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1 Introduction

1.1 Scope

This document is part of MTAS TPD document series and focuses on MMTel AS. For MTAS common features and other application server features, please read other TPD documents.

1.2 Change History

Revision	Date	Comments/Changes
A	2015-11-17	Updated: ch 2.45 (CAT) ch 2.8.3 ch 2.35.1 (MDUCAC)
B	2016-04-22	Updated: Ch. 2.46 (NRBT) Ch 2.57 added (Long Duration Call) Ch 2.11 (STOD) Cname interface info removed - Ch 2.28 (Identity Presentation)
C	2016-08-12	Updated: Ch 2.58 Dialog Event Notifier service Ch 2.44 Network Announcement Updated with Network message
D	2016-09-23	Updated: Ch 2.42 GETS priority Service
E	2017-02-27	Add Ch SIP Upstream Overload Control Updates: 2.16.1.11 (OCB) 2.25.1 (DNM) 2.44.1. (NetAnn) 2.28 (IdPres) 2.53 (WS)
F	2017-02-27	Updated: 2.49 (SSF)
G	2017-04-06	Editorial fixes
H	2017-08-23	Updated: Ch 2.54 Ro retarget and vMTAS Licensing



2 MMTel AS Features

2.1 Overview

Ericsson MTAS MMTel AS provides a number of services for the operator and end users. Most MMTel services are 3GPP R8 compliant at the least and some services are proprietary and unique to Ericsson's MMTel AS.

2.1.1 Call handling

- Basic voice and multimedia communication
 - Basic Voice Communication
 - Video Communication
 - Text Chat
 - File Sharing
 - Add/Drop Media
 - Precondition
- Hold communication
- Communication Diversion (CDIV)
 - Communication Forwarding – Unconditional (CFU)
 - Communication Forwarding – Busy (CFB)
 - Communication Forwarding – No Reply (CFNR)
 - Communication Diversion Notification – Served-user
 - Communication Diversion Notification – Reminder
 - Communication Diversion Notification – Caller
 - Communication Diversion – Operator Blacklist
 - Communication Diversion – to Voice Mail
 - Communication Diversion – Rule Based
 - Communication Deflection (CD)
 - Communication Diversion after BYE
- Flexible Communication Distribution
 - Divert Primary
 - Communication Deflection from Primary



- Communication Distribution to Voicemail
- Parallel Ringing
- Serial Ringing
- Flexible Ringing
- FCD to Primary User's Devices
- Caller Preference Filtering
- Auto-Answer Avoidance
- FCD based on Presence
- Support of preconditions
- Communication Waiting (CW)
- Communication Completion (CCxx)
 - Communication Completion on Busy Subscriber (CCBS)
 - Communication Completion on No Reply (CCNR)
 - Communication Completion on Not Logged-in (CCNL)
- Three Party Call
- Ad-hoc Conference
 - Dial-out Multi-party Conference
 - Conference Notifications
- Explicit Communication Transfer (ECT)
 - Consultative Transfer
 - Check of the status of the transferred sessions
 - Termination of the transferred sessions
- Session Transfer to Own Device (STOD)
 - Originating STOD
 - Terminating STOD
 - STOD triggered by Call Pull
- Operator Controlled Transfer
- Call return
- Hotline



- Unconditional hotline
- Operator controlled hotline
- End user controlled hotline
- Media Policy

2.1.2 Communication Barring and Address Policing

- Dial Plan
 - Nodal Dial Plan
 - Dial Plan – OTP-controlled per-VTP Dial Plan
 - Dial Plan – VTP-controlled per-VTP Dial Plan
- Communication Barring (CB)
 - Basic Barring – Barring Programs
 - Incoming Communication Barring Rule Based (ICB)
 - Outgoing Communication Barring Rule Based (OCB)
 - Anonymous Communication Rejection (ACR)
 - Operator Black List
 - Operator White List
 - Operator Anonymous Communication Rejection
- Malicious Communication Identification
- Malicious Communication Rejection
- Dynamic Black List
- Address Policing
 - Long Distance Mobile Number Policing

2.1.3 Number handling

- Abbreviated Dialing
- Short Number Dialing
- Number Normalization
- Dialed Number Mapping



- Number Translation Service
- Carrier Select
 - Pre-Select
 - Select

2.1.4 Identity handling

- Identity Presentation
 - Originating Identity Presentation (OIP)
 - Originating Identity Restriction (OIR)
 - Terminating Identity Presentation (TIP)
 - Terminating Identity Restriction (TIR)
 - Flexible Identity Presentation (FIP)
 - Calling Name Identity Presentation (CNIP)

2.1.5 Charging

- Advice of Charge
- Number Portability
- Account activation
- Administration of user's language preference
- Japanese Charging (JC)
 - Interconnection Charge Billing System (ICBS)
 - Flexible Charging (FCH)
 - Telephone Directory Service (TDS)
- Multi-Device Charging
- Terminating OCS initiated final announcements towards a caller

2.1.6 Call admission control

- Call Admission Control (CAC)
 - User Call Admission Control (UCAC)

2.1.7 Multiple Numbers

- Multiple Subscriber Number



- Distinctive Ring

2.1.8 Interworking

- Gateway Model
- Flexible Service Format Selection (FSFS)
- Video Fallback to Audio
- Bandwidth Optimization

2.1.9 Call prioritization

- Priority servicePriority call
- Calling Party Category (CPC)

2.1.10 Tone and announcement management

- Network Announcement
 - Requested Announcement
- Customized Alerting Tones (CAT)
- Network Provided Ring Back Tone (RBT)
- Generic Announcement

2.1.11 3rd party service interaction

- Parlay X
- GSM compatible Service Switching Function (SSF)

2.1.12 Language

- Multiple Languages Support

2.1.13 Emergency call handling

- Emergency call notification

2.1.14 Closed user group

- Closed user group (CUG)



2.1.15 Long Duration Call Supervision

2.1.16 Long Duration Call SupervisionWholesale

- Wholesale

2.1.17 Dialog Event Subscription and Notification

- Dialog Event Notifier (DEN)

2.2 Basic Voice and Multi-media Communication

2.2.1 Description

MMTel provides the end user with an enriched real time communication experience based on several media components such as voice, video, text chat, sharing of different file attachments and presence information. Two or more users can thus communicate in real time using different media components.

MTAS MMTel AS supports the following basic voice and multi-media communication features.

2.2.1.1 Basic Voice Communication

Traditional voice communication between two parties.

2.2.1.2 Video Communication

Video communications between two parties.

2.2.1.3 Text Chat and File Sharing

Text chat and media sharing between two parties.

This feature involves the creation of a MSRP (RFC4975 [32]) session between two user entities (UEs). An MSRP session is negotiated in the same way as RTP based sessions using SIP and SDP signaling. The main difference is that MSRP requires the establishment of a TCP connection between the UEs.

Once the MSRP session has been established it can be used to send (share) different types of media e.g. pictures, movie clips, audio clips. It can also be used to send text messages on a line by line basis.

For sharing, the MSRP session will typically be setup in parallel with an audio session.

Supported media types are:

- Image



- Voice
- Video
- Text

2.2.1.4 Add/Drop Media

Once an MMTel session is established, media can be re-negotiated at any time from the point where the early session is successfully established until it is being terminated. Media re-negotiation can be requested by both user agent A (UA-A) and user agent B (UA-B). Re-negotiation of media follows the same basic flow as when media is initially negotiated on session establishment in the initial SIP INVITE. The main differences are that it is also possible to use the SIP UPDATE method if supported by both UA-A and UA-B (see ref [21]).

2.2.1.5 Announcement

The terminating MMTel AS may, depending on the configuration, play an audio-only, video-only, or audio-and-video announcement towards UA-A before rejecting the session when there is a terminating call for an unregistered user. MTAS does not play this announcement in case a call diversion (CDIV) rule containing **not-registered** is matched for the served user. See chapter 2.4 for more detail about CDIV.

2.2.2 Example Call Flows

2.2.2.1 Successful MMTel session set-up

The following diagram shows an example call flow of successful session set-up with no precondition, no forking and no reliable provisional responses. Note that IMS network nodes like CSCF are omitted from the diagram in order to keep the diagram simplified and also to highlight the role of the application server. Note that both originating and terminating MMTel AS nodes are described as one node. This is also for simplification purpose. See [1] for more details on MMTel call flows.

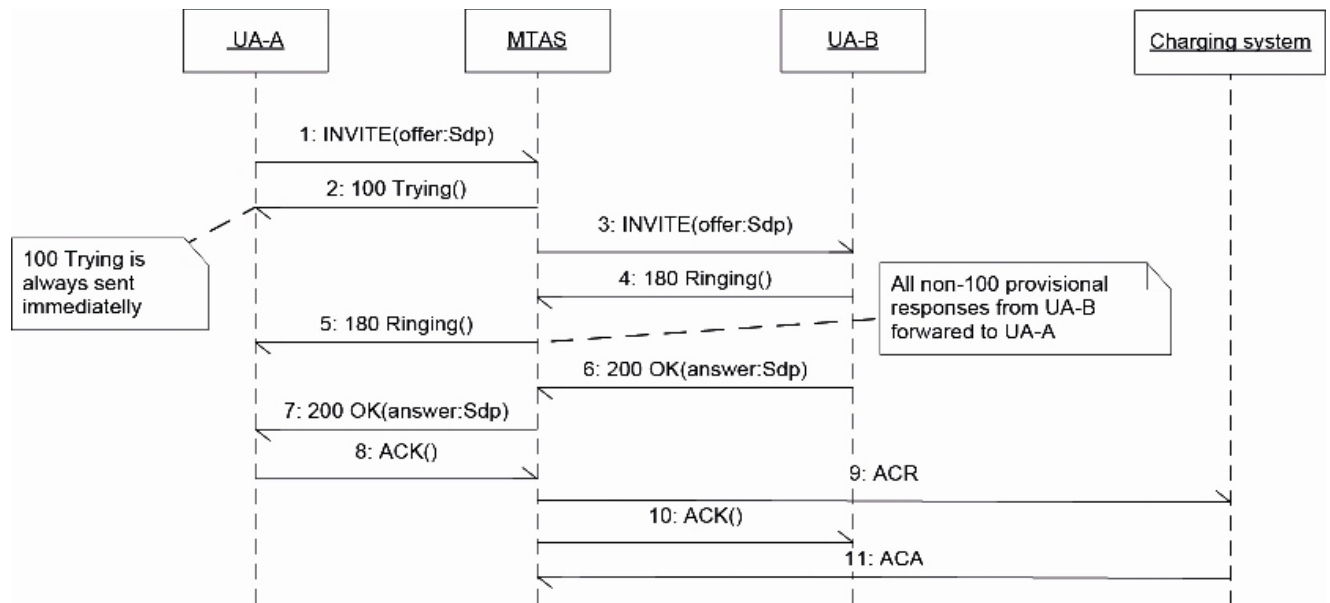


Figure 1 - Successful session establishment with no precondition, no forking and no reliable provisional responses

1. MTAS receives an INVITE request message from UA-A.
2. MTAS sends immediately back the provisional response message 100 Trying to UA-A.

MTAS checks the Accept-Contact ICSI feature tag value with the values configured as primary and secondary feature tags. If any of the known feature tags are present in the header, the session is accepted as an MMTel session and primary feature tag is inserted if not present. On the other hand, if none of the configured feature tag is present in any of the Accept-Contact header, the session is rejected with 503 error code. It is configurable whether MTAS shall accept the session when no Accept-Contact header is present or no feature tag is present in Accept-Contact header.

MTAS fetches the subscriber data from HSS if necessary.

MTAS executes policing in order to ensure that the number of parallel sessions do not exceeds the configured limit and that the number of active media streams within one session does not exceed 10.

Above-mentioned feature tag checking and policing are executed both in originating and terminating MTAS.

3. MTAS sets up a new SIP dialog by sending an INVITE message to the S-CSCF. The SDP in the message body of the request is the same SDP received from UA-A, i.e. MTAS does not affect the media negotiated between UA-A and UA-B for MMTel sessions.
4. UE-B sends 180 Ringing
5. Any provisional response message except 100 Trying received from the terminating side is forwarded to the originating side. MTAS starts a 'No Reply' timer at reception of 180 Ringing.



6. MTAS receives the message 200 OK from UA-B. MTAS stops the 'No Reply' timer.
7. MTAS sends a 200 OK response to UA-A.
8. MTAS receives the ACK request from UA-A.
9. An ACR is sent to the Charging System if the MMTel charging profile has offline charging turned on.
10. MTAS sends the ACK request to UA-B.
11. MTAS receives the ACA from the charging system. The call is considered stable when both the ACK and the ACA have been received.

2.2.2.2 Successful Session Establishment using reliable provisional responses and preconditions

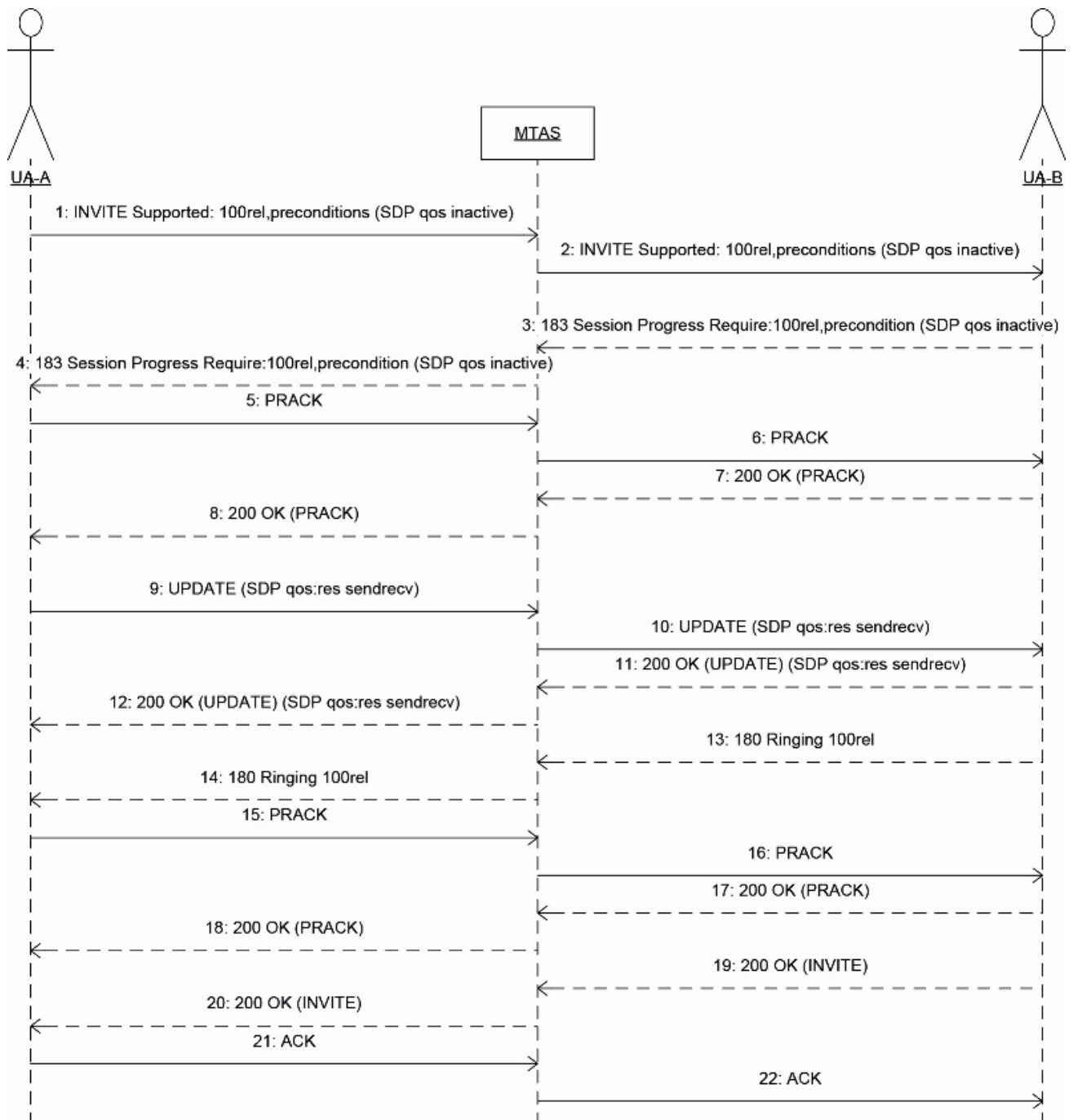


Figure 2 - Session establishment using reliable provisional responses and preconditions



1. INVITE is received from the calling user agent. INVITE contains the following headers:

Supported: 100rel, precondition

Allow: UPDATE

(The INVITE may alternatively contain "Require: 100rel, precondition".)

The MTAS does not police the presence of either of these options in the SIP for end-to-end precondition negotiation. It is the responsibility of the UAs to ensure that the correct Supported or Require tags are included for the precondition negotiation to occur.

The SDP for each media type for which QoS preconditions are used will contain the following attributes which match the following:

a=curr:qos
a=des:qos

If non-QoS preconditions are used then the "qos" will be replaced by the precondition name; e.g. "foo".

2. The MTAS forwards the INVITE towards the called party with the SDP and Supported header unchanged. The SDP includes the precondition attributes as provided by UA-A in step 1.
3. The called party UA responds with a 183 Session Progress which contains the Require header:

Require: 100rel, precondition

Both these capabilities are required to be used for a session set-up using preconditions.

The SDP in the 183 response will contain attribute lines matching the following:

a=curr:qos
a=des:qos
a=conf:qos

The called party UA has now indicated his requirement to reserve resources for the session by inclusion of the 'precondition' and '100rel' tags in the Require header and including their local desired QoS as mandatory, optional or none in the accepted media streams.

E.g. a=des:qos mandatory local sendrecv

4. The MTAS forwards the 183 response to the calling party with the SDP and Require header unchanged. The SDP includes the precondition attributes as provided by UA-B in step 3. It is not the responsibility of the MTAS to police the header tags or SDP content for end-to-end precondition negotiation.



5. The calling UA sends a PRACK to acknowledge the 183 response.
6. The PRACK is forwarded to the called UA.
7. A 200 OK to the PRACK is returned
8. The 200 OK to the PRACK is forwarded.
9. Upon reservation of resources the calling UA sends an UPDATE to indicate that resources have been reserved.

The UPDATE will contain SDP attribute lines which indicate that resources have been reserved by the sending party and that the state of the media stream is now recv/send/sendrecv.

The new SDP offer could also have been included in the PRACK, step5, if the originating UA has reserved resources when this is sent. If this occurs the corresponding SDP answer will be included in the 200 OK to the PRACK, step7, and the UPDATE request and response will not be used. If more than one media stream offered and accepted then the preconditions for one or more media streams may be indicated as reserved in the PRACK with an UPDATE used for other media streams. Multiple UPDATE requests may be used to indicate reservation of different media streams.

10. The MTAS forwards the UPDATE to the called user unchanged.
11. The called UA sends a 200 OK to the UPDATE with an SDP answer confirming that resources have been reserved.

The called UA now indicates that resources have been reserved, this may have previously been notified in the 183 response if resources were reserved at this time.

The 200 OK also confirms the agreed media direction as specified in the UPDATE request.

12. The called UA knowing that resources have been reserved at both ends of the call can now alert the called user.
13. Called UA now sends 180 Ringing indicating that the called user is alerted.
14. 180 Ringing is forwarded by the MTAS to the calling UA.

From this point in the sequence the set-up is the same as for a session set-up without preconditions, Section 2.2.2.1.

2.2.2.3 Play Announcement due to session set-up toward unregistered PUI using MRFP

The following diagram shows an example call flow of session set-up toward an unregistered PUI with no reliable provisional responses. Announcement playing is controlled using an MRFP over the Mp interface.

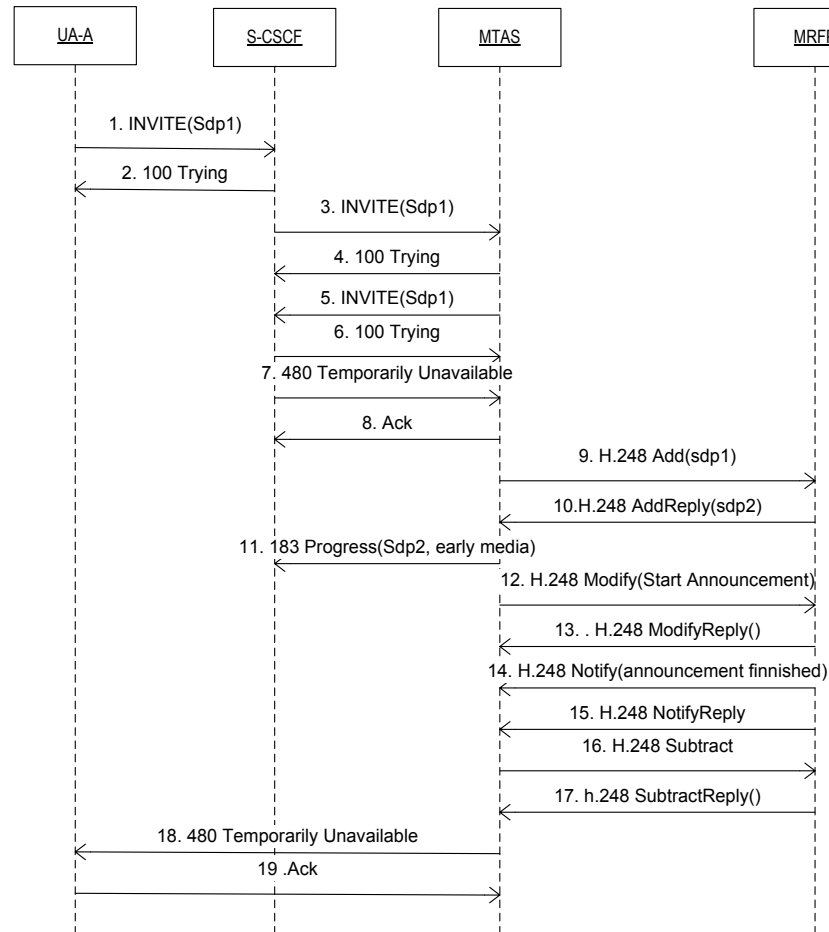


Figure 3 - Session initiation toward unregistered PUI with no PRACK

1. UA-A sends an INVITE toward the destination PUI which is not registered. S-CSCF receives the INVITE.
2. 100 Trying is sent immediately
3. S-CSCF sends the INVITE to MTAS that receives this INVITE message indicating “terminating unregistered” session case.
4. 100 Trying is sent immediately
5. The MTAS sends an INVITE to S-CSCF to break the ISC chaining.
6. 100 Trying is sent immediately
7. As the destination PUI is not registered and no other AS shall be triggered the S-CSCF sends a SIP 480 Temporarily Unavailable response to MTAS.



8. MTAS returns an Ack
9. The Unregistered User service is triggered and MTAS sends a H.248 Add to the MRFP
10. MRFP responds with AddReply containing an updated SDP
11. MTAS sends 183 Progress requiring SDP answer with the P-Early-Media header set.
12. MTAS sends H.248 Modify to start playing the announcement
13. MRFP confirms started announcement by sending H.248 ModifyReply
14. MRFP sends H.248 Notify when the announcement has finished.
15. MTAS sends a NotifyReply
16. MTAS sends H.248 Subtract to remove the termination
17. MRFP answers with H.248 SubtractReply
18. MTAS rejects the session with 480 Temporarily Unavailable
19. UA-A sends ACK

2.2.2.4 Play Announcement due to session set-up towards unregistered PUI using MRFC

The following diagram shows an example call flow of session set-up towards unregistered PUI with no reliable provisional responses. Announcement playing is controlled using an MRFC over the Mr interface.

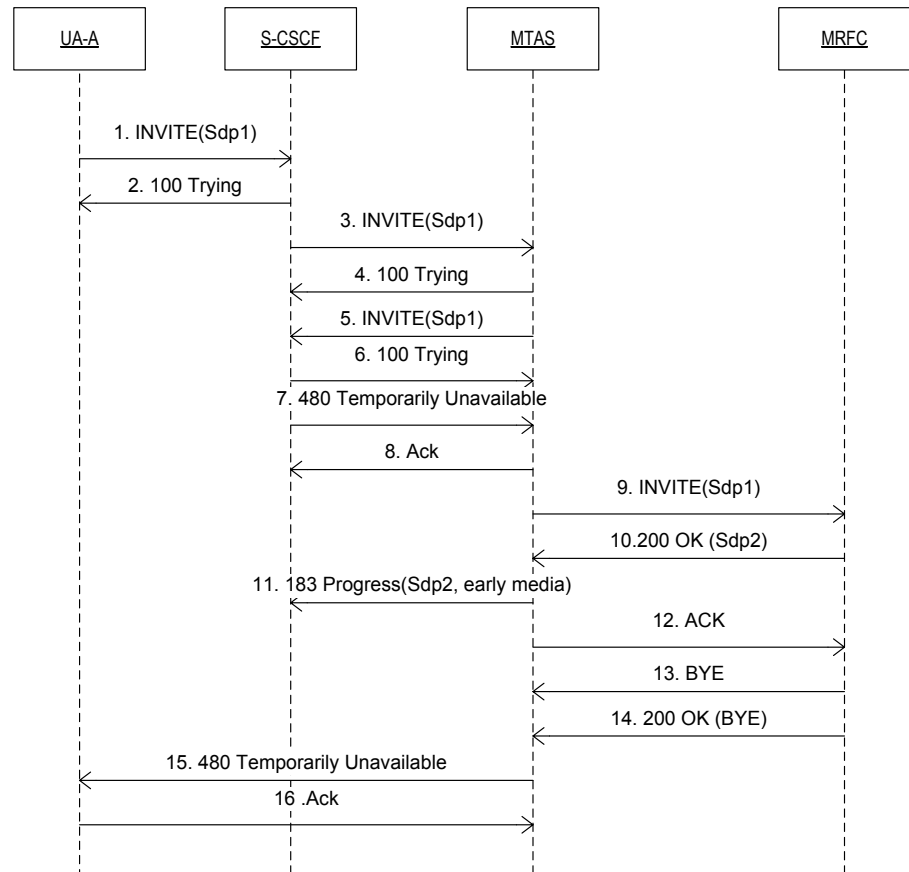


Figure 4 Session Initiation towards unregistered PUI with no PRACK

1. UA-A sends an INVITE toward the destination PUI which is not registered. S-CSCF receives the INVITE.
2. 100 Trying is sent immediately.
3. S-CSCF sends the INVITE to MTAS triggering the “terminating unregistered” session case.
4. 100 Trying is sent immediately.
5. The MTAS sends an INVITE to S-CSCF to break the ISC chaining
6. 100 Trying is sent immediately.
7. As the destination PUI is not registered and no other AS shall be triggered the S-CSCF sends a SIP 480 Temporarily Unavailable response to MTAS
8. MTAS returns a SIP Ack to S-CSCF.



