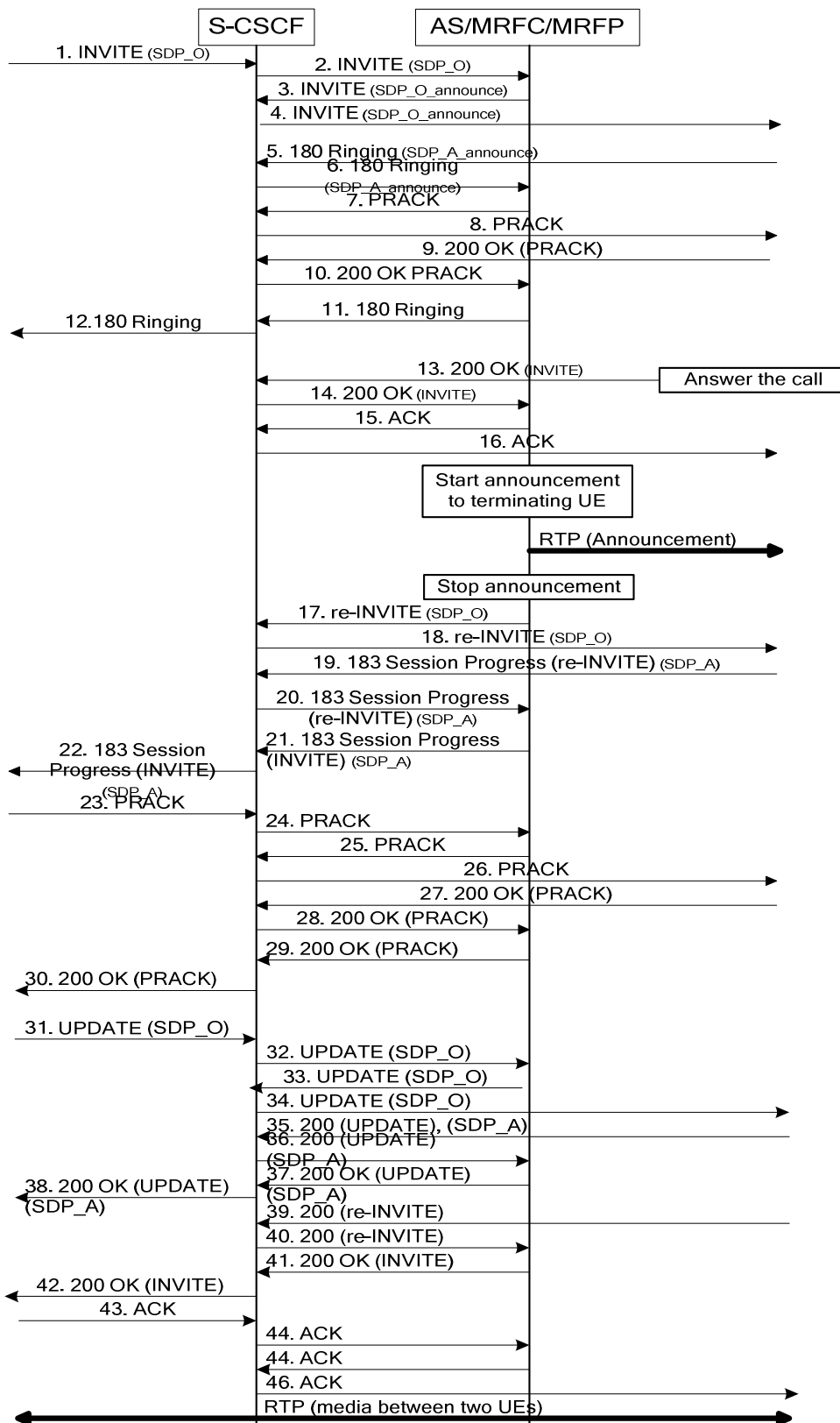


Figure A.14: Announcement provided to the terminating UE when the called user answers the call and prior to establishing media stream between end users

NOTE: For clarity, the SIP 100 (Trying) responses are not shown in the signalling flow.

The steps of the flow are as follows:

