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Contents

Foreword

This Technical Specification (TS) was been produced by ETSI Technical Committee Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN) and originally published as ETSI TS 183 028 [17]. It was transferred to the 3rd Generation Partnership Project (3GPP) in January 2008.

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x the first digit:

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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".

[22] IETF RFC 7647 (September 2015): "Clarifications for the Use of REFER with RFC6665".

[23] 3GPP TS 22.173: "Multimedia Telephony Service and supplementary services".

[24] 3GPP TS 22.001: "Principles of circuit telecommunication services supported by a Public Land Mobile Network (PLMN)".

[25] IETF RFC 4796: "The Session Description Protocol (SDP) Content Attribute".

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	3rd 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. Services may provide an alternative ring tone to override local ring tones provided by the UE as described in subclause 4.3. Subclause 4.7.2.1 describes the procedure by which the UE can locally generate the communication progress information.

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.

NOTE: An announcement can be triggered by various conditions such as other supplementary services or receiving an indication. However triggers by which the AS makes the decision to send an announcement is outside the scope of this specification.

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 described in annex G 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;
- 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
- create a new media stream by SDP offer/answer mechanism for providing 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

A service may provide an announcement when a communication request is rejected e.g. in order to explain the reason for the rejection of 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 described in annex G 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 field with the proper cause value in the 3xx, 4xx, 5xx or 6xx response to the INVITE request; 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 field with the proper cause value in the 3xx, 4xx, 5xx or 6xx response to the INVITE request; 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 filed 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.

NOTE 1: In-band information received from the network overrides any locally generated communication progress information also when the most recently received P-Early-Media header fields of all early dialogs contain "inactive" or "recvonly".

NOTE 2: When multiple early dialogs exist with authorization as "sendrecv" or "sendonly", the mechanism used by the UE to associate the received early media with the correct early dialog is unspecified in this version of this specification.

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".

If an early dialog exists where a SIP 180 response to the SIP INVITE request 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.

NOTE 3: According to 3GPP TS 22.173 [23] the UE for an MMTel session generates the communication progress information specified in clause F.2 of 3GPP TS 22.001 [24], with parameters applicable for the home network of the UE.

If the UE supports the P-Early-Media header field as defined in IETF RFC 5009 [12], and at least one P-Early-Media header field has been received on at least one early dialog, 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" 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 or has been put on hold by both users at both ends; and

- "sendrecv" otherwise.

During the established communication, if a video stream is provided with a media level attribute "a=sendonly" and the media level attribute "a=content: g.3gpp.announcement-no-confirmation" as specified in 3GPP TS 24.229 [1], the UE should play this video stream without confirmation with the user if playing video announcement without confirmation is allowed based on UE's local policy (e.g. configuration on the UE).