9. The Unregistered User service is triggered and MTAS sends an INVITE to MRFC to reserve resources.
10. MRFC responds with a 200 OK containing an updated SDP.
11. MTAS sends 183 Progress requiring SDP answer with the P-Early-Media header set.
12. MTAS sends ACK start playing the announcement.
13. MRFC sends BYE when the announcement has finished.
14. MTAS sends a 200 OK for the BYE.
15. MTAS rejects the session with 480 Temporarily Unavailable.
16. UA-A sends ACK.

2.2.3 Configuration

Examples of node-level configuration parameters related to the basic voice and multimedia communication service are:

- Ports on MTAS differentiating originating, originating-unregistered, terminating and terminating-unregistered session cases.
- Timers related to SIP protocol, sessions and no reply.
- Primary/Secondary Feature Tag(s)
- Max number of parallel MMTel sessions per user
- Allow or reject session establishment with no feature tag
- Enable and disable announcement

2.2.4 Performance Management

Examples of performance counters related to the basic voice and multimedia communication service are:

- Number of successful and unsuccessful MMTel session initiations in originating MTAS
- Number of successful and unsuccessful MMTel session terminations in terminating MTAS
- Number of media streams per type audio/video/text/message
- Number of failed announcement



2.3 Hold Communication

2.3.1 Description

The Communication Hold/Resume (hereafter HOLD) supplementary service enables a user to suspend active media stream(s) of an established session, and resume the media stream(s) at a later time. This service is standardized by 3GPP, see ref [17].

This supplementary service is executed in the originating AS meaning the AS for the user which initiated the HOLD procedure.

2.3.1.1 Hold

A *hold* invocation is recognized when a stream's direction attribute is changing and the change matches one of the predefined *hold* transitions.

An SDP offer, in a SIP re-INVITE or SIP UPDATE, where a stream's direction attribute change is interpreted to be a *hold* request, when the direction attribute changes from:

- recvonly -> inactive
- sendrecv -> sendonly

Hold can also be performed for a group of streams by changing the direction attribute in the same SDP offer.

The HOLD service may be configured to play an audio and/or video announcement to the end user being put on hold.

2.3.1.2 Resume

An SDP offer, where a stream's direction attribute change is interpreted to be a *resume* request, when the direction attribute changes from:

- inactive -> recvonly
- sendonly -> sendrecv

Resume can also be performed for a group of streams by changing the direction attribute in the same SDP offer.

At *resume* any announcement to the end user on hold will be removed. It also maintains counters to evaluate the usage of resume.

2.3.1.3 Announcement

If audio or/and video announcement is configured to be enabled, the held user will receive an audio or video announcement corresponding to the type (audio or audio/video) that was held.



Announcements are played in-band, as specified in 3GPP. MTAS does not support out band announcements.

2.3.2 Example Call Flow

2.3.2.1 Hold with no announcement

The following diagram shows an example call flow of HOLD service invocation using the SIP UPDATE method. Communication has previously been setup between User A and User B, and RTP is sent in both directions.

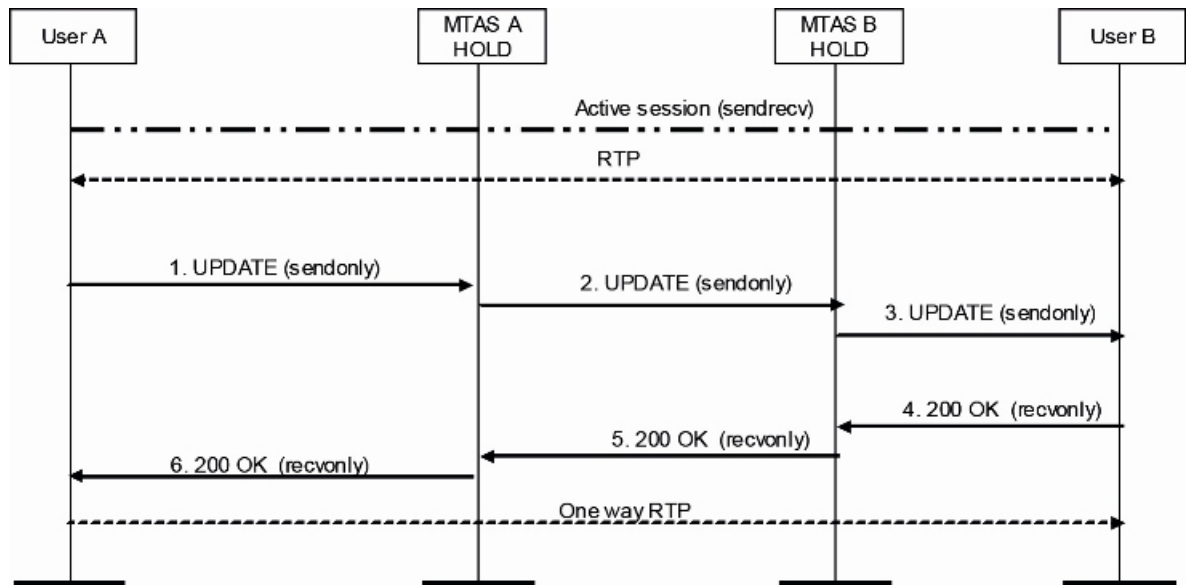


Figure 5 - Hold with UPDATE method, no reliable provisional response

1. An UPDATE is received in MTAS-A as a consequence of User A initiating a procedure to put User B on hold. As the SDP stream direction parameter in the UPDATE is "sendonly" and the stream is currently in "sendrecv", MTAS-A detects that User A is putting User B on hold.
2. An UPDATE is received in MTAS-B as a consequence of User A initiating a sequence to put User B on hold. As the UPDATE did not originate from the User B, no further action is taken. The UPDATE is forwarded to the B User.
3. User B inhibits the outgoing stream as a consequence of receiving the UPDATE with "sendonly".
4. User B responds with 200 OK with parameter "recvonly" indicating that the User B shall only receive data.
5. The User B response 200 OK is received in MTAS-B. As the MTAS-B is not the controlling MTAS for the user initiating the hold, no further action is taken. The OK is forwarded to the MTAS-A.
6. As announcement is not enabled, no further action is taken but forwarding of the 200 OK to User A.



2.3.2.2 Hold with announcement

The following diagram shows an example call flow of hold with announcement. Note that MTAS B is omitted as its behavior is the same as the previous call flow (i.e. transparent from Hold service point of view).

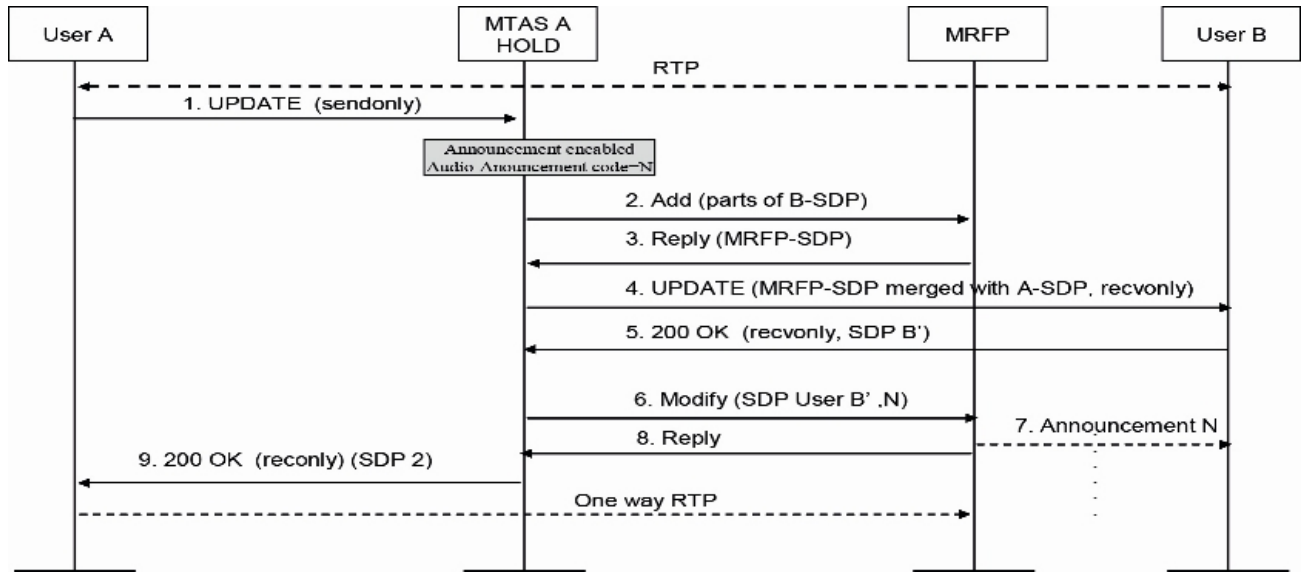


Figure 6 - Hold with announcement, no reliable provisional response

1. User A sends a SDP offer indicating a request to put user B on hold.
2. Since user B should be played an announcement and user A shall be on silent, two new MRFP terminations are needed.

The Add request to the MRFP includes the codec for user B's stream which will be put on hold, N which indicates which announcement (code) to be played and user A's IP, port and codec ('m' and 'c' line information of SDP) and local control mode "recvnly".

3. MRFP returns SDPs describing the resource reserved:
 - i. MRFP termination for User A:
'c': IP address of MRFP
'm': Port and Codec
 - ii. MRFP termination for User B:
'c': IP address of MRFP
'm': Port and Codec
4. The SDP offer received from User A is passed on to User B with UPDATE, but the 'm' and 'c' lines are substituted/modified with the information received for the MRFP termination for User B (3. ii)
5. The 'c' and 'm' lines for the stream put on hold is extracted from the received B-SDP answer and prepared to be sent to MRFP. Independent of the announcement handling it is checked if it's a legal media direction transition and successful counter is stepped.



6. The extracted B-SDP 'c' and 'm' lines are sent to MRFP. The announcement is requested.
7. MRFP starts sending the announcement using the codec indicated by B's new SDP
8. Reply received from MRFP
9. 200 OK passed on to User A with the SDP content received for the MRFP termination for User A (3. i).

2.3.3 Configuration

Examples of node-level configuration parameters related to the HOLD service are:

- Enable and disable audio-video announcement

2.3.4 Performance Management

Examples of performance counters related to the HOLD service are:

- Number of initiated *hold* and *resume*
- Number of successful initiation of *hold* and *resume*
- Number of failed initiation of announcement

2.4 Communication Diversion

2.4.1 Description

The Communication Diversion (CDIV) supplementary service enables a user to have the network redirect incoming communication to another user either automatically – communication diversion - based on the served user's CDIV rule set or based on interaction with the served user's UA – communication deflection. This service is standardized by 3GPP, see ref [13].

The CDIV feature exists both in the originating and terminating MTAS. Communication diversion and deflection is executed by the terminating MTAS while service activated indication is executed by the originating MTAS when the served user initiates an outgoing communication.

The CDIV rules are built up with different conditions and actions and can be combined in many ways to express if a communication shall be diverted or not and if notifications shall be sent.

The different condition can be based on:

- Whether the served user is busy.
- Whether the served user is not registered.



- Whether the served user is not reachable.
- The served user's current presence activity status.
- The calling user's identity.
- Whether the calling user's identity is anonymous
- What media the incoming communication offers
- Whether the served user does not answer.
- Time periods during which diversion is valid.
- Time periods during which diversion is NOT valid.
- Assembly of complex time condition based upon several calendar sub-conditions (times of day; days of the week; calendar months; calendar weeks; private and public holidays; daily, weekly and monthly repetitions etc.)

In case the Multi Subscriber Number service is enabled also the served user's MSN identity may be used in the condition. For further details see chapter 2.36.

There is also a possibility to deactivate a rule without deleting the rule configuration.

In the action part of the rule the target of the diversion is defined. Other action settings are:

- Caller can be notified.
- Diverted-to user's identity can be revealed to caller.
- Served user's identity can be revealed to diverted-to party.
- Communication is forwarded due to that Do Not Disturb is active
- Play generic announcement
- Notify served user that diversion occurred.
- Notify that diversion is active, at outgoing call attempt.

The rule sets include one or more rules, each rule having one or more conditions as illustrated in the figure below. There are two ruleset in the operator part. Pre-and post-evaluate ruleset, pre-evaluated is before the user's ruleset and post-evaluate is evaluated after the user's ruleset. Each rule in the full set of rules is evaluated, from top to bottom.

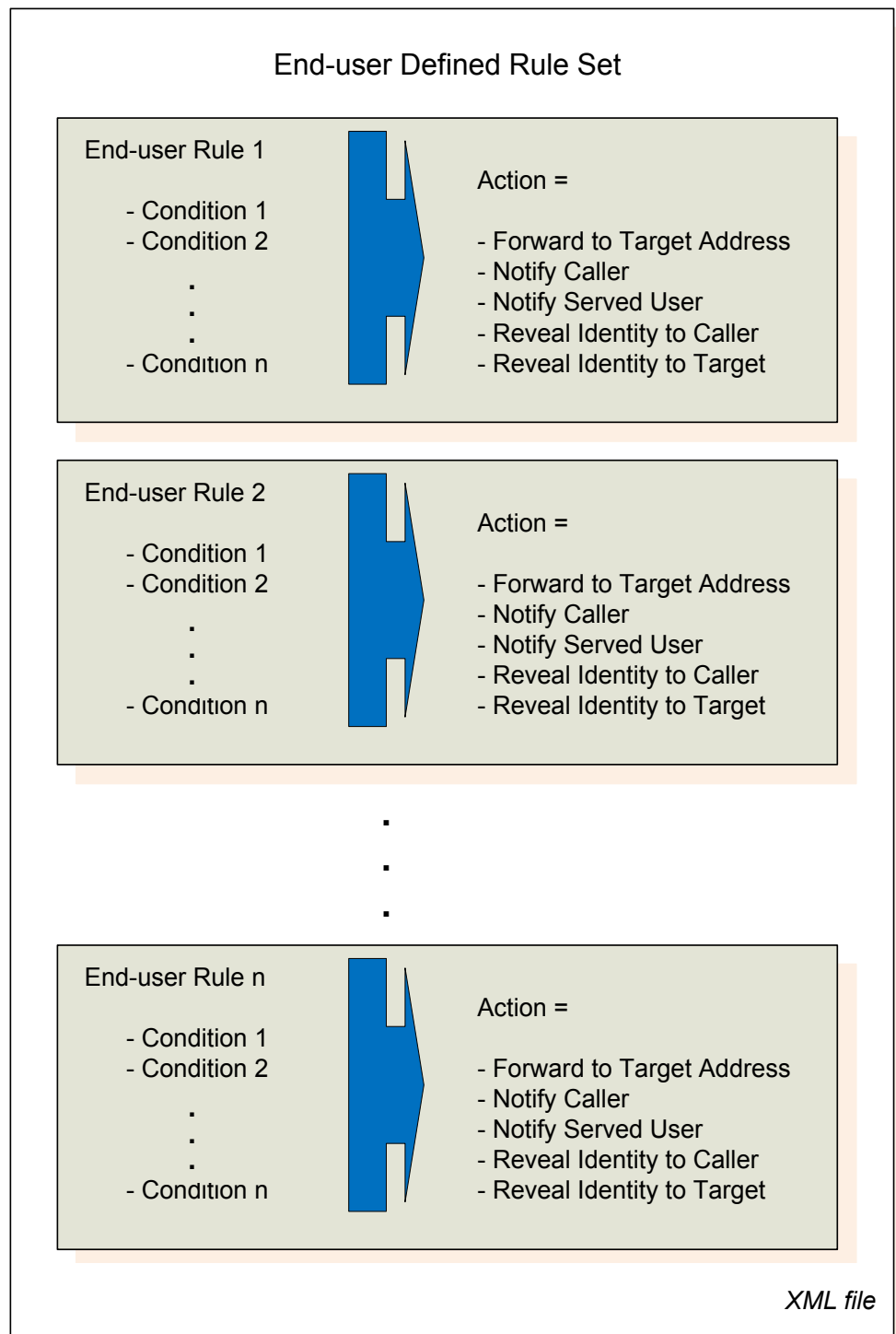


Figure 7 - CDIV Rule Set

It is possible to configure the maximum number of diversions permitted for each communication in MTAS. If enabled, the communication is rejected when the maximum is exceeded. The number of diversions is calculated based on the number of entries in the History-Info header including a 'cause' URI parameter related to the communication diversion service.

MTAS can support two originating AS modes for handling of originating services after re-targeting.



When originating AS chaining is enabled MTAS adds the served user's URI to the header of the re-targeted request. MTAS then sends the request directly to the S-CSCF without immediately executing originating services of the MTAS. The S-CSCF can then initiate triggering of originating services after re-targeting, both in MTAS and in other AS:es. See figure Figure 8.

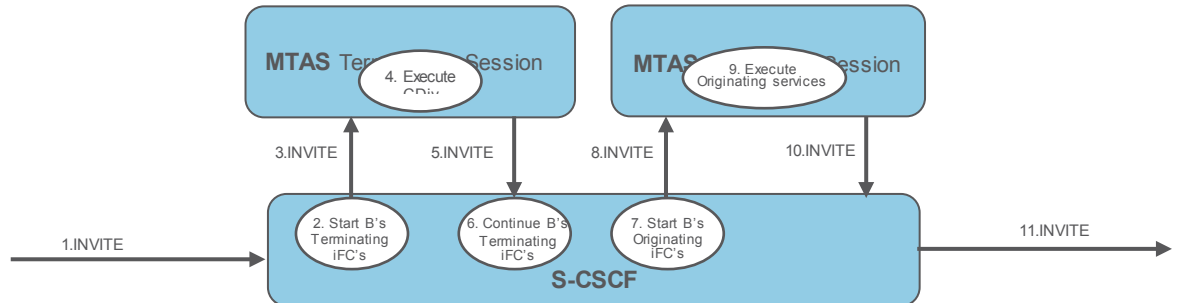
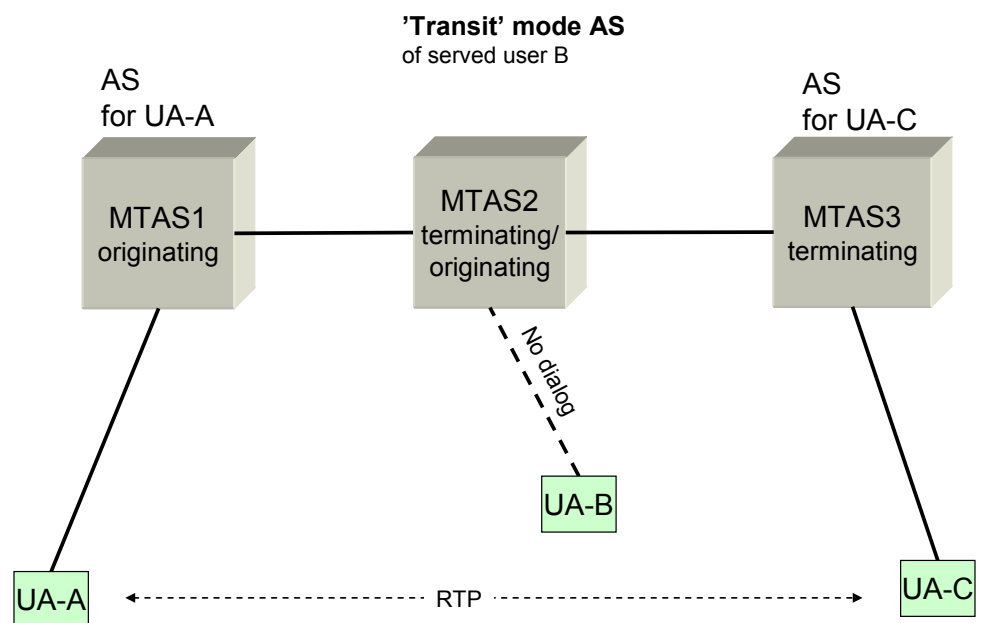


Figure 8 - Originating services after re-targeting in transit AS

In the alternative case when originating AS chaining is disabled, MTAS will not add the P-Served-User header but execute originating services of MTAS for the served user immediately before sending the re-targeted request to the S-CSCF. S-CSCF will not trigger any additional originating AS:es before routing the request towards the diverted-to user.

When the communication is diverted the terminating AS of the served user becomes a transit AS which is kept in the message path. MTAS services will act differently for in a transit AS compared to the originating or terminating AS.



UA-A has called UA-B which has diverted the call to UA-C. UA-B's serving AS stays in the chain due to charging.



CDIV is broken down into the following features:

2.4.1.1 Communication Forwarding – Unconditional (CFU)

This feature enables the end-user to forward all incoming communication to another destination.

In case of CFU, the rule does not contain any condition therefore each incoming communication will be forwarded.

The CFU only operates on the initial INVITE method.

2.4.1.2 Communication Forwarding – Busy (CFB)

The end-user can forward the incoming communication if his/her is busy.

In case of CFB the condition in the rule is the busy status of the served user. If a busy condition is matched the communication is forwarded to the target address.

2.4.1.3 Communication Forwarding – No Reply (CFNR)

The end-user can forward the incoming communication if he/she does not answer the call.

In case of CFNR MTAS starts a timer at the reception of the SIP response 180 (Ringing). The duration of the timer can be configured in the rule, in the CDIV service or on MTAS node level. If the timer expires the communication is forwarded to the target address.

2.4.1.4 Communication Forwarding – Origination (CFO)

The end-user can forward his/her incoming communication based on the Caller's identity.

In case of CFO the condition in the rule is the calling user's identity (i.e. the "P-Asserted-Identity"). If there is more than one "P-Asserted-Identity" in the header then MTAS iterates over all identities, and evaluates the rule(s). If one identity matches then the MTAS forwards the communication.

This condition is part of the commercial feature Communication Diversion – Rule Based (see ref [1]).

2.4.1.5 Communication Forwarding – Time (CFT)

The end-user can forward his/her incoming communication based on the Time.

In case of CFT the condition in the rule is matched against the time of the incoming communication request.

Three time related conditions can be used, and even combined in the rules:



- validity
MTAS will divert the communication when the current time is within the validity period. End-user can define one or more time periods. Time periods are defined by a starting time and an ending time.
- invalidity
MTAS will divert the communication when the current time is NOT within the invalidity period. End-user can define one or more time periods. Time periods are defined by a starting time and an ending time.
- valid-periods
The valid-periods condition allows assembly of complex time condition based upon several calendar sub-conditions (times of day; days of the week; calendar months; calendar weeks; private and public holidays; daily, weekly and monthly repetitions etc.).

MTAS will divert the communication when the current time matches to all included sub-conditions.

This condition is part of the commercial feature Communication Diversion – Rule Based (see ref [1]).

2.4.1.6 Communication Forwarding – Media (CFM)

The end-user can forward his/her incoming communication to different destinations depending on the media type of the incoming call request.

This condition is part of the commercial feature Communication Diversion – Rule Based (see ref [1]).

2.4.1.7 Communication Forwarding – Not Logged-in (CFNL)

The end-user can forward his/her incoming communication when he/she is unregistered.

This condition is part of the commercial feature Communication Diversion – Rule Based (see ref [1]).

2.4.1.8 Communication Forwarding – Anonymous (CFA)

The end-user can forward his/her anonymous incoming communication.

In case of CFA the communication will be forwarded if the identity of the calling user is set to anonymous, e.g. the “Privacy” header field exists in the INVITE message. MTAS will also forward the communication if the “P-Asserted-Identity” header is not provided in the message.

This condition is part of the commercial feature Communication Diversion – Rule Based (see ref [1]).



2.4.1.9 Communication Forwarding – Not Reachable (CFNRc)

The end-user can forward his/her incoming communication when he/she is out of reach, e.g. in some kind of access network temporary failure or out of reach for radio.

In case of CFNRc the communication will be forwarded if the end-user is not reachable, i.e. when MTAS receives a Not Reachable response (default 408, 500, 503 or 504) from the served user.

This condition is part of the commercial feature Communication Diversion – Rule Based (see ref [1]).

2.4.1.10 Communication Forwarding – Do Not Disturb (DNDCF)

This feature enables the end-user to forward all incoming communication to another destination when end-user activates do not disturb.

DNDCF is active when the rules actions contains the do-not-disturb element.

This condition is part of the commercial feature Communication Diversion – Rule Based (see ref [1]).

2.4.1.11 Communication Diversion – Presence

The end-user can forward his/her incoming communication based on his/her presence activity status. This requires a presence system in the IMS network.

This condition is part of the commercial feature Communication Diversion – Rule Based (see ref [1]).

2.4.1.12 Communication Deflection (CD)

Communication Deflection (CD) is a flavor of CDIV which does not require any rules but always diverts a call when triggered by a 'deflection' redirect response from the served user's UA.

At Communication Deflection the served user sends back a SIP response (302 - Moved Temporarily) to an INVITE message with the "Contact" header containing a URI of a party to which the call should be diverted.

The main difference between Communication Diversion and Communication Deflection is that diversion requires the data to be pre-configured and stored while deflection allows the end-user to send back a new destination for the specific communication request (INVITE), i.e. on case-by-case basis.



2.4.1.13 Operator Blacklist

The operator can define a blacklist to which it is not possible to perform communication diversion. The CDIV blacklist has the same format as the Communication Barring Operator Blacklists (chapter 2.16.1.7). When a call is diverted, MTAS bars the call if the new destination matches the blacklist, responding 486 Busy Here. Each time a diversion rule is inserted or amended, MTAS only allows the change if the target does not match the blacklist.

2.4.1.14 Communication Diversion after BYE

Operator can divert calls in terminating MMTel AS when the called party answered the initial INVITE with 200 OK but it sends a BYE request within a specified time period with a configured value in the Reason header.

MMTel AS starts a timer when 200 OK for initial INVITE is received on the terminating side. If a BYE request is received from the called party while the timer is still running and a Reason header in this BYE matches any of the preconfigured reason values and the subscriber has voicemail service enabled then the call is transferred to the voicemail address.

If the subscriber has an activated and provisioned voicemail address in the subscriber profile, then it is used as target, otherwise the service uses the configured default voicemail address.