- 1-2) The originating UE sends INVITE request to the intermediate IM CN subsystem. The originating UE does not have resources available.
- 3-4) The AS modifies the SDP in the received INVITE request so that the media stream will be established to the MRFP associated with the AS. The media parameters may be different than those received in the offer. The AS indicates that the MRF resources are available.
- 5-6) Since the MRF resources are available, the terminating UE starts reservation of the resources and when the resources are available the called party is alerted. The terminating UE sends SIP 180 (Ringing) provisional response for the INVITE request.
- 7-8) PRACK request is sent towards the terminating UE.
- 9-10) terminating UE confirms the PRACK with 200 (OK) response to PRACK.
- 11-12) The AS sends a 180 (Ringing) response without pre-condition towards the calling party.
- 13-14) The called party answers the call. The terminating UE sends SIP 200 (OK) response for the INVITE request. The AS providing the announcement terminates this response.
- 15-16) The AS sends ACK request to the terminating UE. The MRFP associated with the AS starts the announcement to the terminating UE, and the called user will receive the announcement.
- 17-18) When the announcement finishes, the AS sends a re-INVITE request containing the SDP offer received in the INVITE request from the originating UE to the terminating UE. All media offered for providing the announcement is removed (i.e. media lines are set to port "0") if not included in the original offer from the calling party. The remaining media are indicated without having resources at the originating side.
- 19-20) Since the originating UE does not have resources, the terminating UE sends the SDP answer in 183 (Session Progress) response to the re-INVITE request based on the information received in the re-INVITE request.
- 21-22) The AS sends a 183 (Session Progress) response towards the calling party with the SDP received in message 20. Note that the response is a different dialog than the dialog created by the 180 (Ringing) response in step 11-12.
- 23-30) The calling party sends the PRACK request which the AS forwards to terminating UE and which the terminating UE confirms by 200 (OK) for PRACK.
- 31-38) When resources are reserved by the originating UE the originating UE sends an UPDATE request which the AS forwards to the terminating UE and which the terminating UE confirms by 200 (OK) for UPDATE.
- 39-40) When the terminating UE has all the resources available, the terminating UE sends 200 (OK) response to the re-INVITE request.
- 41-42) The AS sends the 200 (OK) response to the original INVITE request.
- 43-46) The originating UE sends ACK request. The AS providing the announcement forwards the ACK to the terminating UE.

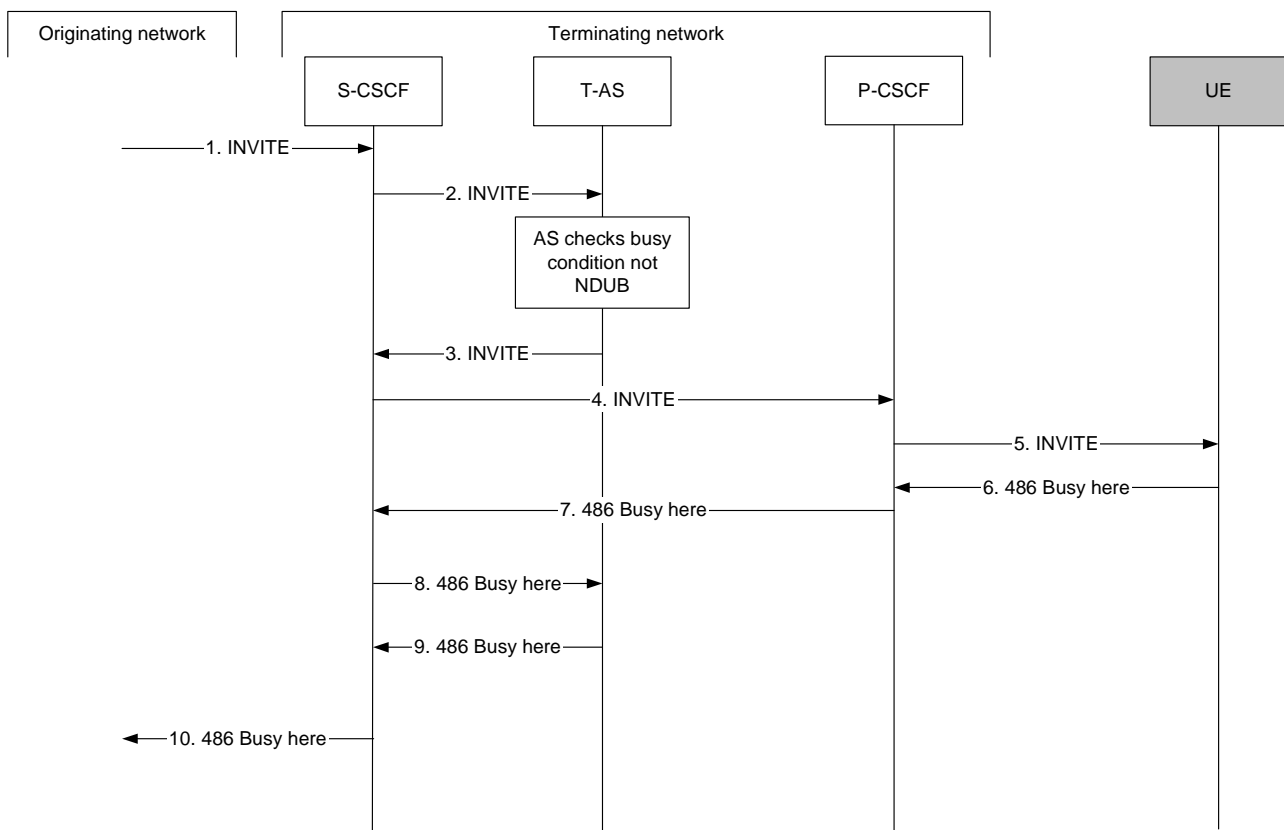
The media between the two UEs are established.

Annex B (informative): Signalling flows for Network Determined User Busy (NDUB)

B.1 Basic call with UE busy with T-AS involvement (NDUB condition check)

This subclause describes the signalling flow for the case when the user is busy but the network does not consider the user to be busy.

Figure B.1 shows the signalling flow for the scenario.



NOTE: The signalling flow is simplified for readability reasons.