Call forwarding consists of a new call initiated to voicemail server and the existing call leg to calling party is updated with a new SDP by sending reINVITE request.

Diversion after BYE is indicated in charging message by sending a service specific SSID.

2.4.2 Example of CDIV rule

The following example shows a CDIV rule which forward all anonymous incoming communication to voicemail@example.com.

```
<ss:communication-diversion active="true">
  <cp:ruleset>
    <cp:rule id="anon">
      <cp:conditions>
        <anonymous/>
      </cp:conditions>
      <cp:actions>
        <forward-to>
          <target>sip:voicemail@example.com</target>
        </forward-to>
      </cp:actions>
    </cp:rule>
  </cp:ruleset>
</ss:communication-diversion>
```



2.4.3 Charging

Terminating charging is performed on the incoming leg (A->B) and Originating charging is performed on the outgoing leg (B->C).

2.4.4 Service Interaction

When diversion occurs in the transit MTAS (see Figure 9), the following service interactions are taken into account:

2.4.4.1 Identity Presentation

- TIP: The originating network handles TIP.
- TIR: TIR is applied to the diverting served user's identity information.
- OIP: The terminating network handles OIP.
- OIR: OIR is applied to the diverting served user's identity information.
- CNIP: The terminating network handles CNIP.

Communication diversion is not invoked if CLIR interworking is active and the user calls with hidden identity.

2.4.4.2 Communication Barring

Barring has precedence over CDIV. Note that outgoing barring is performed on the diverted outgoing terminating INVITE from the served user's MTAS.

2.4.4.3 Multi Subscriber Number (MSN)

When the MSN feature is enabled the CDIV conditions may also be specified per MSN identity.

2.4.5 Self administration

The end user can by itself administer his/her CDIV rules via the Ut-interface and via SSCs.

For the list of supported SSCs for Communication Diversion please refer to "Self Administration via Service Codes" feature.



2.4.6 Provisioning

MTAS enables the operator to configure the service on user-level through the CAI3G interface. Possible settings are:

- To set pre- and/or post-evaluated CDIV rule set, which will be evaluated before or after the user's CDIV rule set
- Which individual conditions and actions the user is allowed to use by providing the XML structure to the user's CDIV rule set.
- Activate/deactivate the Voice Mail subscription
- Voice mail address
- Maximum number of CDIV rules

2.4.7 Configuration

Examples of node-level configuration parameters related to the Communication Diversion service are:

- Communication Diversion enable/disable
- Max number of Communication Diversions in the network
- Audio and Video Announcement configuration
- Text for Communication Diversion Notification – Reminder
- Text for Communication Diversion Notification – Served-user
- Error response when CFB condition met – 486, 600 or 603
- Communication Deflection enable/disable
- CDIV Operator Blacklist
- Voicemail default address
- CDIV on BYE Reason
- CDIV on BYE timer

2.4.8 Performance Management

Examples of performance counters related to the CDIV services are:

- Number of successful communication diversions
- Number of failed communication diversions



2.4.9 Fault Management

No specific alarms for CDIV.

2.5 Flexible Communication Distribution

2.5.1 Description

The Flexible Communication Distribution (FCD) service is an IMS terminating service which rings specified targets in a serial, parallel or in a combined flexible mode. The served user can set rules via the Ut interface.

These rules may be defined for parallel, serial and flexible modes of ringing. The served user may also define the duration allowed before answer for targets in parallel, serial and flexible modes.

The INVITE request to the FCD user may use the default PUI or an alias PUI for the IMS Primary User or non-IMS Primary User.

The IMS served (Primary) user may select one or more of his/her own statically configured terminals or Voice Mail to be addressed in all of the distribution schemes.

The FCD to Primary User's Devices function allows for "dynamic" addressing of all (mobile and fixed) currently registered terminals of a served user or a mobile terminal only.

Specifically, when the primary user is an ICS user for VoLTE in Fixed Mobile Convergence scenarios (FMC), a terminating call can be distributed to the user's devices based on device identifier (sip.instance) and mobility selector, thereby allowing SCC AS to distinguish target device type and only apply T-ADS on calls to VoLTE devices.

Active Call Preference for Fixed Devices is an extension of the FCD to Primary User's Devices function, letting a new incoming call be distributed only to served user's fixed devices with ongoing active calls and a mobile device.

User can configure Divert Primary with a forward-to target to enable communication forwarding from IMS served (Primary) user to a new destination.

The IMS served (Primary) user may deflect the call to another destination.

Related users can be marked for auto-answer avoidance in the served user's profile. When a related user target answers the incoming call with 200 OK, FCD requests the user to press any digit (DTMF key) as a confirmation of not being an automaton (like voicemail). The auto-answer avoidance function can be applied to any type of FCD distribution.



Filtering caller preferences of incoming call allows for avoiding collisions with caller preferences added by the FCD service itself. It is possible to configure the FCD service at node level so that selected caller preferences from the Accept-Contact and Reject-Contact headers of an incoming request are not forwarded to outgoing requests towards distribution targets.

The FCD rules are built up with different conditions and actions and can be combined in many ways to express if a communication shall be distributed or not.

The different condition can be based on:

- Whether the served user is busy.
- Whether the served user is not registered.
- Whether the served user is not reachable.
- The served user's current presence activity status.
- The calling user's identity.
- The served user's identity
- Whether the calling user's identity is anonymous
- What media the incoming communication offers
- Whether the served user does not answer.
- Time periods during which diversion is valid.
- Time periods during which diversion is NOT valid.
- Assembly of complex time conditions based upon several calendar sub-conditions (times of day; days of the week; calendar months; calendar weeks; private and public holidays; daily, weekly and monthly repetitions etc.)
- Regexp pattern matching with a SIP header or header parameter, directly or inversely

There is also a possibility to deactivate a rule without deleting the rule configuration.

In the action part of the rule the targets of the distribution is defined. Other action settings are:

- Parallel distribution
- Serial distribution
- Flexible distribution



- Specify a generic announcement which is played to the caller during the distribution

The rule sets include one or more rules, each rule having one or more conditions. Rules in ruleset are evaluated from top to bottom to the moment when the rule conditions are fulfilled and the rule is executed.

2.5.2 Example Call Flow

2.5.2.1 Parallel ringing

The sequence shown use a Supported: 100rel, 199, precondition header in the received initial INVITE request.

MTAS has the transparent mode switched on.

MTAS has the 199 generation switched on.

The User Agents, UA, shown in the following sequence may be in the IMS network or any other network, including the PSTN. Additional network nodes used for routing are not shown to simplify the sequence.

The Primary User is configured as IMS

The served user has the FCD service provisioned and active and configured to route the call as follows:

Parallel ringing of:

- PRIMARY User B
- Related User C, a voicemail device with Auto-Answer Avoidance
- Related User D

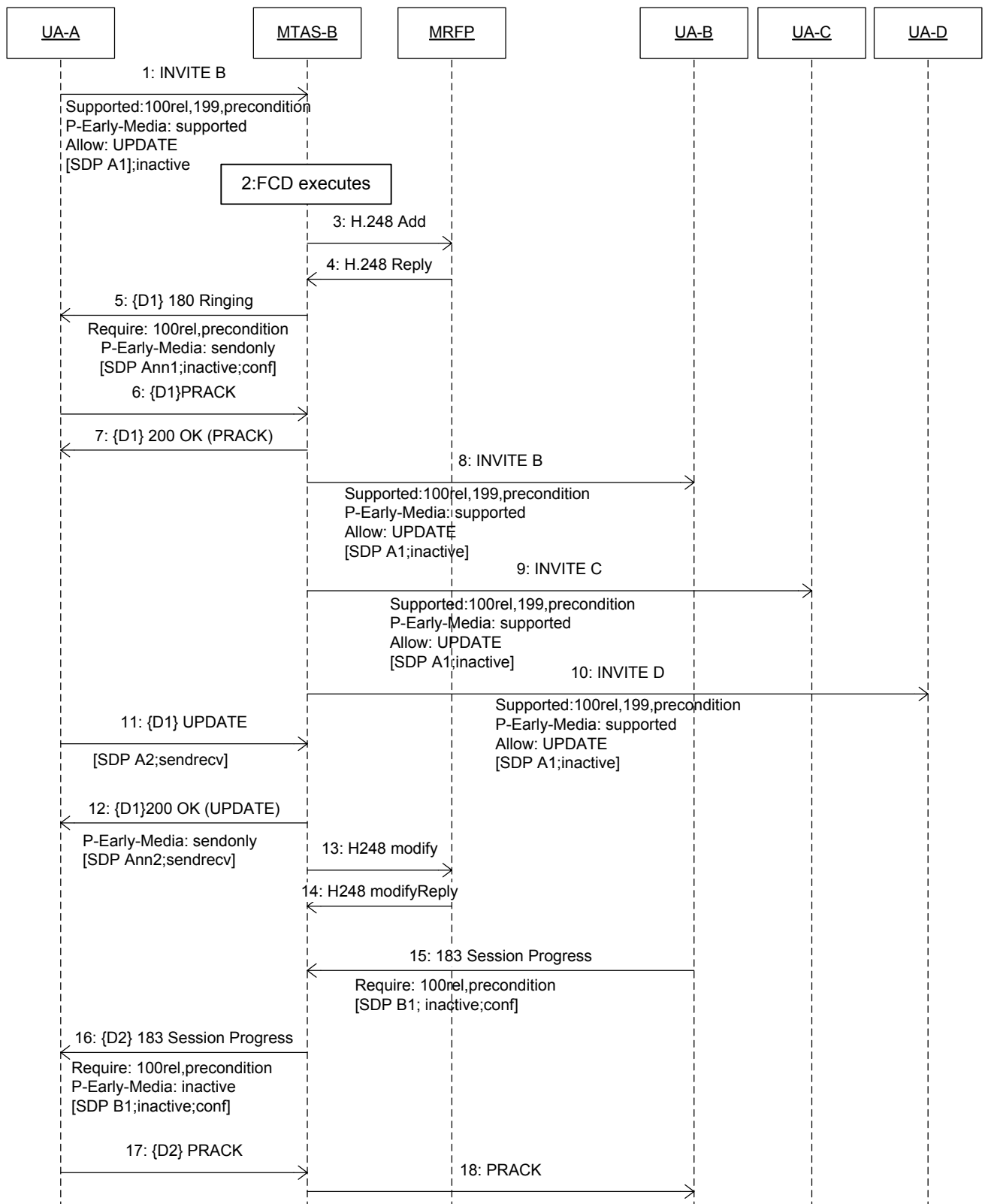


Figure 10 Parallel Ringing – part 1

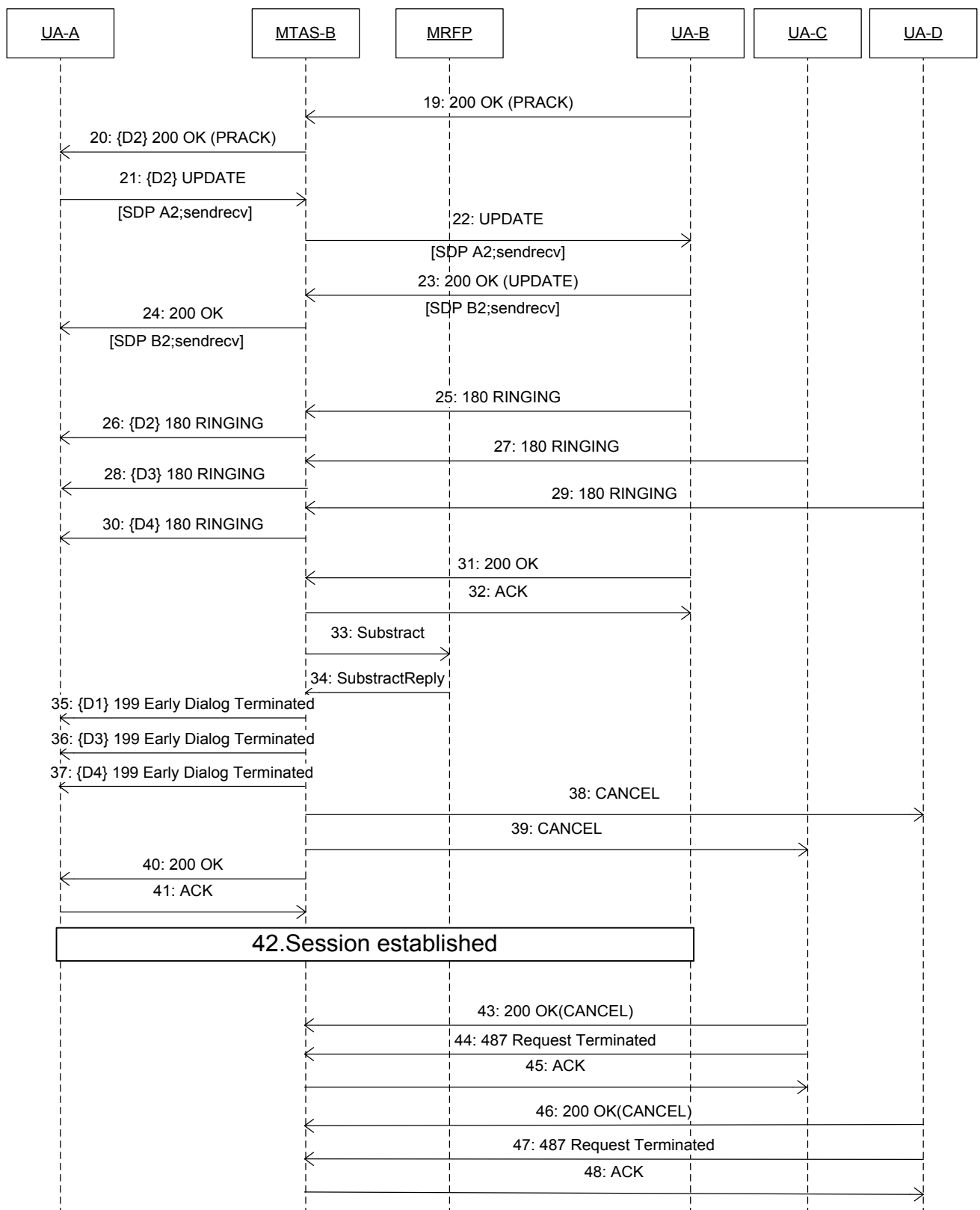


Figure 11 Parallel Ringing – part 2



- 1 The MTAS receives an initial INVITE at the registered terminating port with the identity of the caller (A) in the P-Asserted-Identity header, and the identity of the Primary User, served user (B) in the Request URI.

The INVITE contains an SDP offer (A1) from user A. The SDP has:

- all media set to inactive
 - local current media status set to 'none'
 - remote current media status set to 'none'
 - local desired media status set to 'sendrecv'
 - remote desired media status set to 'sendrecv'
- 2 MTAS executes the FCD service, the served user has activated parallel ringing.
 - a) FCD service checks that there is a valid license to execute the service. A license is present.
 - b) FCD service also checks if the History-Info header does not have more hops than the maximum value set, indicating looping.
 - 3 MTAS initiates reservation of resources in MRFP for playing of the FCD progress announcement. MTAS issues an H.248 Add request towards the MRFP for a new context and to reserve resources and connect media stream for the Caller A. The SDP received in the INVITE for the Caller A is used as the remote descriptor.
 - 4 H.248 AddReply is received which contains the context to be used for FCD progress announcement and the local descriptor.
 - 5 MTAS sends a 180 Ringing to the caller with an SDP answer based upon the local descriptor received from the MRFP. The SDP has:
 - all media set to inactive
 - local current media status set to 'sendrecv' because MRFP has finished resource reservation in B-A direction
 - remote current media status set to 'none'
 - the confirmation of remote media status set to 'sendrecv' so UA-A will send a confirmation when the status of network resources reaches these condition
 - 6 PRACK is received from UA-A.
 - 7 200 OK (PRACK) is returned.



- 8 INVITE is sent to IMS Primary user B. As this is the IMS Primary User the prefixes configured in CM attributes are included in the P-Asserted-Identity and R-URI respectively. Prefixes are only included if the URIs contain global numbers. The SDP offer in this request is the offer received in the INVITE from UA-A. New icid-value is generated by MTAS and included in the P-Charging-Vector header. FCD answer timer is started. The timer value is taken from the ring-period defined in the user data for the parallel distribution. If this is not configured a value of 30 seconds is used.
- 9 INVITE is sent to the Related User C. The SDP offer in this request is the offer received in the INVITE from UA-A. A new icid-value is generated by MTAS and included in the P-Charging-Vector header using the "orig-icid" parameter.
- 10 INVITE is sent to the Related User D. The SDP offer in this request is the offer received in the INVITE from UA-A. A new icid-value is generated by MTAS and included in the P-Charging-Vector header using the "orig-icid" parameter.
- 11 After successful finishing resource reservation A sends an UPDATE containing current local media status attribute set to 'sendrecv'.
- 12 MTAS sends to A 200 OK (UPDATE).
- 13 H.248 modify is sent to the MRFP to commence playing of the FCD progress announcement.
- 14 H.248 ModifyReply is received.
- 15 MTAS receives 183 Session Progress from B with SDP contains:
 - all media set to inactive
 - local current media status set to 'none'
 - remote current media status set to 'none'
 - local desired mandatory media status set to 'sendrecv'
 - remote desired mandatory media status set to 'sendrecv'
 - the confirmation of remote media status set to 'sendrecv' so UA-A will send a confirmation when the status of network resources reaches these condition
- 16 MTAS relays 183 Session Progress to A unchanged except adding P-Early-Media:inactive header field. P-Early-Media suppresses unwanted backward early media.
- 17 MTAS receives PRACK from A
- 18 MTAS relays PRACK to B
- 19 MTAS receives 200 OK(PRACK) from B
- 20 MTAS relays 200 OK(PRACK) to A



21 MTAS receives from A an UPDATE with SDP(A2) contains:

- all media set to sendrecv
- local current media status set to 'sendrecv'
- remote current media status set to 'none'
- local desired mandatory media status set to 'sendrecv'
- remote desired mandatory media status set to 'sendrecv'

22 MTAS relays the UPDATE to B unchanged

23 MTAS receives 200 OK (UPDATE) from B with SDP contains:

- all media set to sendrecv
- local current media status set to 'sendrecv'
- remote current media status set to 'sendrecv'
- local desired mandatory media status set to 'sendrecv'
- remote desired mandatory media status set to 'sendrecv'

It means that all preconditions needed for establishment of connection are met on B and A side.

24 MTAS relays 200 OK (UPDATE) to A unchanged except adding P-Early-Media:inactive header field. P-Early-Media suppresses unwanted backward early media.

25 MTAS receives 180 Ringing from B.

26 MTAS relays 180 Ringing to A.

27 MTAS receives 180 Ringing from C.

28 MTAS relays 180 Ringing to A.

29 MTAS receives 180 Ringing from D.

30 MTAS relays 180 Ringing to A.

31 A 200 OK (INVITE) is received from UA-B.

32 An ACK is sent to UA-B.

33 An H.248 subtract is sent to the MRFP to stop playing the FCD progress announcement and to release the resources.

34 H.248 subtract Reply is received.



- 35 MTAS sends 199 Early Dialog Terminated to A for release resources on an early dialog established for announcement.
- 36 MTAS sends 199 Early Dialog Terminated to A for release resources on an early dialog established to UA-C.
- 37 MTAS sends 199 Early Dialog Terminated to A for release resources on an early dialog established to UA-D.
- 38 CANCEL is sent to UA-C to stop this terminal ringing.
- 39 CANCEL is sent to UA-D to stop this terminal ringing.
- 40 MTAS sends 200 OK (INVITE).
- 41 MTAS receives ACK(OK) from A.
- 42 Session is established.
- 43 A 200 OK (CANCEL) is received from UA-B.
- 44 487 Request Terminated is received from UA-B.
- 45 An ACK is sent to UA-B.
- 46 A 200 OK (CANCEL) is received from UA-C.
- 47 487 Request Terminated is received from UA-C.
- 48 An ACK is sent to UA-C.

2.5.2.2 Serial Ringing

The User Agents, UA, shown in the following sequence may be in the IMS network or any other network, including the PSTN. Additional network nodes for routing to the networks are not shown to simplify the sequence.

The Primary User is configured as non-IMS.

For the following scenario the user has FCD data configured which rings the users in the following order:

Serial ringing of:

- Related User C
- Related User D
- Primary User B

As the Primary User is non-IMS this user cannot be located in the IMS network. The location of the Related Users, C and D, may be in either the IMS or another network.



For each user the answer timer is separately configured and so the timeouts for each INVITE may differ.

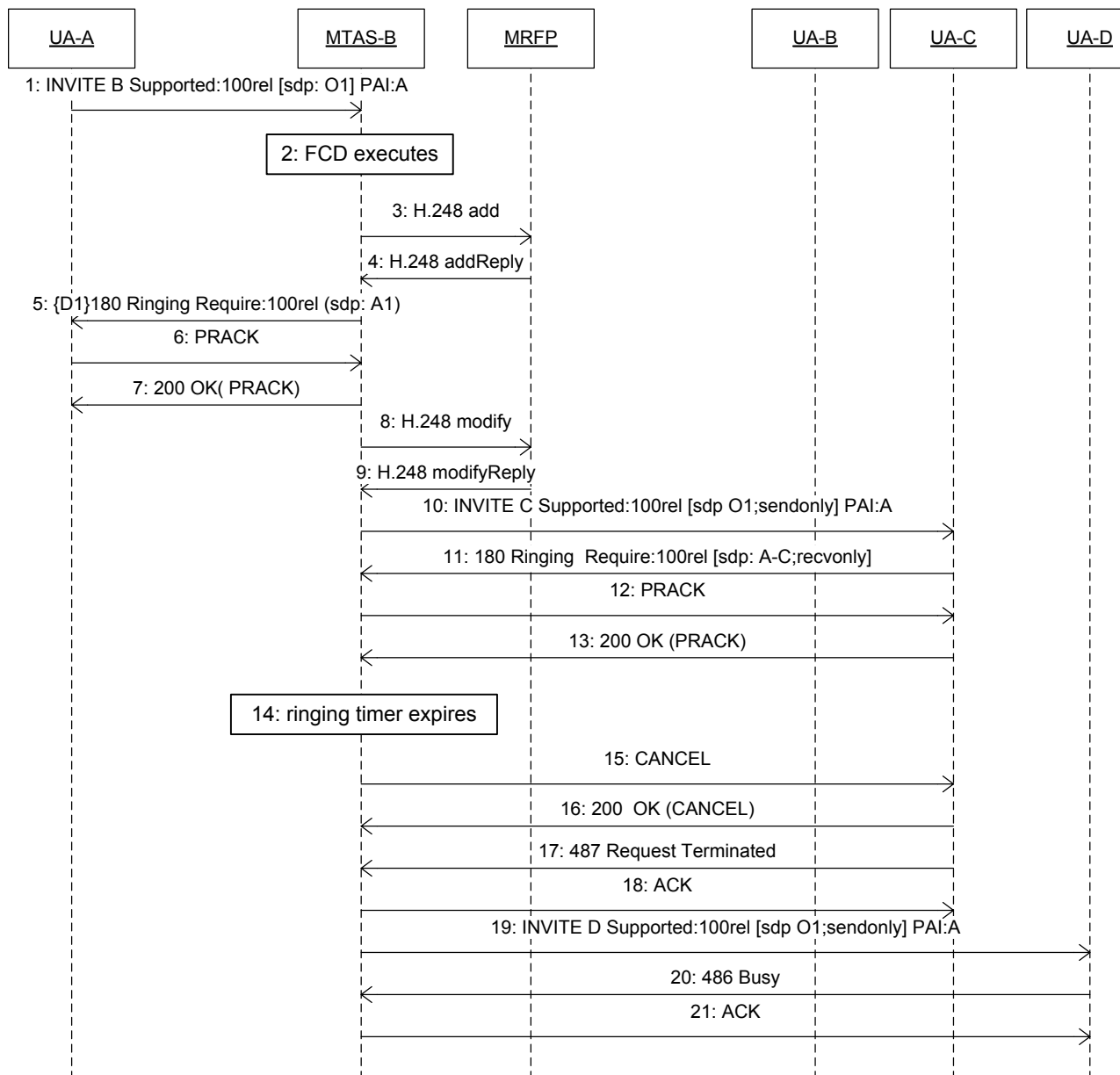


Figure 12 - Last target answers, others time out or reject – part 1

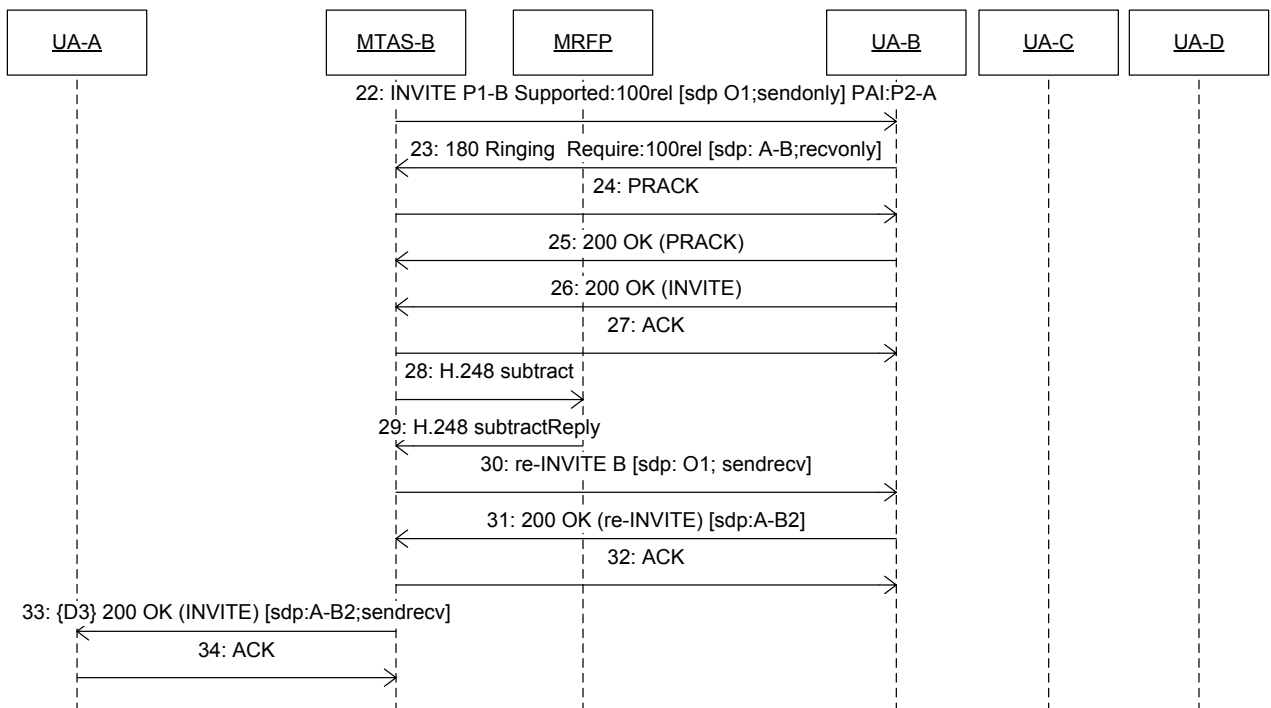


Figure 13 - Last target answers, others time out or reject – part 2

- 1 The MTAS receives an initial INVITE at the registered terminating port, with the identity of the caller (A) in the P-Asserted-Identity header, and the identity of the Primary User, served user (B) in the Request URI. The INVITE contains an SDP offer (O1) from user A.
- 2 MTAS executes the FCD service; the served user has activated serial ringing.
 - a) FCD service checks that there is a valid license to execute the service. A license is present.
 - b) FCD service also checks if the History-Info header does not have more hops than the maximum value set, indicating looping.
- 3 MTAS initiates reservation of resources in MRFP for playing of the FCD progress announcement. MTAS issues an H.248 Add request towards the MRFP for a new context and to reserve resources and connect media stream for the Caller A. The SDP received in the INVITE for the Caller A is used as the remote descriptor.
- 4 H.248 AddReply is received which contains the context to be used for FCD progress announcement and the local descriptor.
- 5 MTAS sends a 180 Ringing to the caller with an SDP answer based upon the local descriptor received from the MRFP.
- 6 PRACK is received from UA-A.
- 7 200 OK (PRACK) is returned.



- 8 H.248 modify is sent to the MRFP to commence playing of the FCP progress announcement.
- 9 H.248 ModifyReply is received.
- 10 INVITE is sent to the Related User C. The SDP offer in this request is the offer received in the initial INVITE from UA-A, except that all media streams are set to 'a=sendonly'. This will prevent any announcements from subsequent nodes. A new icid-value is generated by MTAS and is used in the P-Charging-Vector header. The FCD ringing timer for user C is started. The timer value is taken from the ring-period defined in the user data for the user identity in the serial distribution. If this is not configured a value of 30 seconds is used.
- 11 A 180 Ringing is received from UA-C containing an SDP answer. The SDP answer is recvonly as the SDP offer had all media sendonly.
- 12 A PRACK is sent to UA-C.
- 13 A 200 OK (PRACK) is received from UA-C.
- 14 FCD ringing timer for UA-C expires.
- 15 CANCEL is sent to UA-C to stop this terminal ringing.
- 16 200 OK (CANCEL) is received from UA-C.
- 17 487 Request Terminated is received from UA-C.
- 18 ACK is sent to UA-C.
- 19 INVITE is sent to the Related User D in the same way as done in step 10.
- 20 UA-D responds 486 Busy. The FCD ringing timer for User D is stopped.
- 21 ACK is sent to UA-D.
- 22 INVITE is sent to non-IMS Primary User B. As this is the non-IMS Primary User the prefixes configured in CM attributes are included in the P-Asserted-Identity and R-URI respectively. The SDP offer in this request is the offer received in the initial INVITE from UA-A, except that all media streams are set to 'a=sendonly'. This will prevent any announcements from subsequent nodes. The FCD answer timer for user B is started. The timer value is taken from the ring-period defined in the user data for the user identity in the serial distribution. If this is not configured a value of 30 seconds is used. the calling and called number prefixes included if not disabled with CM. A new icid-value is generated by MTAS and is used in the P-Charging-Vector header.
- 23 A 180 Ringing is received from UA-B containing an SDP answer. The SDP answer is recvonly as the SDP offer had all media sendonly.
- 24 A PRACK is sent to UA-B.
- 25 A 200 OK (PRACK) is received from UA-B.



- 26 A 200 OK (INVITE) is received from UA-B. The FCD ringing timer for user B is stopped.
- 27 An ACK is sent to UA-B
- 28 An H.248 subtract is sent to the MRFP to stop playing the FCD progress announcement and to release the resources.
- 29 H.248 subtract Reply is received.
- 30 re-INVITE is sent to UA-B with an updated SDP offer. The SDP offer is as received from UA-A in the INVITE at step 1.
- 31 200 OK (re-INVITE) is received from UA-B with the SDP answer.
- 32 ACK is sent to UA-B.
- 33 200 OK (INVITE, step 1), on a new dialog to the 180 responses sent at step 5. The new dialog is used even if the 200 OK response was received from the same user as the 180 Ringing response which resulted in the 180 Ringing being sent to the calling user.
- 34 ACK is received from UA-A

If either of the Related Users, C or D, answers the sequence then there will be no attempts to INVITE the other FCD Related or Primary Users.

2.5.2.3 Flexible Ringing

The sequence shown use a Supported: 100rel, 199, precondition header in the received initial INVITE request.

MTAS has the transparent mode switched on.

MTAS has the 199 generation switched on.

The Primary User is configured as IMS.

For the following scenario the served user has FCD data configured which rings the users in the following order:

- Serial ringing of:
 - Related User C
- Parallel ringing of
 - PRIMARY User B
 - Related User D

For the serial and the parallel ringing set the answer timer is separately configured and so the timeouts for each INVITE may differ.



The User Agents, UA, shown in the following sequence may be in the IMS network or any other network, including the PSTN. Additional network nodes used for routing are not shown to simplify the sequence. Caller (UA-A) calls

Related User C then after the related user does not answer the call the call is distributed towards PRIMARY and D. The served user (PRIMARY) answers the call first

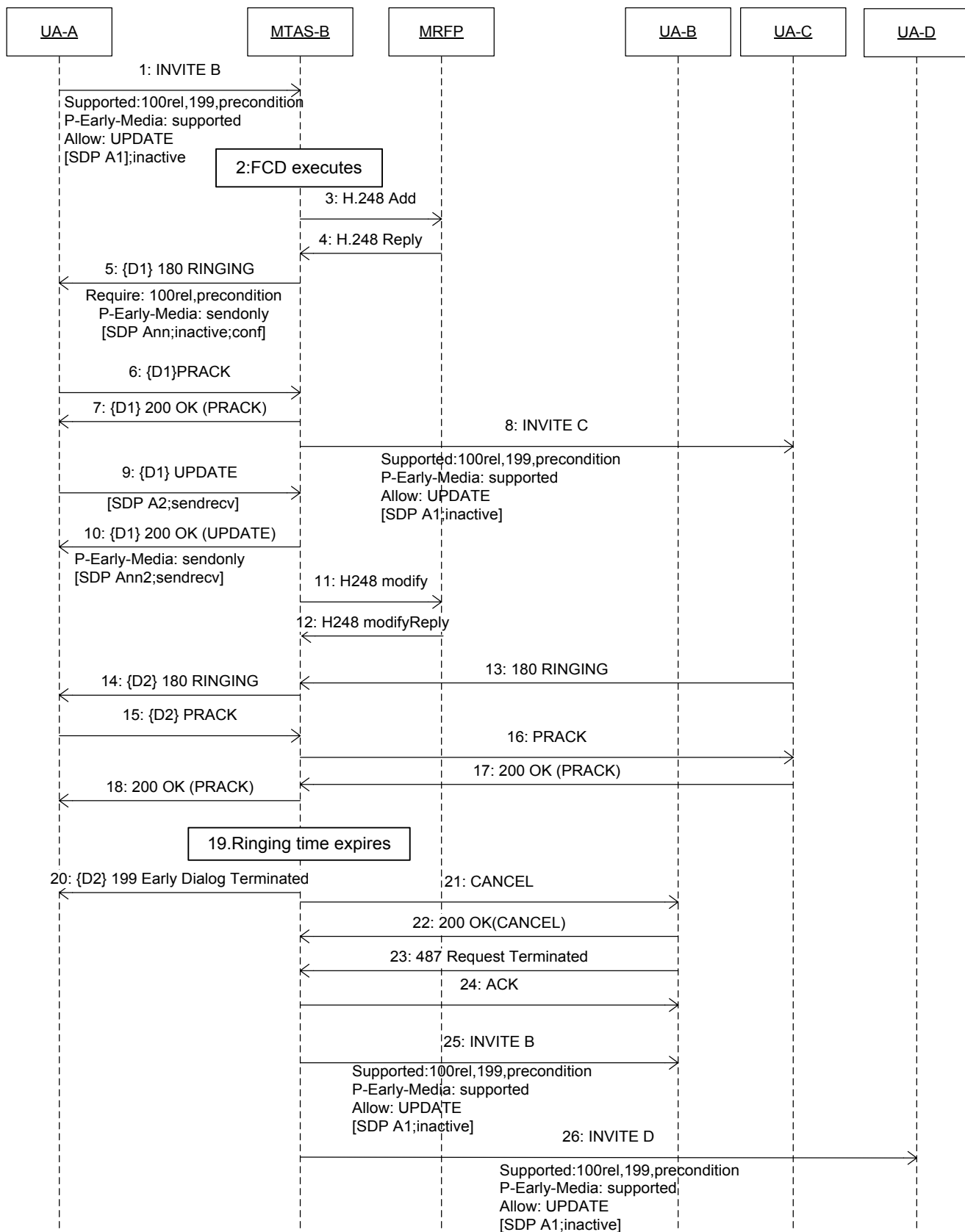


Figure 14 Flexible Ringing – part 1

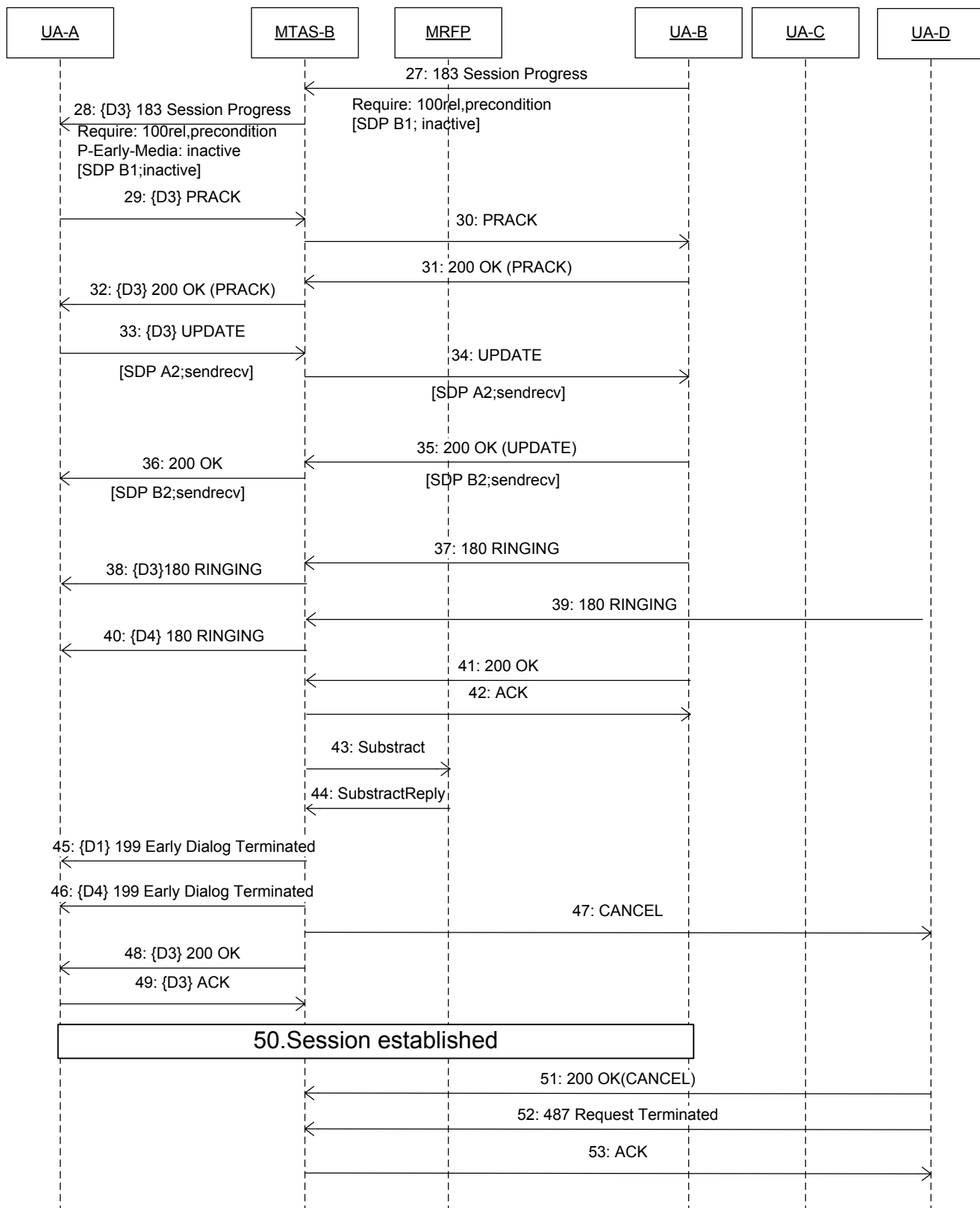


Figure 15 Flexible Ringing – part 2



- 1 The MTAS receives an initial INVITE at the registered terminating port, with the identity of the caller (A) in the P-Asserted-Identity header, and the identity of the Primary User, served user (B) in the Request URI. The INVITE contains an SDP offer (A1) from user A.

The INVITE contains an SDP offer (A1) from user A. The SDP has:

- all media set to inactive
 - local current media status set to 'none'
 - remote current media status set to 'none'
 - local desired media status set to 'sendrecv'
 - remote desired media status set to 'sendrecv'
- 2 MTAS executes the FCD service; the served user has activated serial ringing.
 - a) FCD service checks that there is a valid license to execute the service. A license is present.
 - b) FCD service also checks if the History-Info header does not have more hops than the maximum value set, indicating looping.
 - 3 MTAS initiates reservation of resources in MRFP for playing of the FCD progress announcement. MTAS issues an H.248 Add request towards the MRFP for a new context and to reserve resources and connect media stream for the Caller A. The SDP received in the INVITE for the Caller A is used as the remote descriptor.
 - 4 H.248 AddReply is received which contains the context to be used for FCD progress announcement and the local descriptor.
 - 5 MTAS sends a 180 Ringing to the caller with an SDP answer based upon the local descriptor received from the MRFP.
 - 6 PRACK is received from UA-A.
 - 7 200 OK (PRACK) is returned.
 - 8 INVITE is sent to the Related User C. The SDP offer in this request is the offer received in the INVITE from UA-A. A new icid-value is generated by MTAS and included in the P-Charging-Vector header using the "orig-icid" parameter.
 - 9 MTAS receives from A an UPDATE with SDP(A2) contains:
 - all media set to 'sendrecv'
 - local current media status set to 'sendrecv'
 - remote current media status set to 'sendrecv'



- local desired mandatory media status set to 'sendrecv'
- remote desired mandatory media status set to 'sendrecv'

It means that all preconditions needed for establishment of connection are met on B and A side.

- 10 MTAS sends 200 OK (UPDATE) to A unchanged except adding P-Early-Media:sendonly header field.
- 11 H.248 modify is sent to the MRFP to commence playing of the FCP progress announcement.
- 12 H.248 ModifyReply is received.
- 13 A 180 Ringing is received from UA-C containing an SDP answer. The SDP answer is recvonly as the SDP offer had all media sendonly.
- 14 MTAS sends 180 Ringing to user A
- 15 MTAS receives PRACK from user A
- 16 A PRACK is sent to UA-C.
- 17 A 200 OK (PRACK) is received from UA-C.
- 18 MTAS sends 200 OK(PRACK) to A
- 19 FCD ringing timer for UA-B expires.
- 20 MTAS sends 199 Early Dialog Terminated to A for release resources on an early dialog established to UA-B.
- 21 CANCEL is sent to UA-C to stop this terminal ringing.
- 22 A 200 OK (CANCEL) is received from UA-B.
- 23 487 Request Terminated is received from UA-B.
- 24 An ACK is sent to UA-B.
- 25 INVITE is sent to IMS Primary user B. As this is the IMS Primary User the prefixes configured in CM attributes are included in the P-Asserted-Identity and R-URI respectively. Prefixes are only included if the URIs contain global numbers. The SDP offer in this request is the offer received in the INVITE from UA-A. New icid-value is generated by MTAS and included in the P-Charging-Vector header. FCD answer timer is started. The timer value is taken from the ring-period defined in the user data for the parallel distribution. If this is not configured a value of 30 seconds is used.
- 26 INVITE is sent to the Related User D. The SDP offer in this request is the offer received in the INVITE from UA-A. A new icid-value is generated by MTAS and included in the P-Charging-Vector header using the "orig-icid" parameter.
- 27 MTAS receives 183 Session Progress from B with SDP contains:



- all media set to inactive
 - local current media status set to 'none'
 - remote current media status set to 'none'
 - local desired mandatory media status set to 'sendrecv'
 - remote desired mandatory media status set to 'sendrecv'
 - the confirmation of remote media status set to 'sendrecv' so UA-A will send a confirmation when the status of network resources reaches these condition
- 28 MTAS relays 183 Session Progress to A unchanged except adding P-Early-Media:inactive header field. P-Early-Media suppresses unwanted backward early media.
- 29 MTAS receives PRACK from A
- 30 MTAS relays PRACK to B
- 31 MTAS receives 200 OK(PRACK) from B
- 32 MTAS relays 200 OK(PRACK) to A
- 33 MTAS receives from A an UPDATE with SDP(A2) contains:
- all media set to sendrecv
 - local current media status set to 'sendrecv'
 - remote current media status set to 'none'
 - local desired mandatory media status set to 'sendrecv'
 - remote desired mandatory media status set to 'sendrecv'
- 34 MTAS relays the UPDATE to B unchanged
- 35 MTAS receives 200 OK (UPDATE) from B with SDP contains:
- all media set to sendrecv
 - local current media status set to 'sendrecv'
 - remote current media status set to 'sendrecv'
 - local desired mandatory media status set to 'sendrecv'
 - remote desired mandatory media status set to 'sendrecv'

It means that all preconditions needed for establish connection are meet on B side.



- 36 MTAS relays 200 OK (UPDATE) to A unchanged except adding P-Early-Media:inactive header field. P-Early-Media suppresses unwanted backward early media.
- 37 MTAS receives 180 Ringing from B.
- 38 MTAS relays 180 Ringing to A.
- 39 MTAS receives 180 Ringing from D.
- 40 MTAS relays 180 Ringing to A
- 41 A 200 OK (INVITE) is received from UA-B.
- 42 An ACK is sent to UA-B.
- 43 An H.248 subtract is sent to the MRFP to stop playing the FCD progress announcement and to release the resources.
- 44 H.248 subtract Reply is received.
- 45 MTAS sends 199 Early Dialog Terminated to A for release resources on an early dialog established for announcement.
- 46 MTAS sends 199 Early Dialog Terminated to A for release resources on an early dialog established to UA-D.
- 47 CANCEL is sent to UA-D to stop this terminal ringing.
- 48 MTAS sends 200 OK (INVITE).
- 49 MTAS receives ACK(OK) from A.
- 50 Session is established.
- 51 A 200 OK (CANCEL) is received from UA-D.
- 52 487 Request Terminated is received from UA-D.
- 53 An ACK is sent to UA-D.



2.5.3 Addressing terminals of the IMS Primary User

It is possible to configure a list of “Busy Everywhere” SIP responses at node level, so that, in case of Application Server Controlled Forking, on receipt of such a response from any device of a served user, call legs to all remaining devices of the served user within the same session will be cancelled. Busy Everywhere response codes are determined by `mtasFcdBusyEverywhereResponses` CM attribute. This attribute can not contain the same values as `mtasFcdBusyResponses` or `mtasFcdNotReachableResponses`. The only exception is 486 response code which does not conflict between these CM attributes. But this response can be treated as Busy Everywhere only if SIP message contains Reject reason text “call rejected by user”. After all devices respond 200 OK and 487 Request Terminated for CANCEL messages, MTAS sends 480 Temporary Unavailable or 486 Busy Here (depending on the `mtasFcdBusyIndication`) to related user.

2.5.3.1 Common device data

In all of the distribution schemes (serial, parallel and flexible), when the target device is a specific terminal of the IMS Primary User, then a new Accept-Contact header line set to a device in the common device data is included in the INVITE sent to the IMS Primary User in order to identify the specific terminal.

The target list may contain more than one terminal of the IMS Primary user. In such case, each terminal is addressed with an INVITE from MTAS.

A valid license for AS Controlled Forking is needed for the usage of terminal selectors configured in the common device data.

2.5.3.2 FCD to Primary User’s devices

The FCD to Primary User’s Devices function allows for parallel call distribution to all (mobile and fixed) currently registered terminals of a served user when either no rule has been provisioned or by means of PRIMARY keyword in FCD rule. It is also possible to address a mobile terminal only, using PRIMARY_MOBILE keyword or fixed terminal only using PRIMARY_FIXED keyword.

Devices are addressed by adding sip.instance and optional mobility selector to the caller preferences in the Accept-Contact header for each terminating call leg. Values of sip.instance and mobility selector are based on contact data cached during registration.

This is a “dynamic” alternative of AS Controlled Forking, which eliminates a need to manually configure a “static” list of individual device targets in the User Common Data and FCD rules.

The “Active Call Preference for Fixed Devices” extension lets a new incoming call be distributed only to served user’s fixed devices with ongoing active calls and a mobile device.



A specific application of this function is communication distribution to ICS user's devices in Fixed Mobile Convergence (FMC) scenarios. By comparing a terminal instance received in the Accept-Contact header with stored contact data, SCC AS can distinguish calls distributed to mobile (VoLTE) devices from calls distributed to fixed devices and apply T-ADS to the former only.

2.5.4 Charging

Originating offline and/or online charging are performed on outgoing legs to a non-IMS Primary User, a Related User or Diverted-To User, if applicable.

If Multi-Device Charging has been enabled in a charging profile, then terminating offline and/or online charging is performed on outgoing legs to served user's devices. Charging profiles may be assigned per user, so that for some served users the Multi-Device Charging may be enabled, and for other users registered on the same node the Multi-Device Charging may be disabled.

Multi-Device Offline Charging enables an operator to collect offline charging information per device in AS Controlled Forking scenarios, so that he/she can provide served users with a call log including information about individual devices ringing at an incoming call.

Multi-Device Online Charging distinguishes between mobile device credit control and fixed device usage control. This permits fixed devices to be used although no credit exist for the mobile device.

Although Multi-Device Online/Offline Charging is intended mainly for Application Server Controlled Forking scenarios, it is a separate feature and can be configured independently on any other service settings.

It is possible to configure terminating MTAS to play the announcement specified by OCS to the caller, when OCS denies an incoming call to be terminated to any target, so that the caller is notified that the call denial reason lies on subscriber's side and is not related to any problem with operator's network. This feature uses Subscriber Credit Notification code but it is independent of `mtasChargingSubscriberCreditNotification` CM setting and license for Subscriber Credit Notification feature.

2.5.5 Service Interaction

The served user's outgoing originating services are executed when FCD service sends an outgoing INVITE request to the non-IMS Primary User or to a Related User.

The terminating AS of the served user becomes a transit AS when the 200 OK, answer, to INVITE request is from the non-IMS Primary User or a Related User. Other services will act differently for this transit AS compared to originating or terminating AS.



2.5.5.1 Communication Diversion

When `mtasFcdCDivInvocationSequenceControl` attribute is set to 1, FCD is invoked before CDIV in the terminating MTAS.

When `mtasFcdCDivInvocationSequenceControl` is 0, CDIV is invoked before FCD. It is highly recommended to set the value always to 1.

All FCD-CDIV or CDIV-FCD interworking can be achieved by FCD only rules as FCD supports the same conditions and actions as CDIV.

2.5.5.2 Identity Presentation

A P Asserted-Identity header field received by the FCD AS in a final response is passed unmodified to the originating entity irrespective of whether the response was from the IMS Primary User, non-IMS Primary User, Related User or Diverted-To User.

2.5.5.3 Communication Barring

Related User identities cannot be an identity which would be barred by OCB or included in the CDIV black list

2.5.5.4 Communication Completion

A user can only be provisioned with the FCD service if that user has the Communication Completion Opt-Out service provisioned for CCBS, CCNR and CCNL.

The FCD service removes the call completion possible indications from provisional and final responses from the non-IMS Primary User and from the Related Users.

2.5.5.5 Carrier Select and Pre-Select

FCD allows the Related User identity to include a Carrier Select code.

Initial INVITE requests sent by the FCD service to the non-IMS Primary User and Related Users are subject to the Carrier Pre-Select

2.5.5.6 Real Time Tariff Information

RTTI information is passed to the Online Charging System in a Credit Control Update Request for the outgoing charging for the individual leg of the FCD call to non-IMS Primary User or a Related User in situation when online charging is applicable for that leg and real-time tariff information was received in a provisional response to the initial INVITE.

2.5.5.7 Malicious Communication Identification

The IMS Primary user's call details may be stored by Temporary MCID, non IMS Primary or non-IMS user's details.



No interaction with Permanent MCID

2.5.5.8 Call Admission Control

FCD calls will only count as a single terminating call for User Call Admission Control

2.5.5.9 Subscriber Data

FCD to Primary User's Devices is dependent on registered contacts information, to be able to distinguish the device identifier (sip.instance) and type of device (mobile/fixed).

Mobility (mobile or fixed) classification of devices during registration may be based on either registered contact's feature tags or P-Access-Network-Info (PANI) header.

2.5.5.10 IMS Centralized Services

For MMTel AS collocated with SCC AS when registered contacts information are available to FCD in the Subscriber Data storage, the FCD calls can be distributed to ICS user's devices with caller preference on the sip.instance feature tag. The calls to fixed devices are ignored by SCC AS T-ADS service in this case.

For standalone MMTel AS, it must be ensured that the iFC for MMTel AS in HSS is configured to trigger initial registration, reregistration, and deregistration and to include the user REGISTER request and 200 OK response in the message/sip body of the third-party registration. The mtasSubsDataCacheContactData attribute must also be enabled.