Figure B.1: Basic call with UE busy with T-AS involvement (NDUB condition check)

This signalling flow assumes the following:

- the user in the terminating network needs the involvement of an AS for NDUB or other busy condition activated services like CCBS or CFBS; and
- the filter criteria are set for basic communication accordingly.

NOTE: The same scenario applies also for other error responses e.g. for the 403 (Service Denied) response, the 480 (Temporarily Unavailable) response.

The steps of the flow are as follows:

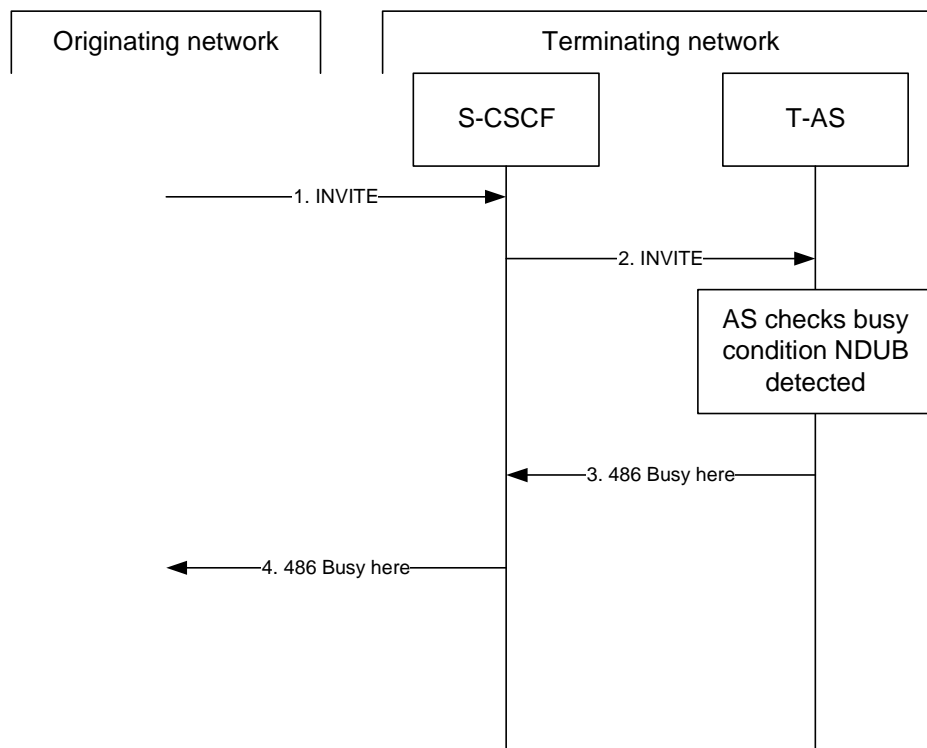
- 1) The S-CSCF serving the terminating user receives an INVITE request from the originating network. The originating network can be an IMS network, a PSTN/ISDN Emulation network, another SIP based network or a MGCF interworking with PSTN/ISDN.

- 2) The S-CSCF checks the IFC and finds that a trigger fires and sends the INVITE request to the AS. The address to the AS is obtained from the IFC.
- 3) The AS checks the busy condition and it is not NDUB and sends the INVITE request to the S-CSCF.
- 4) The S-CSCF sends the INVITE request according to the P-CSCF.
- 5) The P-CSCF sends the INVITE request according to the UE#2.
- 6) The UE#2 is e.g. involved in another communication and determines itself to be busy and sends a 486 (Busy here) response to the P-CSCF.
- 7) The 486 (Busy here) response to originating network via the S-CSCF and the AS.

B.2 Busy condition (NDUB) detected by terminating AS

This subclause shows an example of a signalling flow when a terminating network determines the user to be busy i.e. the NDUB case.

Figure B.2 shows the signalling flow for the scenario.



NOTE: The signalling flow is simplified for readability reasons.

Figure B.2: Busy condition (NDUB) detected by terminating AS

This signalling flow assumes the following:

- the user in the terminating network needs the involvement of AS for NDUB or other busy condition activated services like CCBS or CFBS; and
- that the filter criteria are set for basic communication accordingly.

The steps of the flow are as follows:

- 1) The S-CSCF serving the terminating user receives an INVITE request from the originating network. The originating network may be an IMS network, a PSTN/ISDN Emulation network, another SIP based network or a MGCF interworking with PSTN/ISDN.

- 2) The S-CSCF checks the IFC and finds that a trigger fires and sends the INVITE request to the AS. The address to the AS is obtained from the IFC.
- 3) The AS checks the busy condition and detects that it is NDUB and sends a 486 (Busy here) response to the S-CSCF.
- 4) The AS sends the 486 (Busy here) response to the originating network via the S-CSCF.
- 5) The S-CSCF sends the 486 (Busy here) response to the originating network.

Annex C (normative):
Void

Annex D (normative): AS establishing multiple dialogs with originating UE

D.1 General

If the AS needs to establish an early dialog between itself and the originating UE (or originating network), for example in order to establish a media path in order to send announcements or other kind of early media backwards, it shall do so by sending a provisional response towards the originating UE. The setup procedures between the originating UE and the AS are identical to normal setup procedures.