2.5.6 Self administration

- List of users (targets) used by FCD
- List of users and list of devices with terminal selectors common for more than one service
- absolute time intervals
- parallel/serial/flexible ringing
- announcement played to the caller during the distribution
- activation/deactivation
- divert-primary with forward-to target
- Conditions can be used, and even combined in the rules:
 - o validity



MTAS will distribute the communication when the current time is within the validity period. End-user can define one or more time periods. Time periods are defined by a starting time and an ending time.

- o invalidity
MTAS will distribute the communication when the current time is NOT within the invalidity period. End-user can define one or more time periods. Time periods are defined by a starting time and an ending time.
- o rule-deactivated
This condition always evaluates to false. This can be used to deactivate a rule, without losing information. By removing this condition the rule can be activated again.
- o valid-periods
The valid-periods condition allows assembly of complex time condition based upon several calendar sub-conditions (times of day; days of the week; calendar months; calendar weeks; private and public holidays; daily, weekly and monthly repetitions etc.). A sub-condition evaluates to “true”, if any of the elements within the sub-condition evaluates to “true”.
MTAS will distribute the communication when the current time matches to all included sub-conditions.
- o busy
MTAS will distribute the communication when the served user responds with busy.
- o not-registered
MTAS will distribute the communication when the served user is not registered.
- o not-reachable
MTAS will distribute the communication when there is a signaling channel outage during session setup to the served user's equipment and the served user is registered.
- o presence-status
MTAS will distribute the communication when the served user's current presence activity status contains a value equal to the value set for this condition.
- o identity
MTAS will distribute the communication when calling user's identity matches with the value of the identity element.
- o anonymous
MTAS will distribute the communication when the calling party is anonymous.
- o served-identity



MTAS will distribute the communication when P-Served-User matches the value of the served-identity element. If P-Served-User is not supported in MTAS or is absent then Request-URI is used for matching.

- o media
MTAS will distribute the communication when the calling party's offered media line(s) matches with the media line(s) expected in this condition.
- o no-answer
MTAS will distribute the communication when the served user does not answer.
- o in-sip-request
MTAS will distribute the communication when particular headers or header parameters of a SIP request match given regex patterns.

2.5.7 Configuration

- Prefix handling to facilitate service domain selection/routing of non IMS primary user FCD execution in MTAS
- Allowed consecutive FCD for the same call (loop detection)
- Single and Multi-Media Announcements
- Terminal selector prefix, common for more than one service
- Enable distribution of call to Primary User's devices
- Additional mobile and fixed terminal selectors
- Active Call Preference for Fixed Devices
- Error responses when busy, busy everywhere or not-reachable conditions met
- Timer for the no-answer condition
- Configuration of CDS peer information
- Configuration of CB
- Configuration of Caller Preference Filtering

2.5.8 Performance Management

- Failed FCD attempts
- Successful attempts
- Ongoing parallel FCD session



- Successful FCD and FCDDP attempts

2.5.9 Fault Management

- For information on the alarm, refer alarm OPI [54], [55] and [56].

2.6 Communication Waiting

2.6.1 Description

The Communication Waiting (CW) service enables a subscriber to be informed, while engaged in a communication session, that there is another communication waiting. The subscriber then has the choice of accepting, rejecting or ignoring the waiting communication (as per basic call procedures).

Note that once a waiting session has been accepted, the CW function is inactive, and does not provide features for switching between the existing session and the waiting session. This is provided by the served user's UA. The UA may make use of the HOLD service in order to place an existing session on hold while the waiting communication is answered.

MTAS supports 3 separate modes for communication waiting:

Normal Mode Supports Network and User determined CW.

Alternate Mode 1 Supports Network and User determined CW and differs from Normal mode in the announcement signaling.

Alternate Mode 2 Supports User determined CW, signaled by a 182 Queued provisional response.

2.6.1.1 Normal mode and Alternate mode 1

In Normal mode and Alternate mode 1, the CW function works in conjunction with the CAC function, which determines if the served user is in the Approaching Network Determined User Busy (ANDUB) state. If so, and the served user has an active subscription to CW, the CW function includes a Communication Waiting Active (CWA) indicator in the INVITE request.

The CW function then waits for a response to the INVITE request.

Upon receipt of a 180 Ringing provisional response to the INVITE request the CW function determines if CW has been used for the subscriber with the CW service active. This can be because:

1. ANDUB for the served user was determined at the outgoing INVITE.

OR

2. ANDUB was not determined for the served user but a CWU Alert-info header is received in the 180 Ringing response (User CW).



Type of Indication is based on value of `mtasCwIndication` configuration parameter. Can be either:

XML CW Indication (`mtasCwIndication = 0`)

“P-Service-Indication=CW” Sip header (`mtasCwIndication = 1`)

Both XML and P-Service-Indication (`mtasCwIndication = 2`)

2.6.1.2 Alternate mode 2

In Alternate Mode 2 the INVITE request is sent to the served user without change.

The CW function then waits for a response to the INVITE request.

Upon receipt of a 182 Queued provisional response to the INVITE request, the CW function determines that CW has been used for the subscriber with the CW service active.

2.6.2 Example Call Flow

Following figure shows call flow of Communication Waiting in Normal mode. In this flow served user accepts waiting communication during announcement.

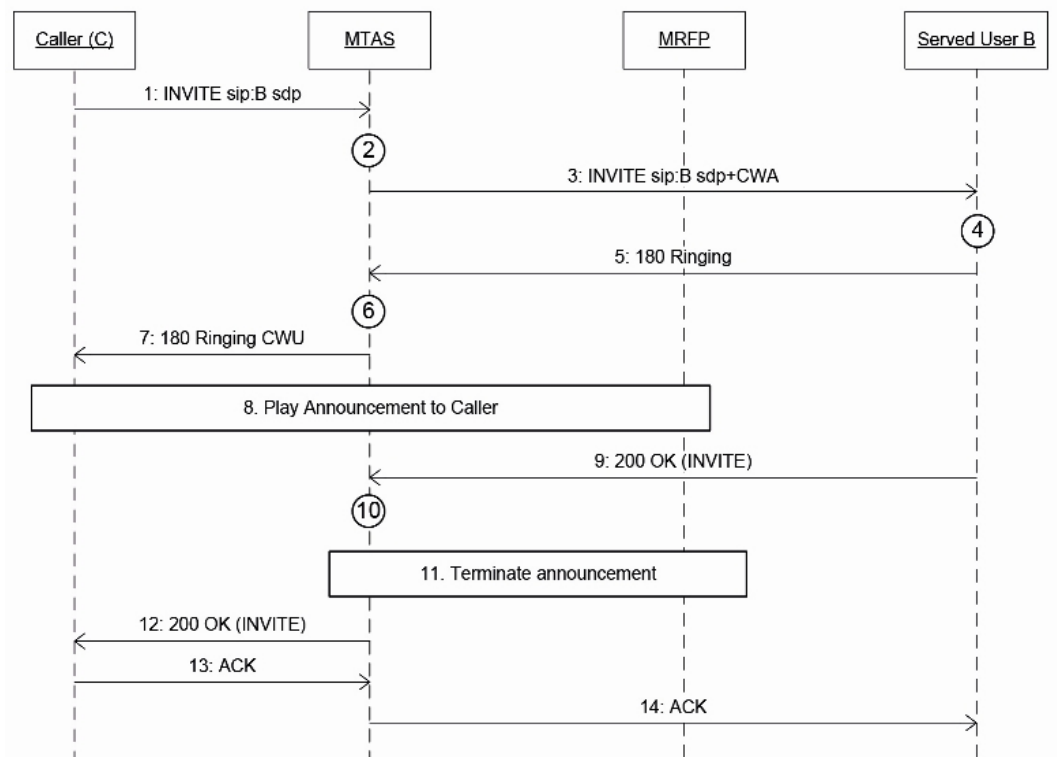


Figure 16 - Served user accepts waiting communication during announcement – ANDUB determined



1. An INVITE is received from the caller for the served user. Note that the INVITE may contain a request for resource reservation.
2. The CAC function determines that the new session puts the served user in the ANDUB (Approaching/Network Determined User Busy) state. The CW function detects the fact that the served user has an active subscription to CW. The CW function adds a CWA indication to the INVITE. Note that the CW function takes no account of the type of communication session that is being requested when deciding whether or not to insert a CWA indication. It only takes account of the active status of the subscription to CW
3. The INVITE containing the CWA indication is sent to the served user.
4. The terminating UE alerts the served user to the presence of the waiting communication.
5. A 180 Ringing is sent back to the terminating MTAS to indicate that the served user is notified of the new communication. The terminating UE may respond with a simple 180 Ringing without a CWU indication.
6. The CW function checks the provisional response. Anything other than a 180 Ringing will be passed back transparently. If the provisional response is a 180 Ringing then CW is determined to have been used as a CWA was sent in the outgoing INVITE, step 3. If CWU is not present, a CWU indicator is added to the 180 Ringing before it is passed on towards the caller. If CW has been used then CW enters an alerting phase. The CW function starts the timer on the alerting phase. The fact that CW has been used is recorded for charging purposes. A count of dialogs on which a CW has been used is incremented and the CW function records the identity of the dialog on which CW has been used.
7. The 180 Ringing is sent back to the caller with the CWU indication present.
8. The sub function Play Announcement is called. A video, audio or Audio/Video announcement is played to the caller if there is a suitable media stream available, otherwise no announcement is played.
9. During the playing of the announcement, the served user accepts the waiting communication and sends 200 OK.
10. The CW function handles the received 200 OK received on the same dialog as the 180 Ringing provisional response. Since the identity of the dialog on which the 200 OK was received matches a dialog on which CW has been used. The CW function stops the timer on the waiting communication.
11. The CW function calls the Terminate Announcement sub function to terminate the announcement.
12. The 200 OK is sent back to the caller.
13. An ACK to the 200 OK is received from the Caller.



14. The MTAS sends the ACK to the served user.

The session is now established.

2.6.3 Service Interaction

CW interacts with following services.

2.6.3.1 CAC: CW Normal mode and Alternate mode 1

The Communication Waiting service is dependent on the CAC service. That is, it is not possible to provision a user with the Communication Waiting service unless that user also has the CAC service provisioned with a waiting limit greater than zero.

The Communication Waiting service is invoked for a terminating served user when the CAC service determines that the served user is in the ANDUB state, or if a 180 Ringing response is received with a CWU indicator.

2.6.3.2 CAC: CW Alternate mode 2

There is no dependency between CAC and CW, when CW is in Alternate mode 2.

2.6.3.3 Charging: CW Normal mode and Alternate mode 1

The MTAS records when CW is used. The use of CW is recorded for charging purposes on the first 180 Ringing received where the CW function determines that CW has been used, i.e. ANDUB or containing a CWU indication only. Any subsequent CWU indications are ignored. The MTAS includes this information in the ACR/CCR generated for the session on which Communication Waiting was used. This is reported even for waiting communications that are not accepted by the served user.

2.6.3.4 Charging: CW Alternate mode 2

In Alternate mode 2 the usage of CW is recorded in the ACR on the first 182 queued.

2.6.3.5 CFU

CFU takes precedence over CW and will result in CW not being invoked.

2.6.3.6 CFNL

CFNL takes precedence over CW and will result in CW not being invoked.



2.6.3.7 CFB

If the response to an INVITE containing a CWA is 486 (Busy here), 600 (Busy Everywhere) or 603 (Decline) then CW takes no action other than to pass the response backwards towards the caller. If the served user also has CFB active and CDiv CM data has been set for the particular response code, then this will be invoked and the communication will be diverted.

2.6.3.8 CFNR

If the subscriber with CW also has CFNR active, then the action depends on the relative values of the CW Alerting timer and the timer on CFNR. If the value of the CW Alerting timer is less than the value of the CFNR timer, and the subscriber does not respond to a waiting communication before the CW Alerting timer expires, then the waiting communication is cancelled and CFNR will not be invoked. If the value of the CW Alerting timer is greater than the CFNR timer and the subscriber does not respond to a waiting communication before the CFNR timer expires, then CFNR will be invoked and the waiting communication cancelled.

2.6.3.9 Communication Deflection

If CD is active at the nodal level, then if a served user responds to a waiting communication with 302, CW will be stopped.

2.6.3.10 Communication Barring

If the CW subscriber has both ICB and CW active, ICB will be invoked prior to CW being invoked. If the communication is barred then CW will not be invoked.

2.6.3.11 CONF: CW Normal mode and Alternate mode 1

The terminating MTAS will receive an INVITE from the focus for the served user. The CAC function determines that the new session puts the served user in the ANDUB state. The CW function adds a CWA indication to the INVITE. The intended conference participant is alerted to the waiting communication. A 180 Ringing response which may contain a CWU indicator is returned. The Focus will send a NOTIFY of the receipt of 180 Ringing to the Conference Creator (CC). The NOTIFY will not contain the CWU indication. If the CW announcement is enabled, then the terminating MTAS will attempt to play an announcement to the caller. The net result is that the CC will receive no indication that the intended participant is busy but does have CW.



2.6.3.12 CONF: CW Alternate mode 2

The terminating MTAS will receive an INVITE from the focus for the served user. If the terminating MTAS receives a 182 Queued response it will convert it to a 180 Ringing. If the CW announcement is enabled, then the terminating MTAS will attempt to play an announcement to the caller (the conference focus in this case), sending the announcement SDP answer in the 180 Ringing. The net result is that the CC will receive no indication that the intended participant is busy but does have CW.

2.6.3.13 Serial and parallel ringing

If the CW subscriber also has serial / parallel ringing, then an INVITE containing a CWA indication will be forked by the terminating CSCF to all terminals that are currently registered. Depending on the busy / free state of each of the terminals, and whether they support the use of the CWA indicator or not, this will result in multiple provisional responses being received by CW, some of which may contain a CWU indication and some of which may not.

The CW timer starts when the first 180 Ringing response to an INVITE with CWA has been sent or when the first 180 Ringing containing a CWU indication is received and stops when the service regardless of how many 180 Ringing with or without CWU indications are received after the first one.

2.6.3.14 Communication Completion: CW Normal mode and Alternate mode 1

Communication Waiting is not offered as a result of a CC Call.

2.6.3.15 Communication Completion: CW Alternate mode 2

The Called Party responds with a 182 Queued provisional response to a CC INVITE. The CW service recognizes that this is a CC INVITE, so does not consider this as a waiting call. The 182 Queued provisional response is changed to be a 180 Ringing provisional response, without a CWU indication. The terminating MTAS CC Monitor forwards the 180 Ringing toward the CC Caller.

2.6.4 Configuration

- CW Activation (active/ disable)
- Audio Announcement activation/ deactivation
- Video Announcement activation/ deactivation
- CW Alert Timer, the time interval within which the served user must respond to a Communication Waiting indication before the waiting session is cancelled.
- CW Operate Mode (Normal, Alternate 1, Alternate 2)
- CW Indication (XML CW Indication, P-Service-Indication sip header, both XML and SIP header)



2.6.5 Performance Management

- Number of waiting communications that have been accepted
- Number of waiting communications that failed because the served user rejected
- Number of waiting communications that have exceeded the timeout on the alerting phase
- Number of waiting communications that have been cancelled
- Number of waiting communications that have been used
- Number of waiting communications attempts

2.7 Communication Completion (CCxx)

2.7.1 Description

The Communication Completion service in MMTel AS supports Communication Completion on Busy (CCBS), Communication Completion on No Reply (CCNR) and Communication Completion on Not Logged-in (CCNL).services. Also CCBS,CCNR and CCNL Opt-Out services are supported.

CCBS allows a caller who has attempted to make a call to a busy subscriber, to activate a CC request against that subscriber and initiates a CC Call after the called subscriber becomes not busy.

CCNR allows a caller who has attempted to make a call to a subscriber who does not answer the call, to activate a CC request against that subscriber and initiates a CC Call when the called party becomes not busy after having initiated and completed a new call.

CCNL allows a caller who has attempted to make a call to an unregistered subscriber, to activate a CC request against that subscriber and initiate a CC Call after the called subscriber registers to the system.

After CCNL, CCBS or CCNR has been activated a CCxx request is stored in a queue on the originating side related to the calling user and on the terminating side related to the called user. The CCxx request is removed from the A and B-queues when a CC call has been successfully performed or when the CCxx service timer expires.

For both CCBS and CCNR, communication is completed after the called user becomes not busy. For CCNL, communication is completed after the called user becomes registered. After the called user becomes not busy (CCBS and CCNR) or registered (CCNL), the network originates a CC Recall to the caller. After the caller answers, the network establishes a CC Call to the original called user on behalf of the caller.



If the CC Call results in busy, no reply or call to an unregistered user, the caller may activate a CCBS, CCNR or CCNL request in the CC Call the same way as activating the CCBS, CCNR or CCNL request in the original call.

The CCBS, CCNR and CCNL Opt-Out services makes it possible for users to be protected from calling users to activate CCXX services against them..

The Communication Completion service is divided into two main functions.

1) Communication Completion Agent

The CC Agent function is located at the originating MMTel AS and is associated with the CCxx Caller. The CC Agent is responsible for maintaining a list of CC Invocation requests that have been made by the CCxx Caller against different Called parties.

The CC Agent is also initiated when the called party becomes available to receive a CC call. The CC Agent function is responsible for performing the CC Recall to the caller and to initiate the setup of the CC Call to the called party

2) Communication Completion Monitor

The CC Monitor function is located at the terminating MMTel AS and is associated with the CCxx Called Party. The CC Monitor is responsible for maintaining a list of CC Invocation requests that have been made by different CCxx Callers against the CCxx Called Party.

For CCBS and CCNR the CC Monitor depends on the CAC service to determine when the CCxx Called Party becomes available to receive a CC Call. When the CCxx Called Party ends a call, the CC Monitor is triggered to check the current user availability and send a ready notification to the 3PCC function.

For CCNL the CC Monitor supervises when the user becomes registered to determine when the CCNL Called Party becomes available to receive a CC Call.

During a CC Recall, the 3PCC function can determine that the CCxx Caller is busy either through the CAC service or by a busy response from the CCxx Caller. If the CCxx Caller is busy, then the CC Recall is aborted and the request is suspended or revoked, depending on configuration data.

When a call ends against a monitored user (CCBS and CCNR) or a monitored user registers (CCNL), an idle guard timer is run in case that user makes another call immediately. During this period and the period of the subsequent CC Recall and CC Call (assuming no outgoing call is made), all other terminating calls to that user are rejected with a busy response.



2.7.2 Example Call Flow

2.7.2.1 Invocation

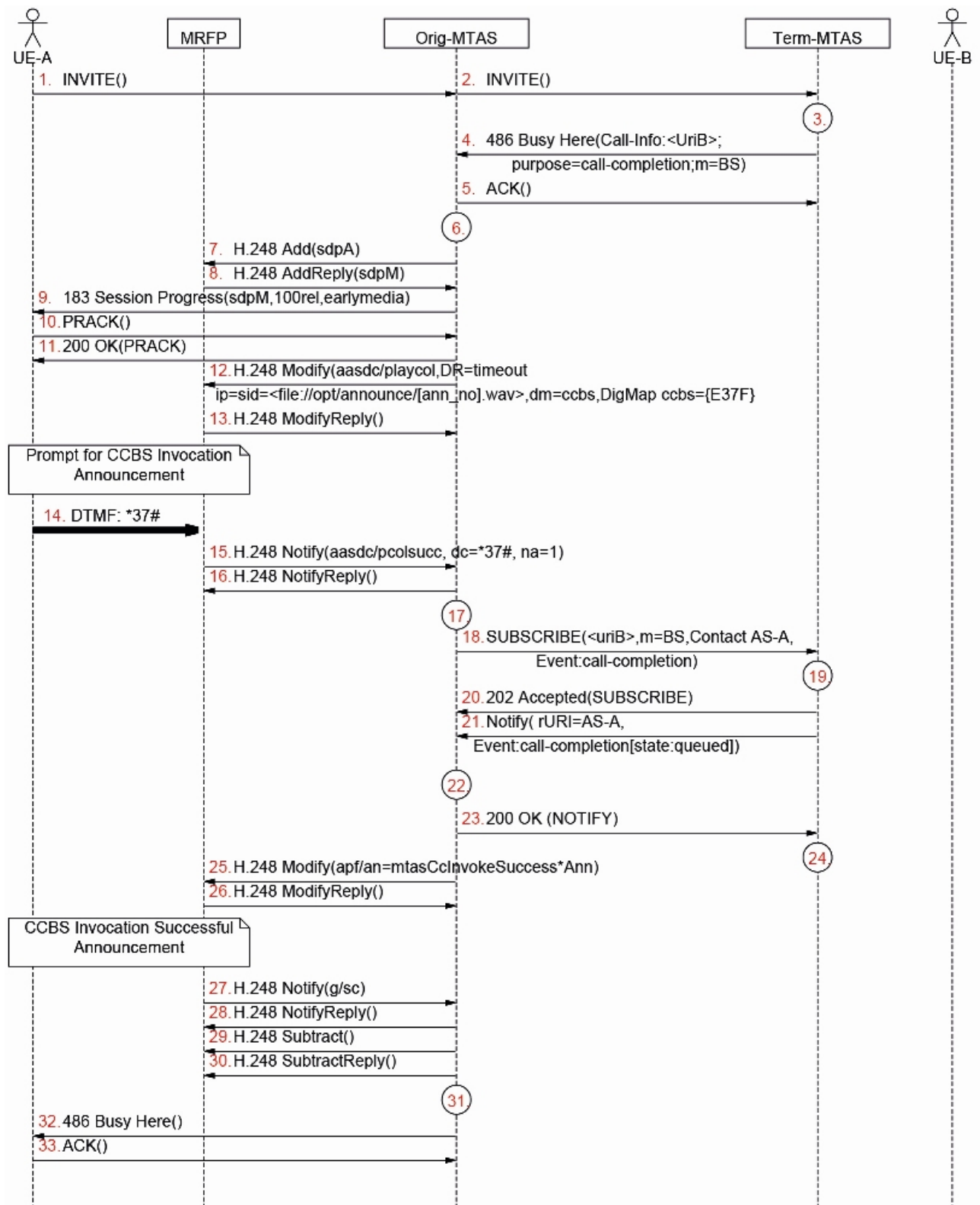


Figure 17 - CCBS Indicated, Invocation Successful



1. An originating INVITE is received at the MMTel AS of the CCxx Caller. The originating MMTel AS CC Agent checks that the user is provisioned with the CCBS service and stores a copy of the INVITE.
2. The INVITE is routed toward the terminating MMTel AS of the CCxx Called Party.
3. The terminating MMTel AS CAC service determines that the CCxx Called Party is busy at the user level. The MMTel AS CC Monitor checks the user is not provisioned with the CC Monitor opt-out service.

The MMTel AS checks that there is free space in the Monitor queue. In the case where the queue is full, the CCBS Possible Indicator is not sent in the following step and the basic called party busy case applies.

Note: The queue limit can either be defined by a configuration parameter or by an alternative queue size that can be provisioned by the operator per user and CCxx service as part of the subscriber data. Provisioning is done over CAI3G interface by the operator.

4. The terminating MMTel AS sends a busy response including the CCBS Possible Indicator containing the default user identity.
5. The originating MMTel AS sends ACK to the busy response.
6. The originating MMTel AS CC Agent checks that:
 - the busy response contains a CCBS Possible Indicator.
 - the user is provisioned with the CCBS service.
 - the Agent queue limit has not been reached.

If all the above conditions are met, then CCBS is offered to the CCxx Caller.

The values of the Privacy and P-Asserted-Identity headers from any 18x response (not shown) and the busy response are stored.

7. The CC Agent uses an algorithm to determine whether the user should be prompted for digits only, voice and digits or voice only. Another algorithm is used to determine which announcement should be played.

The originating MMTel AS creates a standard early media connection on the MRFP as for other early media announcements.

The CCxx Caller sdpA is added to the H.248 Add request, which is sent to the MRFP.

8. The sdpM is returned by the MRFP in the response.
9. The originating MMTel AS generates a 183 Session Progress provisional response message to the CCxx Caller requesting provisional reliable responses (100rel) and including the MRFP sdpM and a P-Early-Media header set to sendrecv.
10. The CCxx Caller's UE responds with a PRACK message.



11. MMTel AS responds by sending a 200 OK to the PRACK.
12. The originating MMTel AS CC Agent sends an H.248 Modify message to the MRFP to prompt the user and collect user input. The content of the H.248 message is dependent on whether the user is prompted to enter voice and digits, digits only or voice only.

In this case user is prompted only for digits, the originating MMTel AS CC Agent sends an H.248 Modify message to the MRFP to invoke the PlayCollect feature to play the CCBS Invocation announcement and collect the invocation digit sequence inband. This uses the "aasdc/playcol" event descriptor.

The CC Agent uses the announcement determined at step 7 for value of the 'ip' parameter (Initial Prompt) of PlayCollect. Note: The format of the 'ip' is determined based on configuration data. There is also a CM attribute for the used to specify maximum time to play the announcement. The 'dm' parameter (digitmap) is constructed from the value of another CM attribute.

13. The MRFP responds and starts to play the announcement and listen for the Invocation code.
14. User enters the activation code * specified by the 'dm' parameter in the Modify message. It is received as a sequence of in-band DTMF digits. For example *37#
15. The MRFP sends an H.248 Notify message to the originating MMTel AS and reports an "aasdc/pcolsucc" completion event against the CCBS invocation code received inband from the CCxx Caller. The notification contains the invocation code entered by the user in the dc parameter.
16. The originating MMTel AS sends a reply to the MRFP.
17. The CC Agent checks that there is not already a request against the CCxx Called Party in the CCxx Caller queue.

The CC Agent creates a SUBSCRIBE message. The subscription dialog Id (CallId and From tag), To and From header URI values are stored. The request URI and the 'm' parameter values are obtained from the Call-Info header included in the busy response above.

MMTel AS starts the Request Operation timer.

18. The originating MMTel AS CC Agent sends the SUBSCRIBE message to the terminating CC Monitor making a CC Invocation request.



19. The terminating MMTel AS CC Monitor checks that the queue limit has not been reached and accepts the request and places an entry in the monitor queue. The information from the SUBSCRIBE message is stored.

Note: The queue limit can either be defined by configuration data or by an alternative queue size that can be provisioned by the operator per user and CCxx service as part of the subscriber data. Provisioning is done over CAI3G interface by the operator.

20. The terminating MMTel AS CC Monitor sends a 202 Accepted message (with a generated To tag) to the originating MMTel AS.

Note: The To tag is stored at both the originating and terminating ends.

21. The terminating MMTel AS CC Monitor starts the terminating Service Duration timer. The CC Monitor then sends a NOTIFY message with the state field set to queued.

Note: The NOTIFY message and the 202 response sent at step 20 can be received in any order by the originating MMTel AS.

22. The originating MMTel AS receives the NOTIFY message and stores the values of the RequestURI and the To, From, Privacy and P-Asserted-Identity headers from the initial INVITE against the CC agent request. The value of the service-retention field from the NOTIFY message is also stored, which specifies whether Service Retention is applicable to this request.

The originating Request Operation timer is stopped.

The originating Service Duration timer is started.

23. A 200 OK to the NOTIFY is sent.
24. The terminating MMTel AS uses the stored 'm' parameter value to identify this is a CCBS request. MMTel AS checks that the user is busy. The MMTel AS starts to monitor the CCxx Called Party for a reduction in the number of sessions against the user.
25. The originating MMTel AS sends an H.248 Modify message to the MRFP instructing it to play the Invocation Success announcement. The MMTel AS CC Agent determines that an announcement can be played.

The announcement is played using a fixed announcement (apf/an=<announcement id>).

26. The MRFP responds and starts to play the announcement.
27. The MRFP notifies the MMTel AS when the announcement has finished.
28. MMTel AS sends a reply to the H.248 Notify message.



29. MMTel AS subtracts the connection on the MRFP.
30. The MRFP sends a reply to the H.248 Subtract message.
31. The MMTel AS CC Agent removes the CCBS Possible Indicator from the busy response.
32. The MMTel AS sends the busy response to the CCxx Caller.
33. The UE-A sends an ACK to the busy response.



2.7.2.2 User B Available, Recall

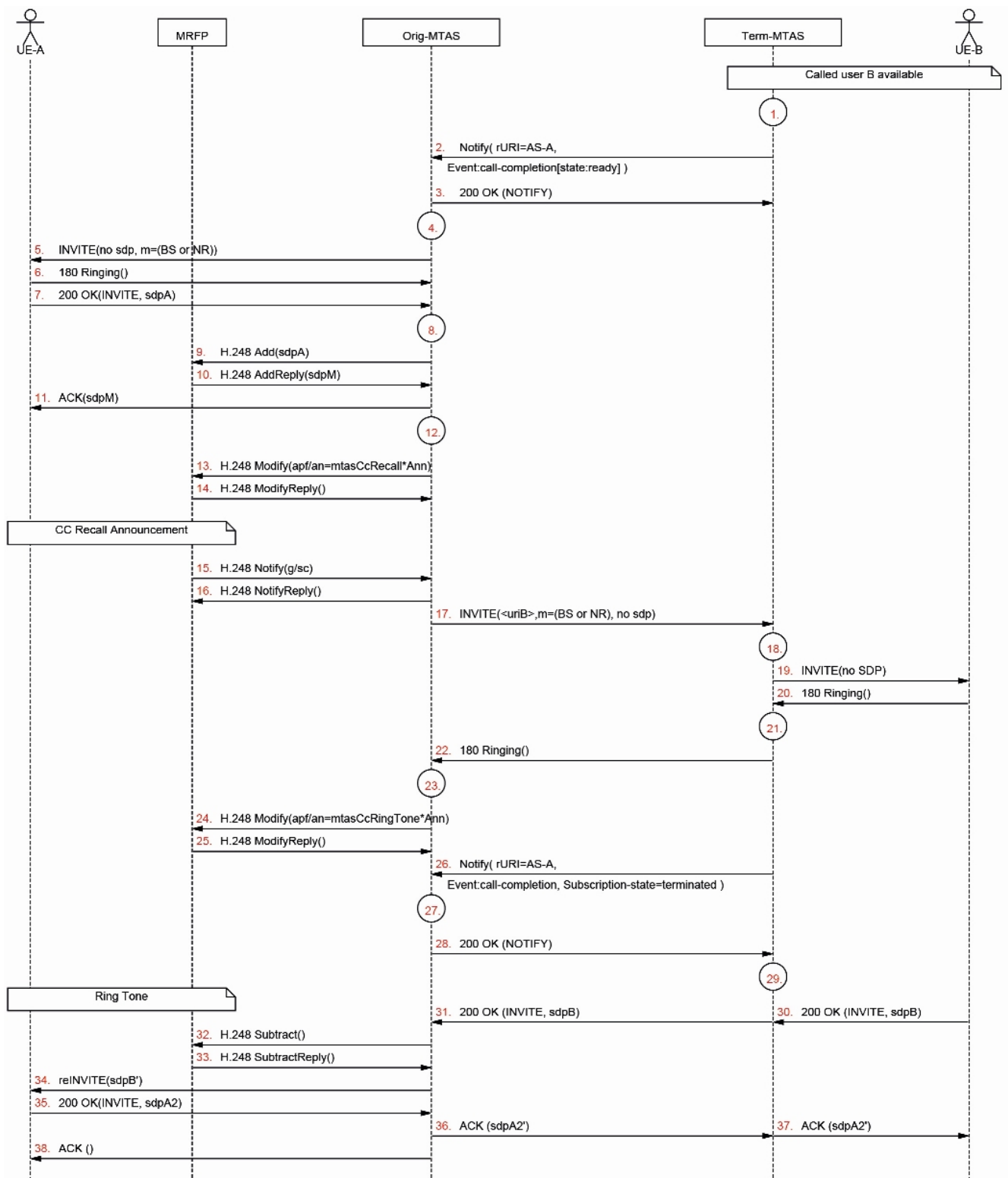


Figure 18 - CCBS Active, User B Available (Part 1)

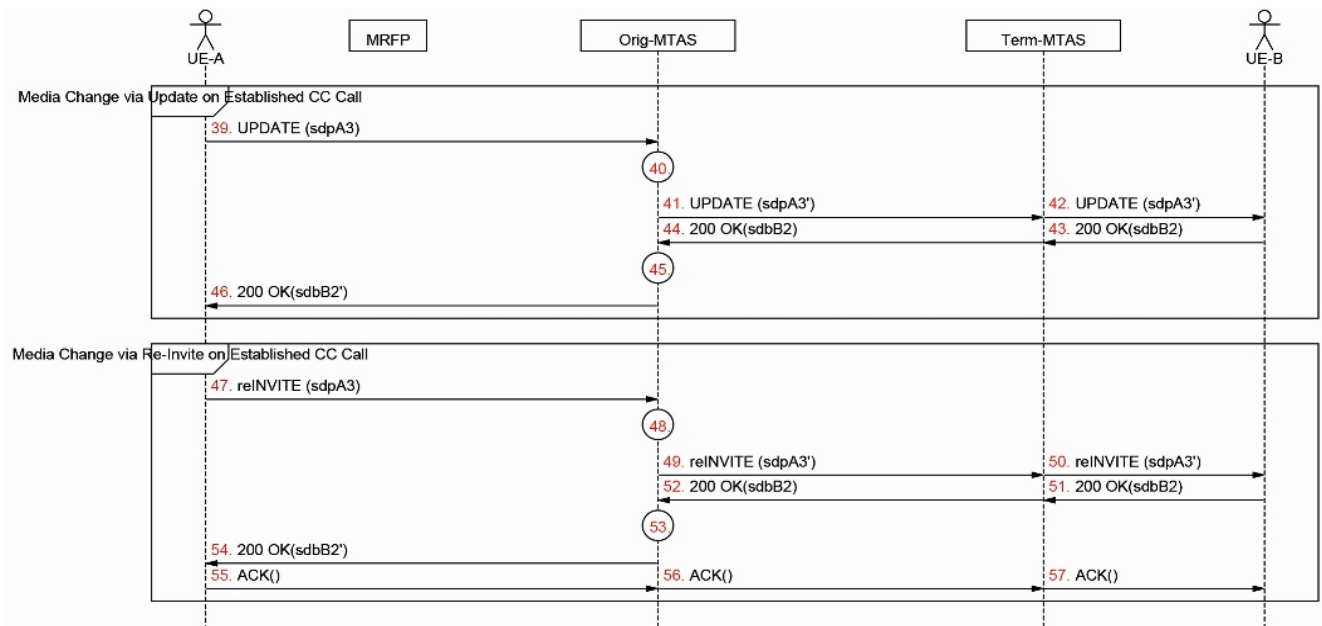


Figure 19 – CCBS Active, User B available (Part 2: Media Update on Established CC Call)

1. When the CCxx Called Party B becomes available to receive a CC call,

MMTel AS creates a NOTIFY message destined for the CC Agent responsible for the request. The NOTIFY is populated with the state field set to ready.

MMTel AS starts the terminating Recall timer of duration.

2. The terminating MMTel AS sends the NOTIFY message to the originating MMTel AS.
3. The CC Agent responds with a 200 OK to the NOTIFY.
4. The originating MMTel AS CC Agent locates the request to which the NOTIFY relates in its queue by matching the dialog identity against an entry in the queue. The MMTel AS CC Agent checks the status of the CCxx Caller A and finds the CCxx Caller is not busy.

The originating MMTel AS CC Agent commences a 3PCC sequence (see [14]) by creating a Recall INVITE destined for the CCxx Caller without including an SDP offer. Note that the 'm' parameter on the Request URI of the INVITE has the value 'BS' for CCBS and 'NR' CCNR.

The MMTel AS CC Agent starts the originating Recall timer of duration.

5. MMTel AS sends the INVITE message to the CCxx Caller user.



6. The user device responds 180 Ringing and alerts the CCxx Caller.
Note: Other initial responses to the INVITE are handled as follows:
 - a 18x, is handled the same as a 180 Ringing.
 - a 200 OK, results in the sequence continuing at step7.
7. The CCxx Caller goes off-hook and a 200 OK is sent including an SDP offer (sdpA).
8. The originating MMTel AS CC Agent stops the originating Recall timer (T4) and stores the SDP against the request (required for later manipulation). MMTel AS determine the media required to play the later announcements.
9. MMTel AS sends an H.248 Add request containing media based on sdpA, to the MRFP.
10. The sdpM is returned by the MRFP in the response.
11. The SDP answer (sdpM) is passed toward the CCxx Caller in the ACK to the 200 OK (INVITE).
12. The CC Agent determines that a CCBS Recall announcement can be played.
13. An H.248 Modify to the MRFP to commence the short recall announcement. This uses a fixed announcement (apf/an=<announcement id>).
14. The MRFP sends a reply to MMTel AS and starts playing the announcement to the CCxx Caller.
15. A general signal completion notification is received from the MRFP.
16. The MMTel AS sends a reply to the MRFP.
17. The originating MMTel AS CC Agent generates an INVITE toward the CCxx Called Party. Note that the 'm' parameter on the Request URI of the INVITE has the value 'BS' for CCBS and 'NR' CCNR.
18. The MMTel AS CC Monitor checks that the userinfo and domain parts of the URI in the To and From headers match the request currently being recalled. If they don't match then the call is rejected

The terminating MMTel AS CC Monitor checks that the user is NOT busy.

The terminating Recall timer (T9) is stopped.
19. The MMTel AS CC Monitor forwards the INVITE toward the CCxx Called Party B.



20. The CCxx Called Party responds with 180 Ringing to MMTel AS.

Note: Other initial responses to the INVITE are handled as follows:
a 200 OK, results in the sequence continuing at step 30.
21. The terminating MMTel AS CC Monitor removes the request currently being recalled from the monitor queue.
22. The terminating MMTel AS CC Monitor forwards the 180 Ringing toward the CCxx Caller.
23. The originating CC Agent starts the MMTel No Reply timer and checks that there is an established media stream to the CCxx Caller and then modifies the existing MRFP connection created at step 9 to apply a Ring Tone announcement to the CCxx Caller.
24. An H.248 Modify message is sent to the MRFP to start playing the Ring Tone announcement.
Note: The announcement uses a fixed announcement (apf/an=<announcement id>).
25. The MRFP sends a reply to MMTel AS and starts playing the announcement to the CCxx Caller.
26. The terminating MMTel AS CC Monitor sends a NOTIFY message toward the CC Agent responsible for the request. The NOTIFY is populated with the Subscription-state header set to terminated.
27. The originating MMTel AS CC Agent stops the originating Service Duration timer (T3) associated with the request and removes the request from its queue.
28. The CC Agent responds with a 200 OK to the NOTIFY.
29. The terminating MMTel AS CC Monitor stops the terminating Service Duration Timer (T7) and removes the CC request from its queue.
30. The CCxx Called Party answers with a 200 OK containing an SDP answer (sdpB).
31. The 200 OK is sent on toward the originating side.
32. The originating MMTel AS CC Agent removes the earlier announcement by sending an H.248 Subtract message to the MRFP.
33. The MRFP sends a reply to MMTel AS.
34. MMTel AS creates a new SDP Offer (sdpB2) using as input: the last SDP sent to the CCxx Caller at step 11 (sdpM); and the latest SDP answer received from the called party (sdpB). MMTel AS generates a re-INVITE message based on the INVITE sent at step above and sends it to the CCxx Caller. The SDP answer (sdpB2) is included along with the Privacy and P-Asserted-Identity headers.



35. The CCxx Caller responds 200 OK to the INVITE with a new SDP answer (sdpA2).
36. The originating MMTel AS CC Agent creates a new SDP offer (sdpA3) by manipulating sdpA2 and using as input: the last received SDP from the called party (sdpB at step 31); and the received sdpA2. MMTel AS sends sdpA3 in the ACK to the 200 OK toward the terminating MMTel AS.
37. The terminating MMTel AS CC Agent sends sdpA3 in the ACK to the 200 OK toward the CCxx Called Party.
38. The MMTel AS CC Agent sends an ACK to the 200 OK (re-INVITE) toward the CCxx Caller.
39. CC Call session is now established. Either party may initiate a change to the media during the session. The following flow shows media change initiated by the CCxx Caller. If the CCxx Called Party initiates a media change, then the reverse flow applies.
40. The media change can be via UPDATE as follows:
41. The CCxx Caller sends UPDATE containing an SDP Offer.
42. The originating MMTel AS creates a new SDP Offer using as input: the received SDP offer; and the latest SDP sent to or received from the CCxx Called Party. MMTel AS updates the media in the UPDATE message.
43. The originating MMTel AS sends the UPDATE message to the terminating MMTel AS.
44. The terminating MMTel AS forwards the UPDATE to the CCxx Called Party.
45. The CCxx Called Party responds with 200 OK to the UPDATE and includes an SDP Answer.
46. The terminating MMTel AS forwards the 200 OK to the originating MMTel AS.
47. The originating MMTel AS creates a new SDP Answer using as input: the received SDP answer; and the latest SDP received from the CCxx Caller. MMTel AS update the media in the 200 OK response message.
48. The originating MMTel AS sends the 200 OK to the CCxx Caller.
49. Alternatively, the media change can be via re-INVITE as follows:
50. The CCxx Caller sends re-INVITE containing an SDP Offer.



51. The originating MMTel AS creates a new SDP Offer using as input: the received SDP offer; and the latest SDP sent to or received from the CCxx Called Party. MMTel AS updates the media in the re-INVITE message.
52. The originating MMTel AS sends the re-INVITE message to the terminating MMTel AS.
53. The terminating MMTel AS forwards the re-INVITE to the CCxx Called Party.
54. The CCxx Called Party responds with 200 OK to the re-INVITE and includes an SDP Answer.
55. The terminating MMTel AS forwards the 200 OK to the originating MMTel AS.
56. The originating MMTel AS creates a new SDP Answer using as input: the received SDP answer; and the latest SDP received from the CCxx Caller. MMTel AS update the media in the 200 OK response message.
57. The originating MMTel AS sends the 200 OK to the CCxx Caller.
58. The CCxx Caller sends ACK to the originating MMTel AS.
59. The originating MMTel AS sends ACK to the terminating MMTel AS.
60. The terminating MMTel AS sends ACK to the CCxx Called Party.

2.7.3 Charging

MMTel AS includes a P-Charging-Vector header and a P-Charging-Function-Address header in the CC Recall and CC Call INVITE messages.

- Offline Charging

Service indication is included.

- Online Charging

MMTel AS does a Credit Control Request. If the call is not allowed to continue, then MMTel AS rejects the call

2.7.4 Service Interaction

2.7.4.1 Call Admission Control

The CCBS service is dependent on the CAC service in order for the network to be able to determine that the user is busy.

2.7.4.2 CDIV

There are a number of interactions between CDIV and CCxx.



Generally CDIV does not divert CC Recalls or CC Calls nor is CCxx offered on calls diverted when the Diverted-To Party responds with a 180 Ringing provisional response.

Existing CCxx requests are suspended (i.e. users are not monitored and no requests are unqueued) when a user activates CFU.

Suspended CCxx requests are resumed when a user deactivates CFU.

CCBS possible indication is sent to user A, if the call is diverted due to CFB and

- The diverted-to network responds with an error response
- The diverted-to network responds with a 180 Ringing response and the CC Provisional Response timer expires

CCNL possible indication is sent to user A, if the call is diverted due to CFNL and

- The diverted-to network responds with an error response
- The diverted-to network responds with a 180 Ringing response and the CC Provisional Response timer expires.

CCxx is not offered if the call diverts on CFNRc and the Diverted-To Party responds with a 180 Ringing provisional and / or a busy or 480 final response.

2.7.4.3 Communication Barring

ICB takes precedence over CCxx.

For the CC Recall INVITE to the CCxx Caller OCB is not applied.

2.7.4.4 Scheduled and Ad-hoc Conference

The Conference creator/initiator does not receive any indication that Communication Completion is possible to the Conference participant.

2.7.4.5 Serial and parallel ringing

Only the first CC possible indicator received will be permitted to be acted upon the others are filtered.

2.7.4.6 Identity Presentation Services

- OIP

OIP is applied at the originating MMTel AS on the CC Recall INVITE to the CCxx Caller.

OIP is applied at the terminating MMTel AS on the CC Call INVITE to the CCxx Called Party.



- OIR

OIR is applied at the originating MMTel AS on the CC Call INVITE toward the CCxx Called Party, using the same information as received in the original received INVITE.

- TIP

TIP is not applicable at the originating MMTel AS for neither the CC Recall to the CCxx Caller, nor the CC Call to the CCxx Called Party, since a 3PCC sequence is being followed and responses are not passed on.

- TIR

TIR is not applicable at the originating MMTel AS for responses to the CC Recall INVITE to the CCxx Caller, since a 3PCC sequence is being followed and responses are not passed on.

TIR shall be applied at the terminating MMTel AS on responses to the CC Call INVITE that was sent to the CCxx Called Party.

2.7.4.7 CNIP

CNIP is applied at the terminating MMTel AS on the CC Call INVITE to the CCxx Called Party.

CNIP is not applied by the originating MMTel AS on the CC Recall INVITE to the CCxx Caller.

2.7.4.8 Carrier Select

A Carrier selected by the CCxx Caller on an original call, shall also be used for the subsequent CC Call. This applies to the generic service and Carrier Select Rn.

2.7.4.9 Carrier PreSelect

A PreSelected Carrier for an original call shall also be used for the subsequent CC Call. This applies to the generic service and Carrier Preselect Rn.

2.7.4.10 Communication Waiting

CCNR is not offered from B if Communication Waiting is invoked at the CCxx Called Party.

If a Called user rejects a Communication Waiting request, then CCBS is offered, subject to the normal conditions being met.

Communication Waiting is not offered as a result of a CC Call.



2.7.4.11 MCID

- Permanent Mode

Permanent mode MCID shall apply to an incoming CC Call from a CCxx Caller, but not to CC Recall INVITE messages resulting from a maturing CC request.

- Temporary Mode

Temporary mode MCID applies if the last incoming call was a CC Call from a CCxx Caller, but does not apply if the last incoming call was a CC Recall resulting from a maturing CC request. A CC Recall is never reported. The last non CC Recall call is always included in the report.

2.7.4.12 Priority Call

The CC Call INVITE messages shall have the same priority as the original INVITE on which CCxx was invoked.

2.7.4.13 Unregistered User

Invocation of Unregistered User service is suppressed in the terminating MMTel AS if a possible CCNL indication is included in a SIP 480 response.

2.7.4.14 Network Announcement

In the terminating network the Network Announcement service invocation is suppressed if a 4xx-6xx response is received after a 180 Ringing provisional response including a CCNR possible indication has been received in the same session.

In the terminating network, the Network Announcement service invocation is suppressed if a SIP 486/600/603 response is received that includes a CCBS possible indicator.

In the originating network, the Network Announcement service is suppressed if a successful CCBS, CCNR or CCNL activation has been performed.

2.7.5 Provisioning

It is possible to provision/withdraw

- CCBS, CCNL and CCNR service to a user. The service operates on the A-side that is for a calling user.
- CCBS, CCNR, CCNL opt Out services. The service operates on the B-side, that is for a called user.
- Max nbr of CCNL, CCBS and CCNR requests individually per service that are possible on the B-side for a user. If the Maxsizes are not provisioned a configured max number of CCxx requests decides the max B queue size.



2.7.6 Configuration

- Administrative State
- Timers for invocation, recall and queue handling
- Maxsizes of the A queue and the B-queue for invocation from and towards a user. The configured maxsize for the B-queue can be overridden by provision of maxsizes for CCBS, CCNL and CCNR requests on user level as described in previous chapter
- Announcements, IVR, invocation code and ring back tone configuration

2.7.7 Performance Management

Counters for on originating and terminating - successful and unsuccessful – internal and external

- Indications
- Invocations
- Recalls
- Completions
- Queue handling

2.8 Ad-hoc Conferencing

2.8.1 Description

MMTel AS supports ad-hoc (dial-out) conferencing according to TISpan/3GPP Conferencing Service [14].

The ad-hoc conference is an unscheduled conference that is created on-the-fly. The service allows the subscribers to start a conference and invite other users (conference participants, CPs) to the conference. The user creating or starting the conference is called the conference creator (CC).

Both audio and video conferencing (audio + video) are supported. Both conference types are supported in combination with conference content (video slides) sharing. The RTP/AVPF transport profile may be used for each stream where the RTP/AVP transport is supported (see [32] for more details about the RTP/AVPF).

The entity of the MTAS providing the conference function is called the conference server. The term is used to group the factory, focus and policy functions (see [3]) and it is not a physical server.

The conference service includes operations such as: conference creation, conference event notification, invitation of users (participants) to join the conference and conference termination.

An ad-hoc conference session can be created using a conference factory URI in the request line as described in 5.3.2.3.1 of 3GPP TS 24.147 ref [6].

There are many ways in which a user can join (be invited) to the conference. The following methods are supported:

- Invite another user to a conference using REFER request as described in 5.3.2.5.2 of 3GPP TS 24.147 ref [6].
- Invites other users to a conference by including URI list in the initial INVITE request to the conference focus as described in 5.3.2.5.3 of 3GPP TS 24.147 ref [6].

If the answer confirmation function is enabled, then the conference function connects participants to the conference only if they confirm the invitation.

Figure 19 shows the conference server with its external interfaces in an IMS network. (Dotted arrows on the ISC interface indicate that there are intermediate IMS network nodes that are not shown in the figure.

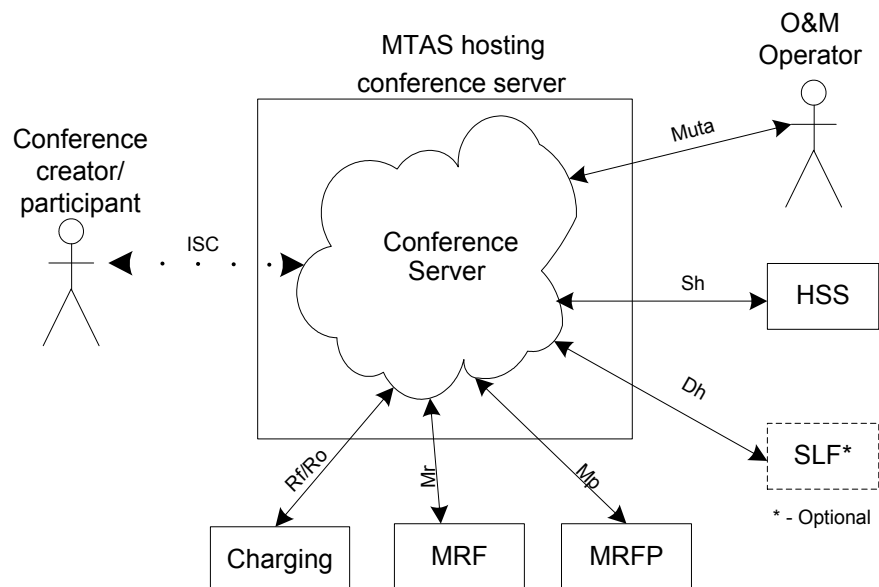


Figure 20. Conference server and its external interfaces

The conference server can be deployed either as a stand-alone endpoint separate from the originating AS or collocated with the originating AS of the conference creator.

The stand-alone deployment is deprecated as only the collocated case works together with PSTN (and other networks), otherwise it cannot be guaranteed that a new INVITE with replaces header (use case: move of active session to a conference) finds the same MGC (or similar NNI node).

When collocated with the originating AS, the conference server is linked in to the service chain together with other telephony services of the subscriber.



The main use-cases of the conference function are:

- Conference creation
- Invitation of users to a conference, Dial – Out
- Invitation of users to a conference with the answer confirmation, Dial – Out
- Conference creation and invitation of users with the help of URI list
- Move of active session to a conference
- Removal of a conference participant from the conference
- Conference participant leaving a conference
- Conference termination
- Conference modification
- Conference notifications
 - Entering together with progress info
 - Leaving
 - Hold/resume events

All use cases are supported using either a Media Resource Processor Function (MRFP) or a Media Resource Function (MRF). However conference content sharing is only supported using an MRFP.

The MRFP is controlled by the conference server's internal Media Resource Function Control (MRFC) through the Mp-interface [10] (H.248). The MRF is used through the Mr-interface [44].

The conference creator can also invite PSTN/PLMN users to the conference (audio only).

MTAS supports content sharing in ad-hoc conferences using a dedicated content sharing video stream. MTAS also supports floor control to coordinate the access to these resources. Floor control is performed between the UE and the MRFP using BFCP (Binary Floor Control Protocol). MTAS is involved in content sharing and the establishment of the floor control channel by monitoring specific policy rules and set floor specific identities to use between the client and MRFP.

When the answer confirmation function is enabled MRFP/MRF is also responsible for:

- playing the conference entry announcement to invited participants to notify them that they are about to join the conference,
- collecting the DTMF confirmation.