The To header tag value in the dialog between the originating UE and the AS shall, in order to separate the dialogs, be different than the To header tag value in messages used on the dialog used between the originating and terminating UEs. The AS normally receives the To header tag value for the dialog between the UEs from the terminating UE (or the terminating network), but if the AS acts as a B2BUA it may also, depending on the functionality, generate a new To header value.

If the AS needs to establish an early dialog between itself and the originating UE triggered by the receipt of a provisional responses of the terminating UE, and the terminating UE has already included a SDP answer in a reliable provisional response, the AS first shall forward the provisional response to the originating UE reliably in the e2e dialog, after changing the Status-Line to SIP 183 (Session Progress) response.

The need for the AS to establish an early dialog between itself and the originating UE is determined on the services offered to the originating UE.

If the AS wants to terminate the early dialog between itself and the originating UE before the terminating UE has sent a final SIP response, and the originating UE has indicated support of the 199 (Early Dialog Terminated) response code [18], the AS shall send a 199 (Early Dialog Terminated) provisional response towards the originating UE.

NOTE 1: Unless the originating UE can determine that the messages sent on the early dialog between itself and the AS are originated from the AS, it will assume that forking has occurred in the network.

NOTE 2: If the originating UE has indicated that it does not want the initial INVITE to be forked, the AS may still establish a separate early dialog between itself and the originating UE, since even though the originating UE may assume that the call has been forked only one terminating UE will actually receive the INVITE request.

NOTE 3: Once the originating UE has received 200 (OK) from the terminating UE the early dialog between the originating UE and the AS will be terminated, as described in IETF RFC 3261 [4].

Annex E (informative): Signalling flows for 3rd party call control

The following signalling flows provide examples for the 3pcc procedures described in subclause 4.7.2.9.7.

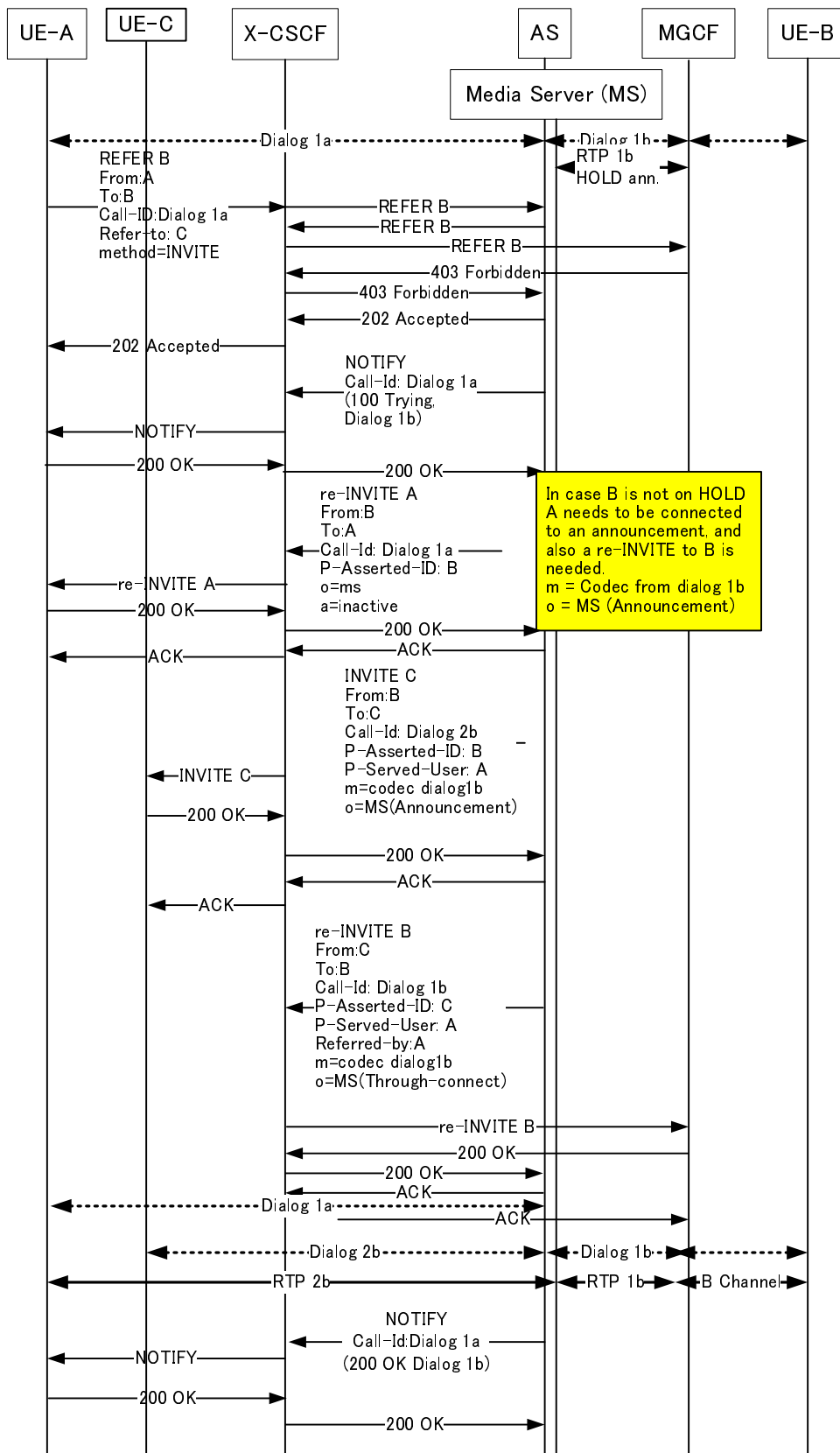


Figure E.1: Example flow for REFER interworking with REFER sent inside a dialog with usage of a media server