The conference policies maintained by the conference server are static. They do not change between conferences, nor do they change during a session. These policies are as follows:

- Only CC may add another CP to the conference. The media types used by the CC will be offered with codecs according to the capabilities of the MRFC.
- Only CC can remove (kick-out) another CP from the conference.
- Any CP (other than the CC) may leave the conference at any time without this affecting the rest of the conference.
- The conference is terminated when the CC leaves the conference,
- Anyone can change the port.
- Any participant can add new media streams to the conference e.g. upgrading an audio conference to video. The CP may use a different transport profile (e.g.: RTP/AVP resp. RTP/AVPF) than the CC.
- However a CP may only add media types used by the CC.
- No one can remove media streams from the conference.
- All participants may subscribe to conference notifications. This is only supported inside a dialog.
- Only one audio, one application and a maximum of 9 video streams are allowed to be used per participant. Other media types are not allowed.
- URI list is only supported for the CC when used in the initial INVITE request towards the factory

The Conference Creator (CC) and the Conference Participants (CP) can subscribe/unsubscribe to the conference notification events, in the created ad-hoc conference, by an in dialog SUBSCRIBE/NOTIFY procedure. The conference server applies congestion control on the notification event messages.

During call setup media can be modified by the CP by sending early UPDATE.

2.8.2 Example Call Flows

2.8.2.1 Conference creation followed by a request to invite one participant

The following figure shows a simplified signalling flow for the creation of an ad-hoc conference call followed by a request to invite a specific conference participant.

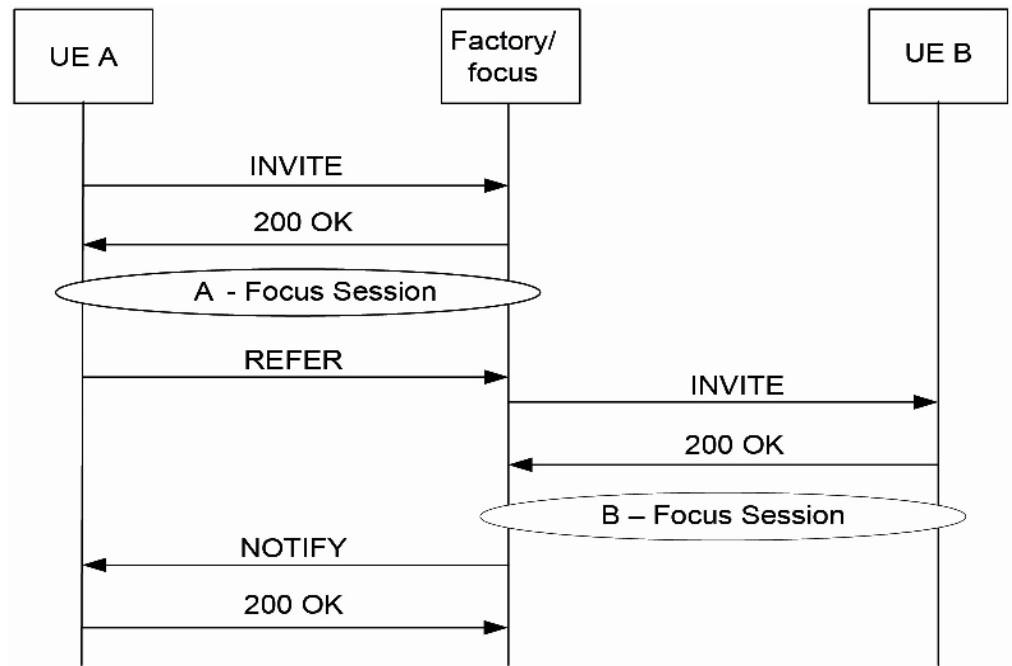


Figure 21 - Simplified call flow of ad-hoc conference creation with dial-out method

In Figure 20, A is the Conference Creator (CC) and B is a new Conference Participant (CP).

New participants can be joined to a conference using a Dial - Out method, where the REFER is sent from the CC to the focus which in turn sends the INVITE to the CP.

2.8.2.2 Conference creation with URI-list

Following figure shows a simplified signaling flow for the creation of an ad-hoc conference including URI-list in the initial INVITE. See reference [3] for a more detailed call flow.

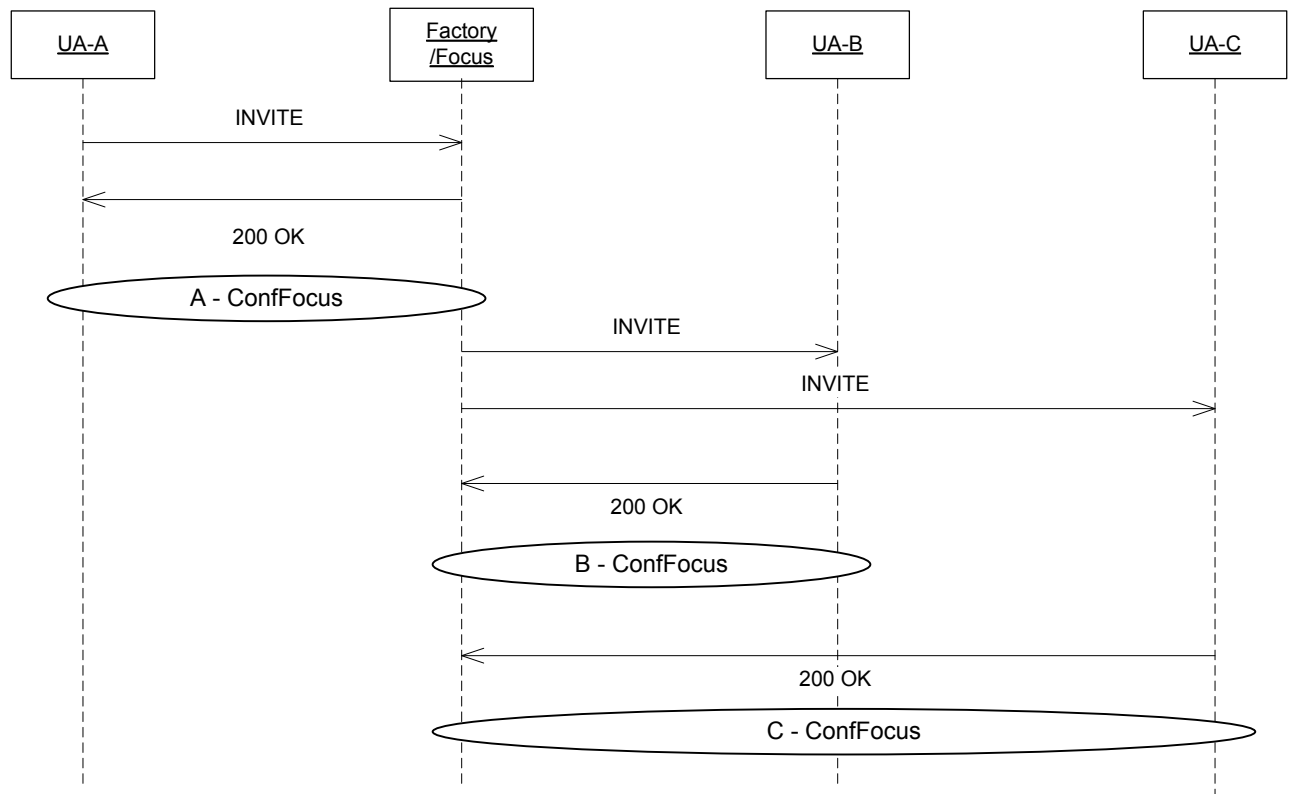


Figure 22 - Simplified call flow of ad-hoc conference creation with URI-list

In Figure 21, A is the Conference Creator (CC) while B and C are a new Conference Participants (CP).

New participants can be joined to a conference including a URI-list in the initial INVITE towards the conference factory. The focus will (once created) invite each participant included in the URI list to the conference.

The INVITE with a URI list does not create implicit subscription; the conference creator has to subscribe explicitly to conference events, if it is wanted.

The URI entries in URI list can be specified either in SIP or Tel URI format.

2.8.3 Charging

The user creating the conference, the conference creator (CC) is charged for each leg. The charging for the conference creation is performed on the originating Application Server (AS), while the charging for the invitation of conference participants is handled by the conference server.

For active sessions that are moved to the conference, the charging session created for the original session may, depending on configuration either be preserved after the move into the conference or a new charging session is created representing the new conference participant and the original 2-party charging session of the participant is terminated.

The charging data includes:



- IMS Charging Identifier – ICID (i.e. creator leg ICID) which is generated by the conference server.
- Supplementary Service Information – indicating creation/termination of a conference at the originating AS and indicating addition/removal of conference participants at the conference server.
- Conference ID – generated at the originating AS and at the conference server.

2.8.4 Service Interaction

CONF interacts with following services.

2.8.4.1 Call hold/resume

The focus acts as a B party in a hold/resume scenario. When Hold is requested towards a conference no announcement is played towards the conference.

2.8.4.2 OCB

The focus does not perform any validation of OCB rules. The validation of OCB rules for the CC is performed on the originating side.

2.8.4.3 OIR/TIR

In Dial - Out scenarios, the focus respects the privacy settings (OIR/TIR) for both CC and CPs, done on the originating/terminating AS respectively. The incoming signals privacy headers are copied to the outgoing signals.

2.8.4.4 Communication Waiting

When CW is requested towards a CP no announcement is played towards the conference and the CC will not get any indication that CW is used to towards the CP.

2.8.4.5 Resource Reservation

The precondition attributes are not supported and not included in the SDP offer sent in the H.248 to the MRFP. Nor are they propagated to the other legs at their creation. Resources will always be reserved for the focus UAs.

2.8.4.6 OPTIONS

Conference focus URI is created dynamically for a conference and it is valid only during the conference session. The conference URI is normally known only to the conference participants (CC and CPs) and it is reasonable to expect the OPTIONS request from them only. The requests initiated inside a dialog are supported only. OPTIONS request received outside a dialog, including a request from a user not participating to the conference, will be rejected with 403 Forbidden error message.



2.8.4.7 OCNIP

If OCNIP is disabled (i.e. locked, not provisioned or no valid license exist), then display-names are removed (if present) from the CCs initial INVITE URI list. Removed display-names will not appear in the display-text tags of the Notifications XML bodies.

Note: If `mtasIdPresDefaultDisplayName` has been configured, it will be set as display name when OCNIP is disabled. If `mtasIdPresDefaultDisplayName` is not configured then the display name is removed.

2.8.5 Configuration

The O&M operator can configure the conference server with regard to:

- Administrative state (locked/unlocked/shutting down)
- Conference factory URI
- Conference URI (only userinfo prefix and subdomain name can be configured)
- The ISC port number of the S-CSCF where an INVITE from the focus is to be routed to.
- Activate/deactivate the conference notification service for all CPs or only for CC
- Activate/deactivate the additional progress information notifications when Focus is dialing-out to CP or when CP is responding for the dial-out with 180 Ringing or any of the 4xx-6xx messages
- Activate/deactivate usage of the display-names from CCs initial INVITE URI-list in a conference events notifications as display-text
- Define if the unlawful conference subscription is rejected with “403 Forbidden” or “489 Bad Event” failure response
- Activate/deactivate collocated deployment with other user services on the originating MMTel AS.
- Activating/deactivating use of URI list
- Announcement ID - played to invited participants to notify them that they are about to join a conference if the conference is provisioned with answer confirmation
- DTMF digit map - controls which answers from participants allow them to join the conference. The parameter is used if the conference is provisioned with answer confirmation
- Activate/deactivate offering of all supported codecs to CP in case of external MRFC.



2.8.6 Performance Management

The O&M operator can monitor the conference server in terms of performance:

- The number of active conference calls
- The number of successfully created conferences
- The number of failed attempts to create conference
- The number of successfully added conference participants
- The number of failed attempts to add conference participant
- The number of failed attempts to add conference participant due to conference license problem
- The number of successful dial-out answer confirmation functions executions
- The number of unsuccessful dial-out answer confirmation functions executions due to an internal error
- The number of unsuccessful dial-out answer confirmation functions executions due to an external error from MRF

2.9 Three Party Call

2.9.1 Description

The MTAS offers the basic 3-party session (3PTY) service to its subscribers. The service allows a user who is involved in two separate 2-party sessions with another two participants to convert to a 3PTY session by reusing the existing dialogs from the 2-party sessions. The user starting the 3PTY session is called “3PTY originator” and is able to toggle participants in and out of the 3PTY session.

The main use-cases of the 3PTY service are:

- Creation of 3PTY session (from the existing two 2-party sessions)
- 3PTY session hold/resume
- Participant leaves 3PTY session
- Termination of 3PTY session
- 3PTY originator resumes to 2-party session

2.9.2 Example Call Flow

The Figure 22 shows a simplified signaling flow for the creation and termination of a 3PTY call.

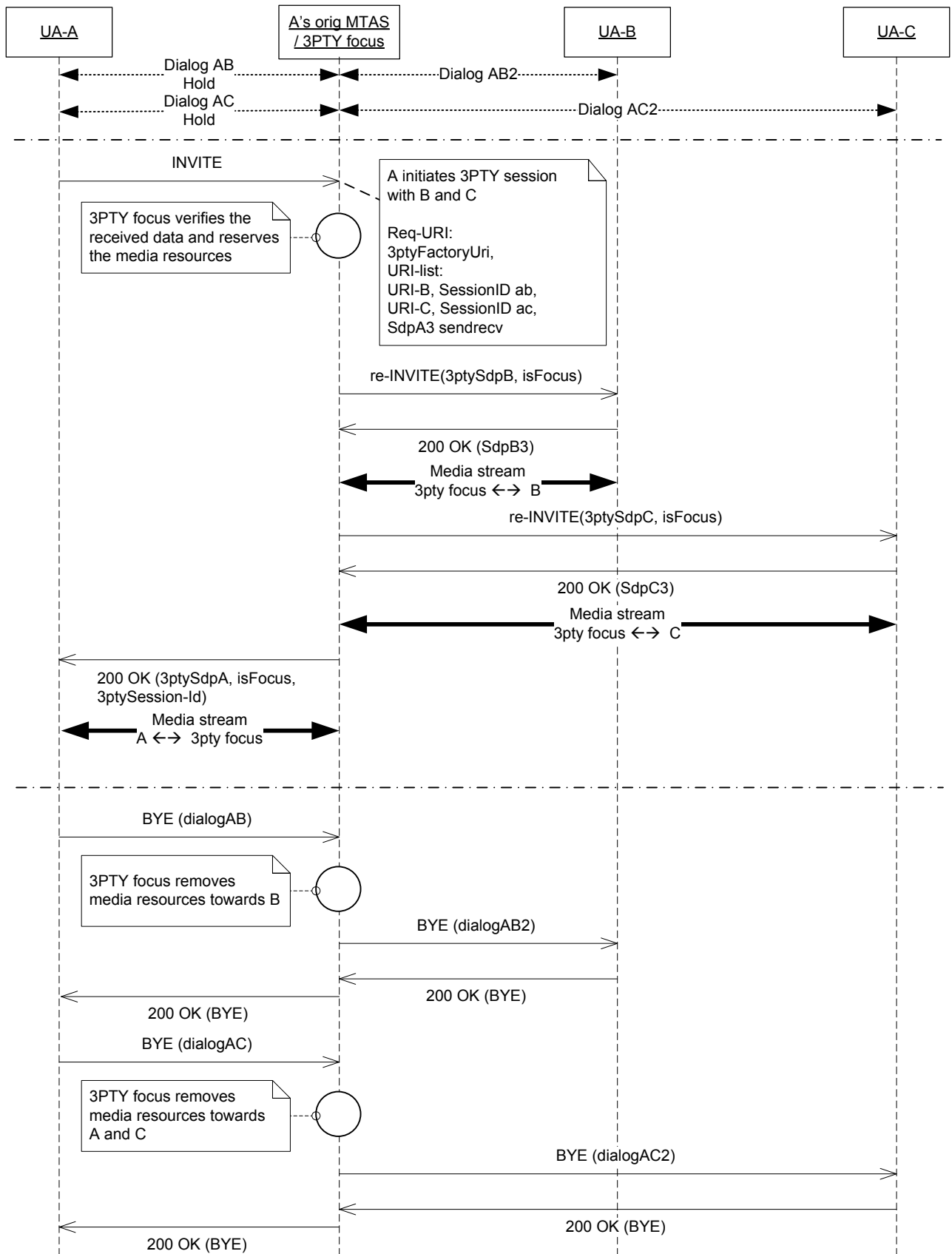


Figure 23: Simplified 3PTY call-flow



The 3PTY call begins with two 2-party sessions in Hold. The 3PTY session is initiated by an INVITE from user A containing the 3PTY factory URI, references to the AB and AC sessions and an SDP offer.

The 3PTY focus verifies the received data, reserves the media resources for the A, B and C parties in the MRS/MRFP (the communication towards the MRS/MRFP is not shown in the figure), updates the B and C parties with the new media streams, and accepts the 3PTY call towards A. All the parties are connected after that.

Normally, the 3PTY call terminates when party A sends BYE towards B and C but it can revert to one or two 2-party sessions as well.

2.9.3 Charging

The communication request initiating the 3-party session is treated as an originating session for charging purposes. Charging for the original 2-party sessions (A-B and A-C) continues throughout the period that the sessions are included in the 3-party session.

The following service specific AVP is applicable to charging for the 3-party session:

- a) Supplementary Service Information - indicating 3-party session.

The following service specific AVPs are applicable to charging for the 2-party sessions:

- a) Supplementary Service Information - indicating 2-party session transition or involvement.
- b) Related ICID - used for correlation purposes.

2.9.4 Service Interactions

2.9.4.1 Hold

The 3PTY service uses the HOLD service when playing announcements on dialogs that are put on hold.

2.9.4.2 CAC

The 3PTY originator have two dialogs when the request for a 3PTY call is received by MTAS. The request for a 3PTY call is counted as one call, thus the user must have a CAC limit of at least 3 to be able to make a 3PTY call.

2.9.4.3 AoC

No AoC information is sent to the user on the SIP dialog for requesting a 3PTY call. AoC information is sent on the SIP dialogs for the two party calls.



2.9.4.4 Explicit Communication Transfer

It is not possible to invoke Explicit Communication Transfer if one or both of the sessions to be transferred are participating in a 3PTY conference. After successful Explicit Communication Transfer the transferred session cannot be involved in a 3PTY conference.

2.9.4.5 Flexible Service Format Selection

The 3PTY service can be suppressed by means of Flexible Service Format Selection (FSFS) service. When at least one of 2-party dialogs is indicated by the FSFS service to be suppressed from joining any 3PTY session, the user will not be able to initiate the 3PTY service, and the 3PTY service invocation attempt will be responded with 403 Forbidden containing the warning header "The service is suppressed".

2.9.5 Configuration

The O&M operator can configure the 3PTY service with regard to:

- Administrative State (locked/unlocked/shutting down)
- 3-party factory URI (subdomain-based PSI), consisting of a user name and a subdomain. Example: 3pty@factory.operator.net
- Operator Subscription Level Service Configuration
In 3PTY configuration data for a subscriber the operator shall indicate whether the subscriber is allowed to create 3PTY session.

2.9.6 Performance Management

The O&M operator can monitor the 3PTY service in terms of performance:

- The accumulated duration time of the 3PTY calls grouped by the S-CSCF IP addresses
- The number of successfully created 3PTY calls
- The number of failed attempts to create 3PTY calls due to node external/internal errors
- The number of abnormally terminated 3PTY calls due to node external/internal errors



2.10 Explicit Communication Transfer

2.10.1 Description

The Explicit Communication Transfer enables a party who has an ongoing communication with a second party to transfer the communication to a third party. The user that initiates the communication transfer leaves the communication so that the second and third parties have an established session. MTAS then enables the transferor to check the status of the transferred session and terminate the ongoing transferred session using SSC codes.

There are two types of Explicit Communication Transfer: blind and consultative.

In case of blind Explicit Communication Transfer the transfer initiator does not have any communication with the third party prior to the transfer.

In case of consultative Explicit Communication Transfer the transfer initiator has a consultative communication session with the third party.

MTAS provides support for the consultative Explicit Communication Transfer.

2.10.2 Example call flow

The Explicit Communication Transfer service defines three different user roles:

- transferor
- transferee
- transfer target.

Transferor is the user that initiates the Explicit Communication Transfer. Initially, he/she has two existing sessions: one with the transferee and one with the transfer target. After the communication transfer is performed these sessions are connected and the transferor is removed from the communication.

Transferee has a communication with the transferor before the ECT is initiated. During ECT the remote communication party will be changed from the transferor to the transfer target.

Transfer target has a communication with the transferor before the ECT is initiated. During ECT its remote communication party will be changed from transferor to the transferee.

The MTAS serving the transferor is called Transferor AS.

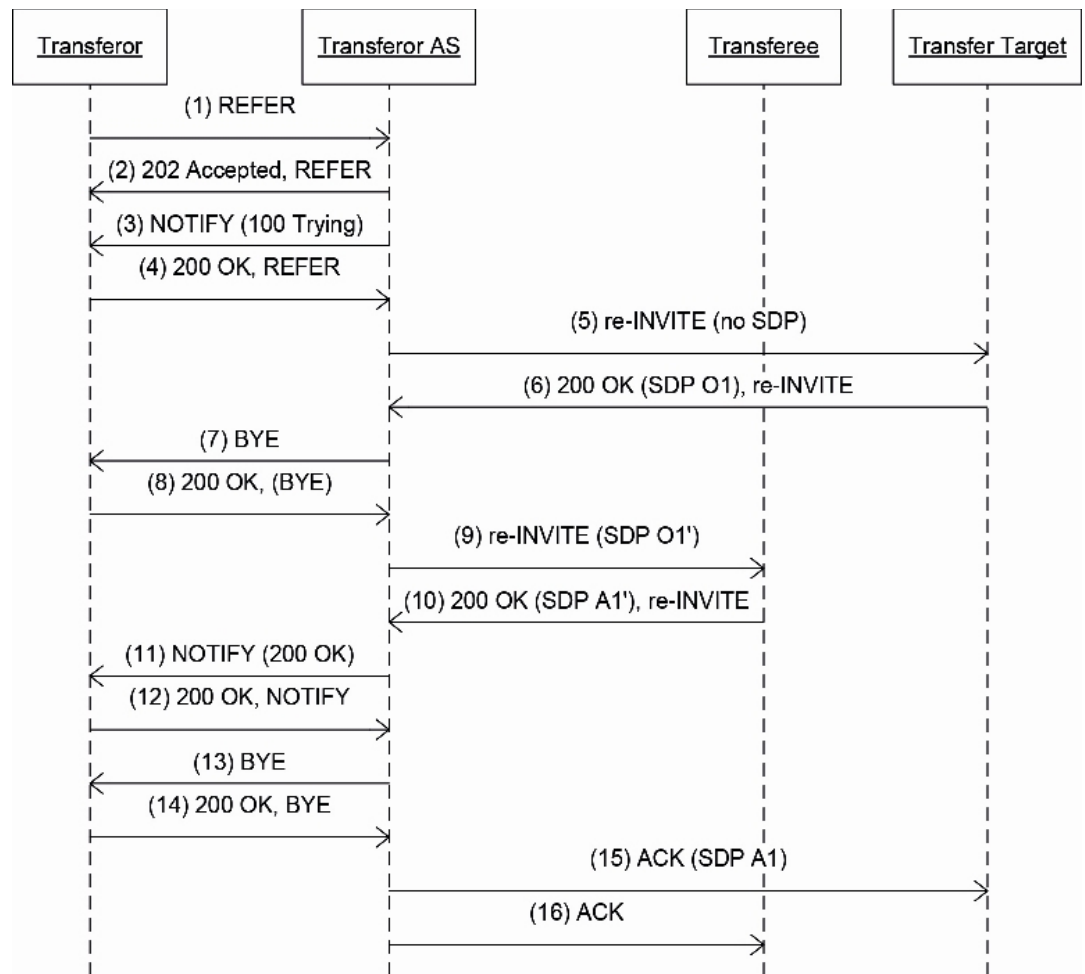


Figure 24 – Consultative Explicit Communication Transfer

- 1 A REFER request is received by the transferor AS.
- 2 The transferor AS accepts the REFER request.
- 3 The transferor AS notifies the transferor that its sessions are being transferred.
- 4 The 200 OK response to the NOTIFY is received.
- 5 The transferor AS sends an empty re-INVITE towards the transfer target. This request is to enforce an SDP offer from the transfer target.
- 6 The transferor AS receives a 200 OK response to the re-INVITE request. The response conveys an SDP offer.
- 7 The transferor AS sends a BYE request towards the transferor in order to terminate its session with the transfer target.
- 8 The BYE request is accepted.
- 9 The transferor AS sends a re-INVITE request towards the transferee.
- 10 The transferor AS receives a 200 OK response to the re-INVITE request.



- 11 The transferor AS sends a NOTIFY request in order to notify the transferor of the successful transfer.
- 12 The transferor AS receives a 200 OK response to the NOTIFY request.
- 13 The transferor AS sends a BYE request towards the transferor in order to terminate its session with the transferee.
- 14 The BYE request is accepted.
- 15 The transferor AS sends an ACK request on the transfer target remote leg. This request is to acknowledge the 200 OK response previously sent by the transfer target. The ACK request contains an SDP answer.
- 16 The transferor AS sends an ACK request on the transferee remote leg. This request is to acknowledge the 200 OK response previously sent by the transferee.

2.10.3 Service Interaction

2.10.3.1 Identity Presentation

The Identity Presentation service is based on the identities stored in the P-Asserted-Identity header. As this header is not present during and after the establishment of the transferred session the identity of the transferee is not presented to the transfer target. Similarly, the identity of the transfer target is not presented to the transferee.

2.10.3.2 Communication Barring

The sessions of the transferor are regular MMTel sessions before ECT is initiated. The Communication Barring rules are applied during the establishment of these sessions.

During and after the communication transfer the Communication Barring service will not be triggered.

The transferred session is established even if the transferee and the transfer target have restrictive rules against each other.

2.10.3.3 Malicious Communication Identification

When registering a communication as malicious after establishing a transferred session, the identity of the malicious party refers to the transferor even if the actual communication partner is the transferee or the transfer target.

2.10.3.4 Call Admission Control

After successful communication transfer in the transferor AS the Explicit Communication Transfer sessions shall not be counted against the transferor.



2.10.3.5 Three Party Conference

The REFER request for Explicit Communication Transfer is rejected if the transferee session or the transfer target session are involved in a 3PTY conference.