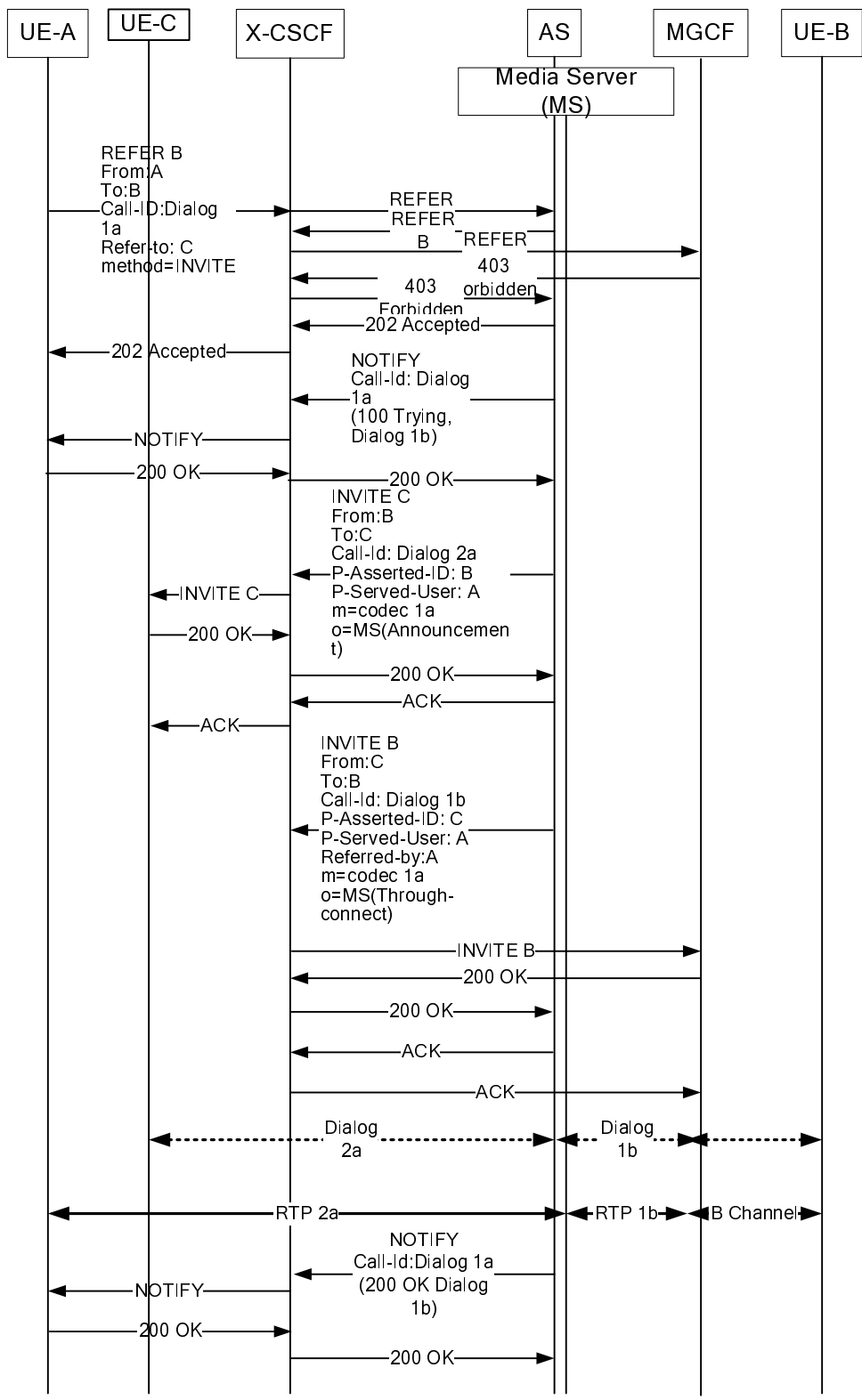


Figure E.2: Example flow for REFER interworking with REFER sent outside a dialog with usage of a media server

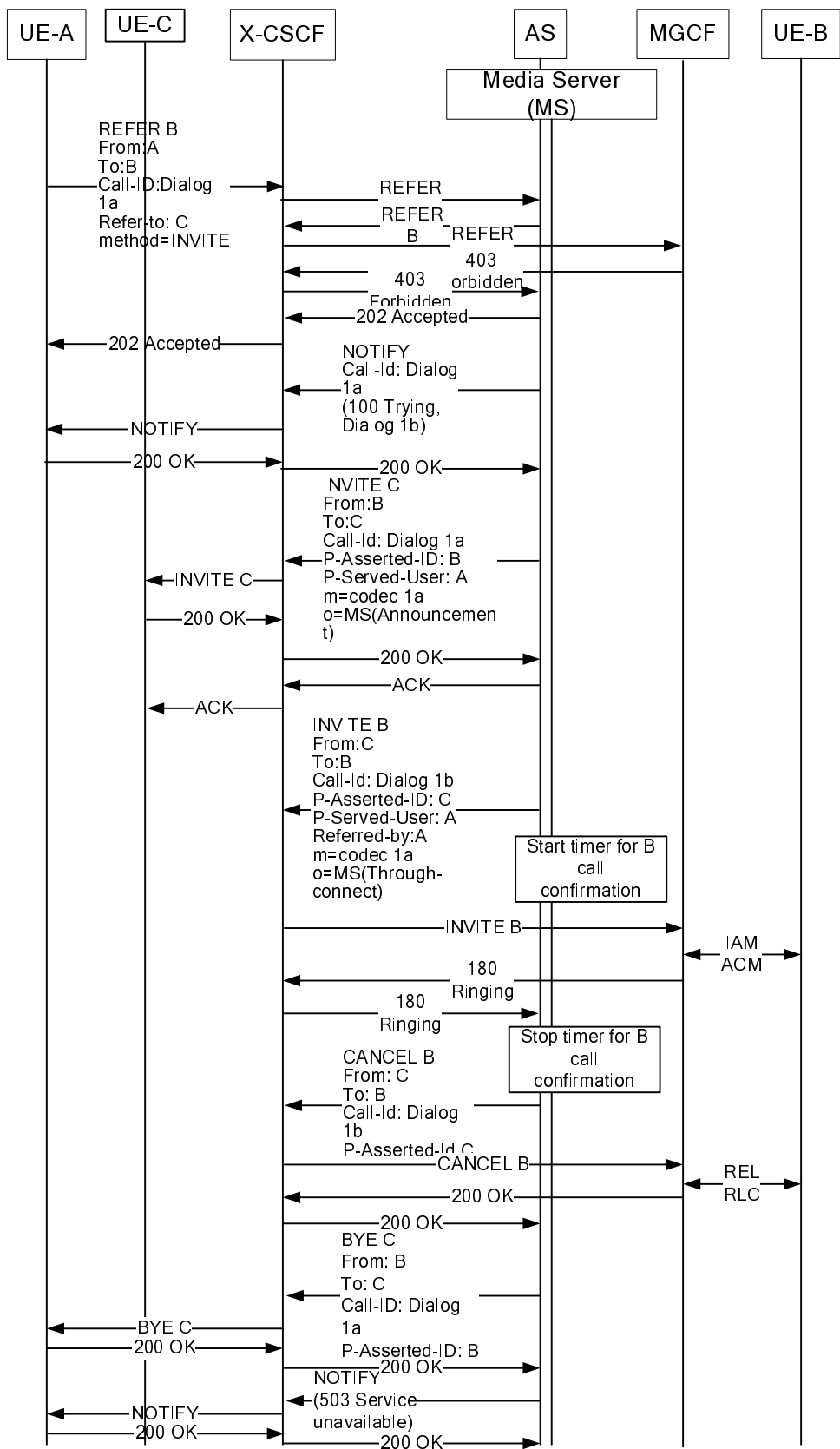


Figure E.3: Example flow for REFER interworking in case of No Reply with usage of a media server

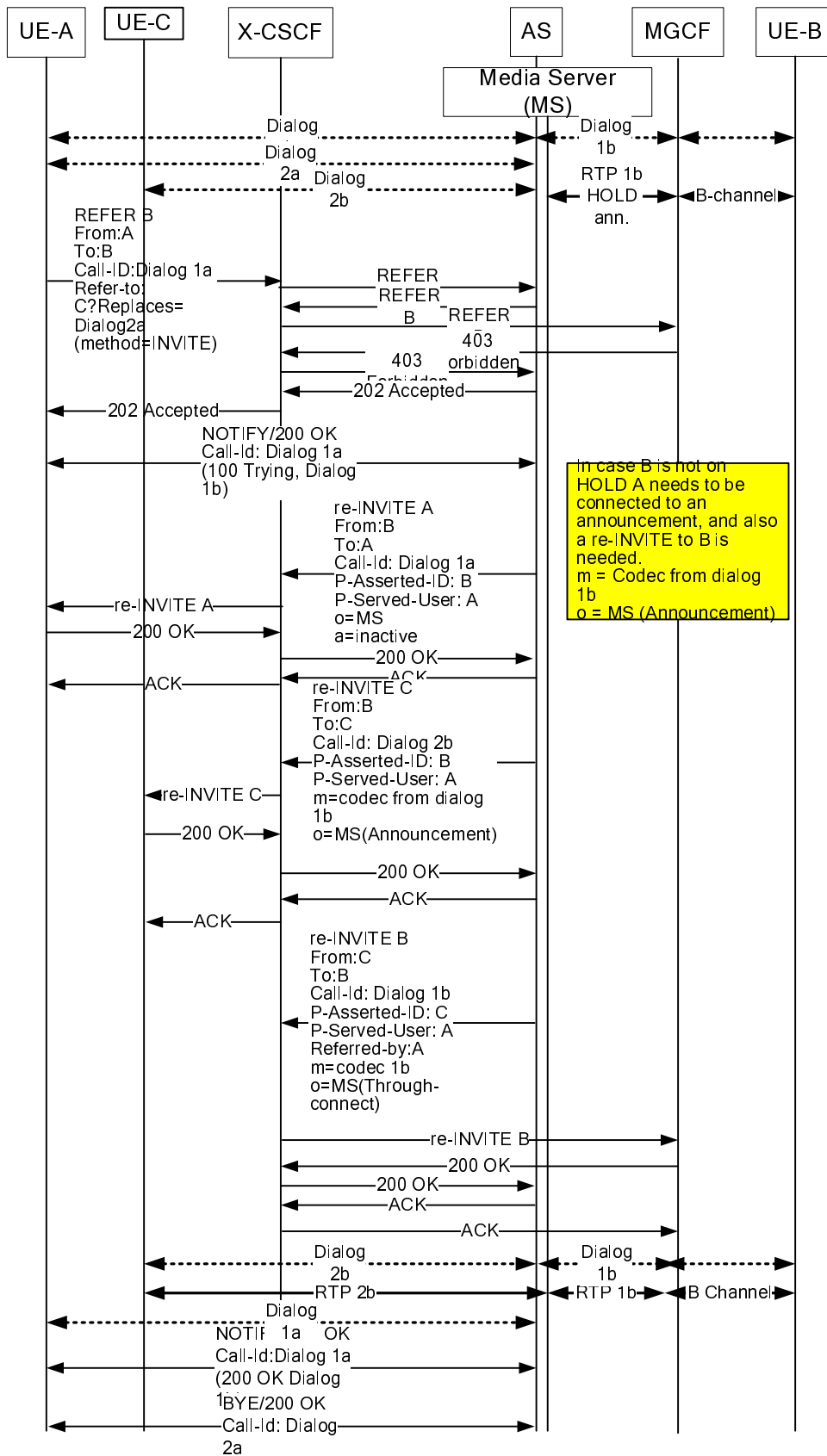


Figure E.4: Example flow for REFER interworking in case the Refer-to header field contains a replaces parameter with usage of a media server

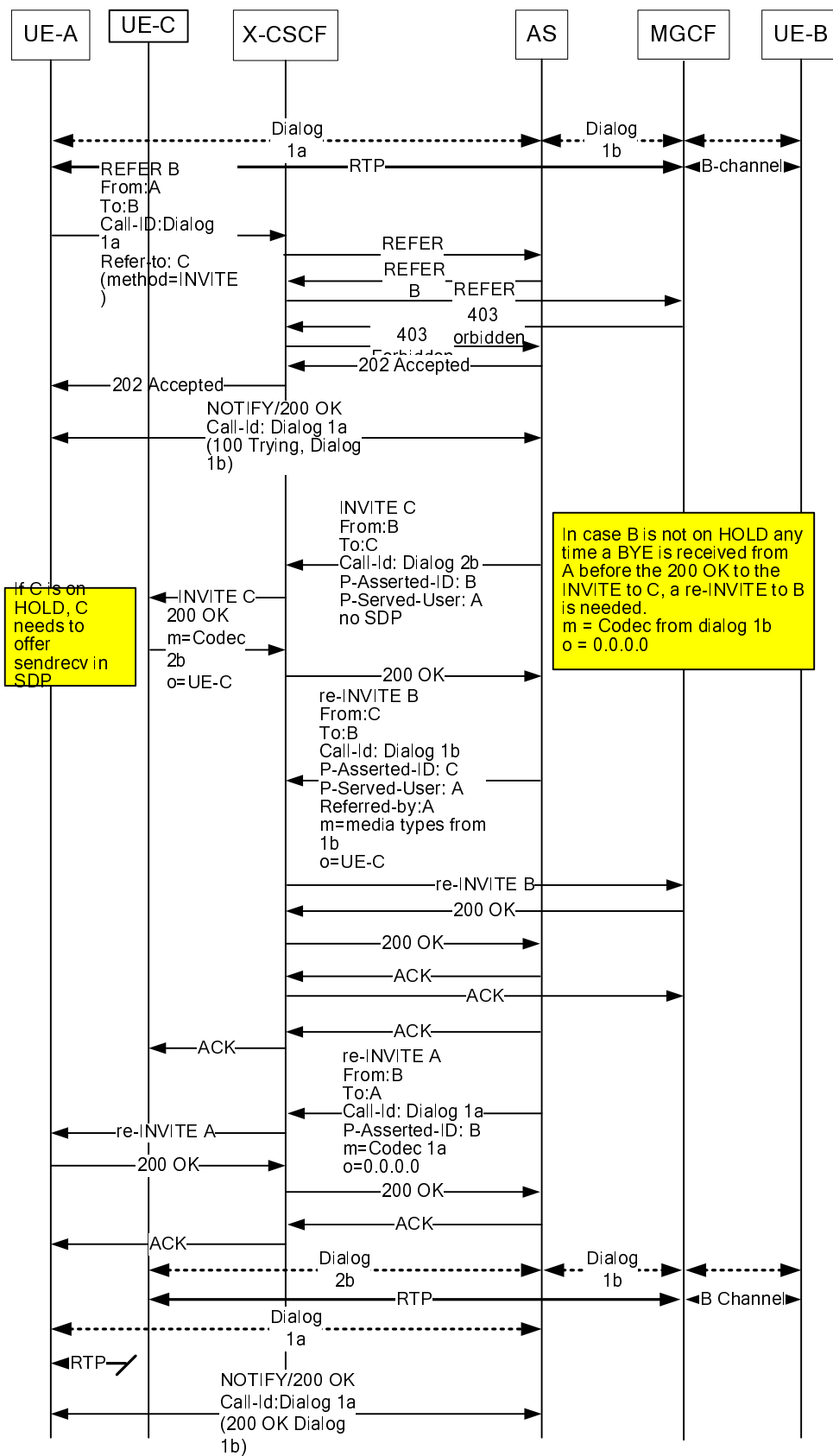


Figure E.5: Example flow for REFER interworking with REFER sent inside a dialog without usage of a media server

Annex F (informative):
Void

Annex G (informative): Change history

Change history							
Date	TSG #	TSG Doc.	CR	Rev	Subject/Comment	Old	New
2008-01					Publication as ETSI TS 183 028		2.4.0
2008-01					Conversion to 3GPP TS 24.528		2.4.1
2008-01					Technically identical copy as 3GPP TS 24.628 as basis for further development.		2.4.2
2008-02					Implemented C1-080103		2.5.0
2008-04					The following CR"s were incorporated and the editor adopted their content / structure to the structure of the TS C1-081007 C1-081096 C1-081097 C1-081236 C1-081243	2.5.0	2.6.0
2008-04					Word generated systematic errors which have been corrected	2.6.0	2.6.1
2008-05					The following CR"s were incorporated and the editor adopted their content / structure to the structure of the TS C1-081560 C1-061617 C1-081835 C1-081904 C1-081917	2.6.1	2.7.0
2008-05					Editorial changes done by MCC	2.7.0	2.7.1
2008-06	CT#40	CP-080334			CP-080334 was approved by CT#40 and version 8.0.0 is created by MCC for publishing	2.7.1	8.0.0
2008-09	CT#41	CP-080533	0001		Editorial fixes for the flows	8.0.0	8.1.0
2008-09	CT#41	CP-080521	0002		Deletion of unnecessary usage of announcement	8.0.0	8.1.0
2008-09	CT#41	CP-080533	0003		Applicability statement in scope	8.0.0	8.1.0
2008-12	CT#42	CP-080865	0004		Usage of 199 (Early Dialog Terminated) for announcements	8.1.0	8.2.0
2008-12	CT#42	CP-080854	0005		TISPAN in-band media for announcements	8.1.0	8.2.0
2008-12	CT#42				Editorial cleanup by MCC	8.1.0	8.2.0
2009-06	CT#44	CP-090430	0008	2	Identification of call answered by "network answering machine"	8.2.0	9.0.0
2009-09	CT#45	CP-090682	0010	3	Addition of an announcement flow	9.0.0	9.1.0
2010-09	CT#49	CP-100519	0012	2	Providing announcement to the terminating user to the immediate established session	9.1.0	10.0.0
2010-09	CT#49	CP-100519	0013	2	Additional callflow for providing announcement	9.1.0	10.0.0
2010-12	CT#50	CP-100864	0014	2	Correcting callflow and procedure for providing announcement to called party	10.0.0	10.1.0
2010-12	CT#50	CP-100864	0015		Correction on providing announcement to the terminating user to the immediate established session	10.0.0	10.1.0
2010-12	CT#50	CP-100872	0018		Reference update: draft-ietf-sipcore-199	10.0.0	10.1.0
2011-03	CT#51	CP-110168	0021	1	Reference update: 199	10.10	10.2.0
2011-06	CT#52	CP-110452	0024	1	Reference update: 199	10.2.0	10.3.0
2011-06	CT#52	CP-110452	0028		P-Served-User in 3pcc REFER procedure	10.2.0	10.3.0
2011-09	CT#53	CP-110693	0029	1	Including 3GPP2 ECT procedure	10.3.0	11.0.0
2012-12	CT#58	CP-120793	0030	1	Alternative ring tone correction	11.0.0	11.1.0
2012-12	CT#58	CP-120793	0034	2	In-band ringing	11.0.0	11.1.0
2012-12	CT#58	CP-120883	0039	3	Communication rejection announcement correction	11.0.0	11.1.0
2013-03	CT#59	CP-130116	0040	1	Communication progress information	11.1.0	11.2.0
2013-06	CT#60	CP-130265	0041	1	Correcting figure numbers	11.2.0	12.0.0
2013-09	CT#61	CP-130504	0042	3	Screening of OPTIONS"s response content	12.0.0	12.1.0
2013-12	CT#62	CP-130770	0043	2	Update to RFC 6665	12.1.0	12.2.0
2013-12	CT#62	CP-130763	0044	2	VoLTE Calls – Forward Early Media	12.1.0	12.2.0
2013-12	CT#62	CP-130763	0045	4	Announcement during hold	12.1.0	12.2.0
2014-03	CT#63	CP-140143	0046	1	Media direction in generated SDP offer when receiving a re INVITE without SDP	12.2.0	12.3.0

History

Document history		
V12.3.0	October 2014	Publication