After the transfer the transferor AS supervises the transferred session and the requests for 3PTY conference creation are rejected by the transferor AS.

2.10.3.6 Ad-hoc Conferencing

The REFER request for Explicit Communication Transfer is rejected if the transferee session or the transfer target session are involved in an Ad-hoc conference.

2.10.3.7 Explicit Communication Transfer

It is possible to transfer a session that has been previously transferred. The number of transfers in a transferred session is not limited by the SIP signaling.

2.10.3.8 Supplementary Service Codes

The transferor can check the status of his/her ongoing transferred sessions by means of SSC interrogation of the ECT service.

The transferor can terminate his/her ongoing transferred sessions by means of SSC invocation of the ECT service.

2.10.4 Configuration

- On/off on node level
- Provisioned to the user as operator domain subscription data
- Announcement that is played to the call parties when the transferor terminates his/her ongoing transferred sessions.

2.10.5 Performance Management

The following performance counters are provided by MTAS to evaluate the usage and quality of service:

- The number of successfully initiated Explicit Communication Transfers.
- The number of unsuccessful Explicit Communication Transfers attempts due to internal error.
- The number of unsuccessful Explicit Communication Transfers attempts due to external error



2.10.6 Fault Management

For information on the alarm, refer alarm OPI [57].

2.11 Session Transfer to Own Device

2.11.1 Description

The Session Transfer to Own Device (STOD) is a service that enables the calling or called user in a session to transfer the communication session to another device belonging to the same user (terminals defined in the subscription data) or to a Related User.

The STOD service consists of Call Push and Call Pull sub-functions.

STOD (Call Push) implies that the communication leg towards the served user is terminated and the serving MTAS initiates serial or parallel (or both in a combined way) session establishment towards the devices and/or related users. The communication session will be transferred to the target first answering the call.

Originating Session Transfer to Own Device (Originating STOD) service lets the caller in an already established session to transfer the communication to another device or on other user (a.k.a. target).

After a successful Originating STOD invocation the session will be established between the called party and the target.

Terminating Session Transfer to Own Device (Terminating STOD) service lets the called party in an already established session to transfer the communication to another device or on other user (a.k.a. target).

After a successful Terminating STOD invocation the session will be established between the caller and the target.

STOD Call Pull is a feature which allows transfer of established communication session between devices belonging to the same subscriber on subscriber request. Subscriber is able to transfer established call to another device by sending a SIP request with dialog replacement information or a special service code from the Pull initiating device. If all preconditions for Call Pull are fulfilled, the MMTel AS seamlessly transfers the communication session.

2.11.2 Example call flow

The STOD service defines the following user roles:

- Caller
- Called PartyTarget
- Served User's Pulling Device



Caller is the user who initiated the communication towards the other party.

Called party is the user who accepted the call from the caller.

Target is the user or device, who will accept the transferred call after the STOD invocation.

Served User's Pulling device is another device of the Caller (Originating STOD) or the Called party (Terminating STOD), that is initiating transfer of the call from Caller/Called-party to itself using STOD Call Pull Feature Code.

2.11.2.1 Establishing call session

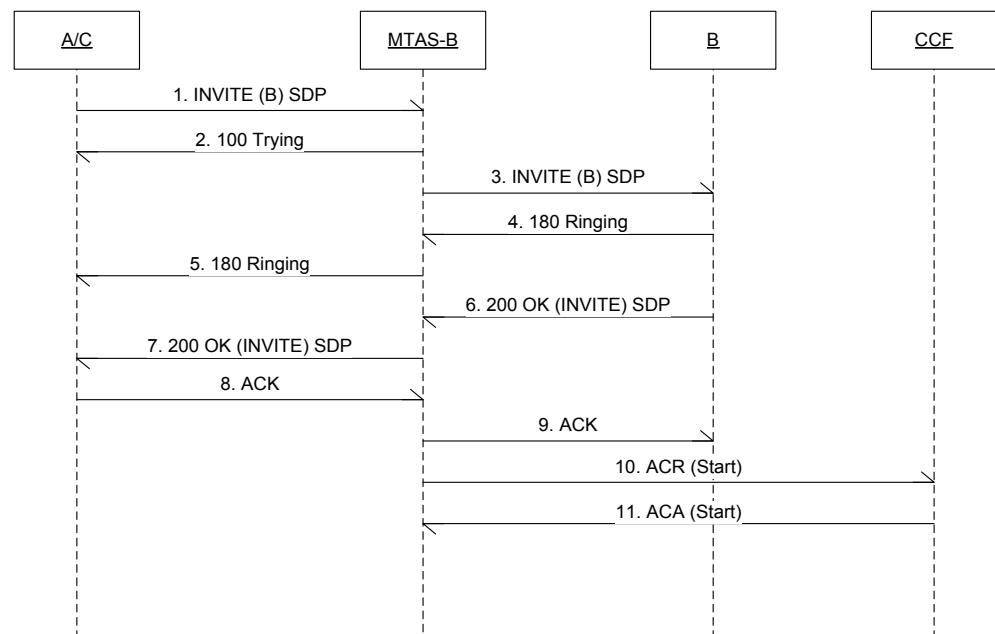


Figure 25 B receives a call from A, which B answers.

A sends an INVITE signal to B in order to establish a call session. B accepts the call.



2.11.2.2 Originating STOD

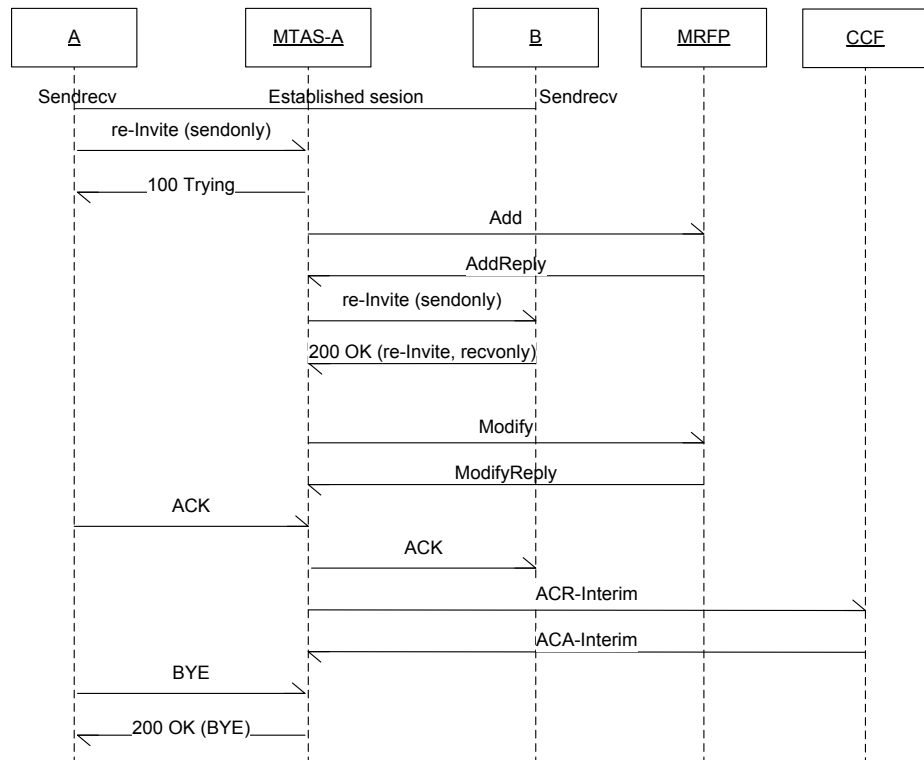


Figure 26 A puts the call on hold, and then sends BYE within the STOD timer to invoke STOD.

A sends a re-Invite to B to put him or her on-hold – media direction is set to sendonly. B accepts the request by sending 200 OK to the re-Invite by setting the media direction to recvonly. Hold service starts to play an announcement to B if configured.

A sends a BYE within a predefined period after the successful hold to invoke STOD. STOD sends the 200 OK to the BYE to A.

STOD transfers the session:

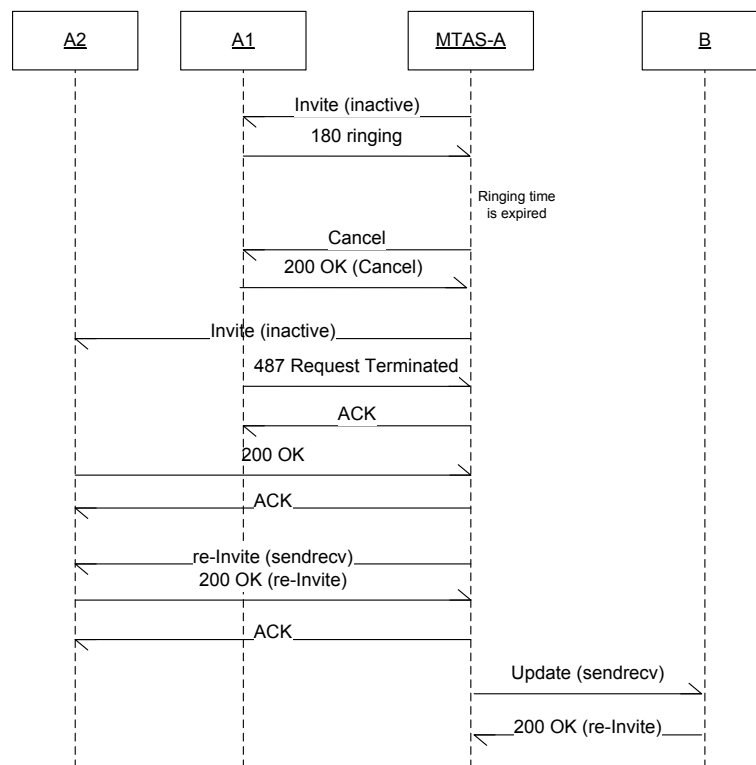


Figure 27 Serial STOD service rings A1 first, ring timer expires, and then A2 is rung, which answers the call.

STOD starts to work, by alerting the targets. In the example call flow above, it uses serial ringing, so it sends the Invites one by one, and waiting for the final response. STOD send the original Invites with SDP, setting the media direction to inactive.

First it tries to call A1 but he or she does no answer within the ringing period, so MTAS decides to Cancel the call.

After the 200 OK for the Cancel, STOD starts to alert to second target, A2 in this case.

In the meanwhile A1 responds to the invite with 487 Request Terminated, and MTAS sends an ACK to it.

A2 accepts the INVITE, and MTAS sends an ACK to it.

After the call is established, MTAS sends a re-INVITE to A2 to set the media to sendrecv.

A2 accepts the re-INVITE, and MTAS sends an ACK to it.

When the re-Invite is ready, and the session is established again with A2, STOD sends an Update to B with the new values within it.

It will explicitly finish the on-Hold state of B, since the update will contain an SDP with sendrecv media direction.



The call session is established between A2 and B.

2.11.2.3 Terminating STOD

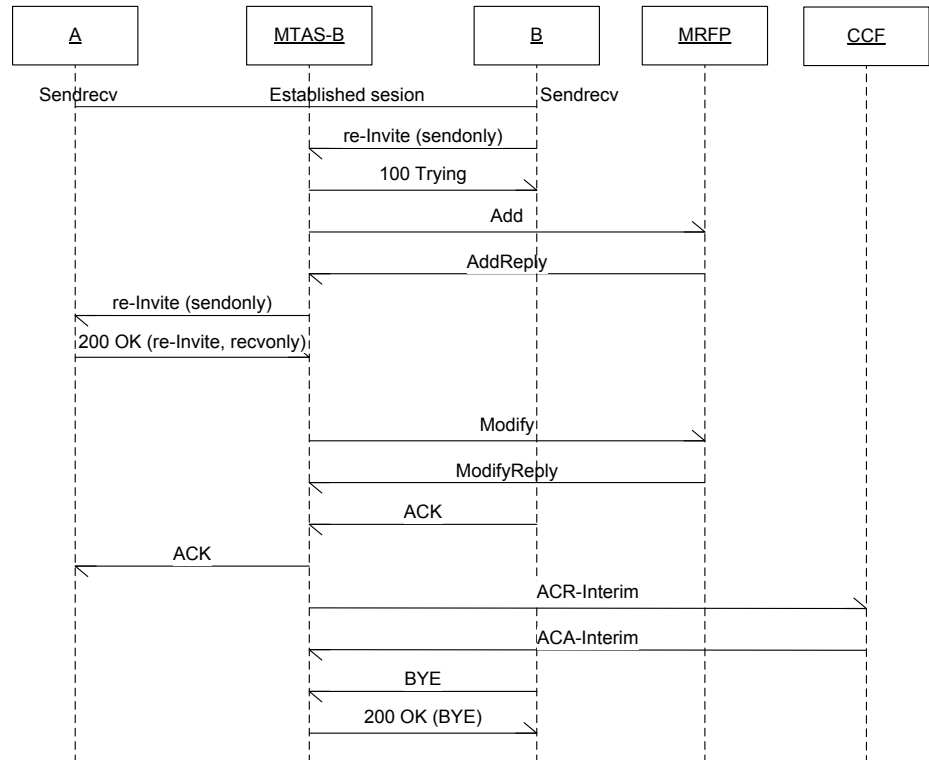


Figure 28 B puts the call on hold, and then sends BYE within the STOD timer to invoke STOD.

B sends a re-Invite to A to put him or her on-hold – media direction is set to sendonly. A accepts the request by sending 200 OK to the re-Invite by setting the media direction to recvonly. Hold service starts to play an announcement to A if configured.

B sends a BYE within a predefined period after the successful hold to invoke STOD. STOD sends the 200 OK to the BYE to B.

STOD transfers the session:

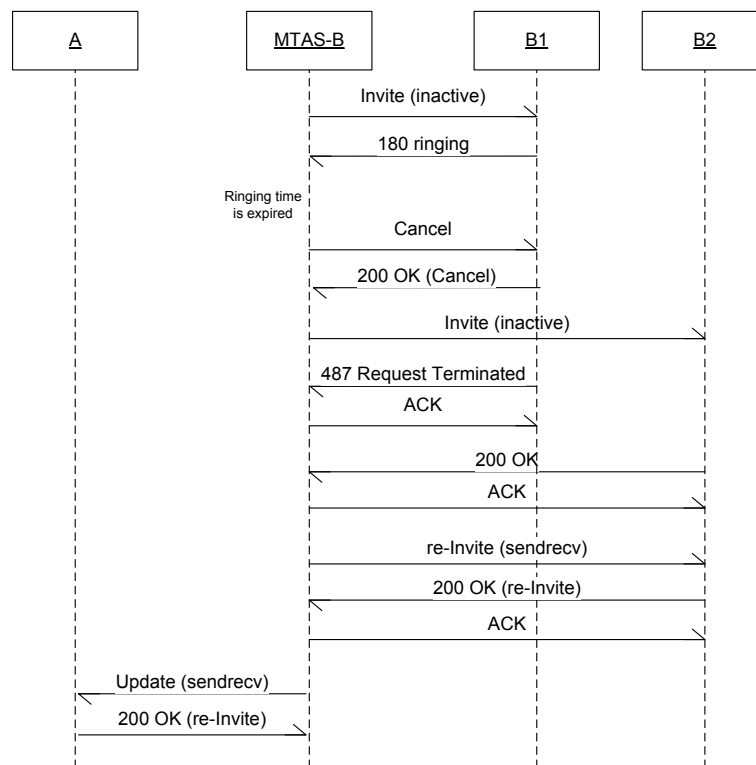


Figure 29 Serial STOD service rings B1 first, ring timer expires, and then B2 is rung, which answers the call.

STOD starts to work, by alerting the targets. In the example call flow above, it uses serial ringing, so it sends the Invites one by one, and waiting for the final response. STOD send the original Invites with SDP, setting the media direction to inactive.

First it tries to call B1 but he or she does no answer within the ringing period, so MTAS decides to Cancel the call.

After the 200 OK for the Cancel, STOD starts to alert to second target, B2 in this case.

In the meanwhile B1 responds to the invite with 487 Request Terminated, and MTAS sends an ACK to it.

B2 accepts the INVITE, and MTAS sends an ACK to it.

After the call is established, MTAS sends a re-INVITE to B2 to set the media to sendrecv.

B2 accepts the re-INVITE, and MTAS sends an ACK to it.

When the re-Invite is ready, and the session is established again with B2, STOD sends an Update to A with the new values within it.



It will explicitly finish the on-Hold state of A, since the update will contain an SDP with sendrecv media direction.

The call session is established between A and B2.

2.11.2.4 STOD triggered by Call Pull

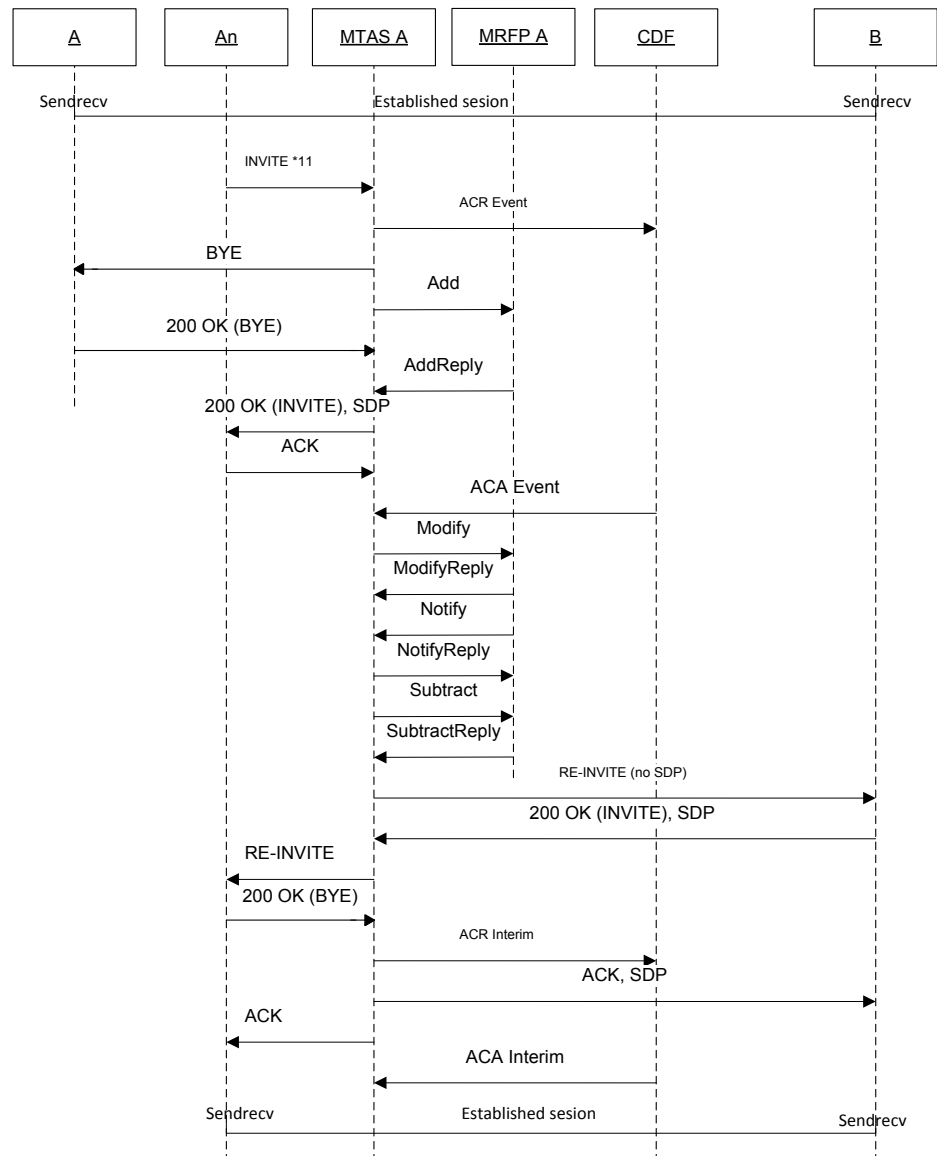


Figure 30 Call Pull invocation with feature code..

An sends an Invite with STOD Call Pull Feature Code to MTAS A to initiate transfer of the existing call where device A is involved as Caller or Called party. MTAS A immediately disconnects device A from the call in order to prevent it from terminating the call.



MTAS A initiates a short announcement to establish the dialog.

RE-INVITE without SDP is sent to B to trigger media negotiation between B and An (new in the call – replacing A). B replies with SDP offer, that is used (after modifications) to send RE-INVITE to An. An responds with SDP answer. SDP negotiation is finished when B receives ACK with SDP from MTAS A.

The call is established between A and B.

2.11.3 Service interactions

2.11.3.1 Flexible Communication Distribution (FCD)

The STOD (Call Push) service utilizes the Flexible Communication Distribution (FCD) service to perform distribution in accordance with the defined rule set; this independent of the administrative state of the FCD service.

The FCD to Primary User's Devices function is applicable also to STOD initiated distributions in case this FCD option is enabled.

The FCD - Divert Primary (FCDDP) option is not invoked at STOD initiated distribution.

When the STOD (Call Push) service is active but no STOD specific ruleset is provisioned to the served user, the STOD service will use the FCD specific ruleset, if provisioned.

STOD (Call Pull) is not available for sessions distributed to Related Users.

2.11.3.2 Ad-Hoc Conference

2.11.3.2.1 Originating STOD

An IMS user can use the STOD service, as an originating Conference Creator and transfer a session in a Standalone Ad-Hoc Conference to a Primary or Related user. The 'isfocus' feature parameter is forwarded to the Primary or Related user. The Originating STOD service is executed in an originating AS separate from the Conference Focus AS.

An IMS user cannot use the STOD service, as an originating Conference Creator and transfer a session in a Collocated Ad-Hoc Conference, if STOD is invoked by Communication Hold followed by SIP BYE it will be handled as a normal BYE and the Conference will be terminated. Terminating STOD

An IMS user can use the STOD service, as a terminating Conference Participant and transfer a session in a Standalone or Collocated Ad-Hoc Conference to a Primary or Related user. The 'isfocus' feature parameter is forwarded to the Primary or Related user.



The Terminating STOD service is executed in a terminating AS separate from the Conference Focus AS.

2.11.3.2.2 Call Pull

An IMS user cannot pull a conference session when being the Conference Creator in a Collocated Ad-Hoc Conference. The Call Pull attempt is in this case rejected with error code 403 and informative warning text.

Call Pull of conference sessions where the served user is an ordinary conference participant is supported. Any active conference event subscription is terminated when Call Pull takes place.

2.11.3.3 Carrier Select and Carrier Select Rn

The STOD service allows the Related User identity to include a Carrier Select code.

The Related Users are subject to Congestion Control in Carrier Select Rn, which means that the CrankBack service can intercept an error response from the user and if it has a specific Q.850 code, CrankBack will repeat the call.

2.11.3.4 Carrier Pre-select and Carrier Pre-Select Rn

Initial INVITE requests sent by the STOD service to Related Users are subject to the Carrier Pre-Select or Carrier Pre-Select Rn service.

2.11.3.5 Hold service

There is no STOD specific announcement towards the originating user. The Hold service will play the Hold announcement during the STOD invocation – if configured.

2.11.3.6 Communication Diversion

The Communication Diversion (CDIV) service is not invoked when the STOD service transfers the session to a Primary User.

For Call Pull, if the established session is diverted Call Pull request may be rejected depending on configuration data..

2.11.3.7 Call Admission Control

The Call Admission Control counters are not affected when the STOD is invoked.

The User and Group Call Admission Control counters are not affected when STOD is invoked.

A STOD call will only count as a single originating call for both User and Group Call Admission Control.



The User Call Admission Control employs predictive algorithm to determine if a successful Call Pull invoked via Replaces header would breach the UCAC limits and may reject the pull request.

Call Pull requests invoked via SSC are counted as a new session by the Call Admission Control service.

2.11.3.8 Communication Barring

Communication Barring is working on outgoing Invites from STOD to other users than the served user. Outgoing Communication Barring is also working in case of STOD.

Related User identities cannot be an identity which would be barred by OCB or included in the CDIV black list.

If Call Pull is initiated by a roaming mobile user and roaming condition is included in the subscriber's barring rules then Call Pull request should be rejected.

2.11.3.9 Call Completion

When STOD is invoked by Communication Hold followed by SIP BYE in an originating session, after the Communication Completion (CCxx) service has been successfully performed, the STOD service cannot be performed, this will instead be handled as a normal BYE and the session will be terminated.

2.11.3.10 Advice of Charge

Credit warning and credit limit reached announcements during a transferred session is played to the credit owner only if the user is involved in the transferred call. Those cases are handled as normal mid-session and session termination credit announcements.

When a STOD user that subscribes to Advice of Charge (AOC) transfers a session to a Related User, no Advice of Charge (AOC) information are sent to the Related User. The MTAS does not initiate an AOC Session with the OCS for the originating charging session to Related User,

When a STOD user that subscribes to AOC transfers a session to a Primary user, AOC information are continued to be sent to the served user, if the terminal supports the AOC MIME Type and if the SIP INFO method is allowed.

2.11.3.11 Dialed Number Mapping

The Dialed Number Mapping (DNM) service will not be invoked after the STOD is invoked and distributes the call. Therefore the seven-digit and short-code number must never be used as the target of STOD service.



2.11.3.12 Emergency State

Call Pull requests from users in Emergency Callback Window state can be allowed by setting depending on configuration data,

2.11.3.13 Flexible Service Format Selection

The STOD service can be suppressed by means of Flexible Service Format Selection (FSFS) service.

When receiving SIP INVITE the MTAS will invoke the FSFS service before the STOD service. The FSFS service provides an indicator for MTAS to suppress the STOD service.

When the FSFS service suppresses the STOD service, the SIP BYE that can trigger the STOD service is processed as if the service was not active. MTAS then forwards the SIP BYE towards the other side and consequently, the communication session is released.

2.11.3.14 Number Translation

Initial INVITE requests sent by the STOD service to Related Users are subject to the Number Translation service.

2.11.3.15 Realtime Transfer of Tariff Information

After transfer of the session to a Related User, Realtime Transfer of Tariff Information (RTTI) received from terminating network is sent to OCS for the originating charging session to Related User.

The received RTTI is removed from the SIP messages before passing them to the other side which remains in the communication.

2.11.3.16 Subscriber Credit Notification

After transfer of the session to a Related User no credit announcements are played for the served user and also not to the Related User.

After transfer of the session to a Primary user, the credit announcements are continued to be played for the served user.

2.11.3.17 Explicit Communication Transfer (ECT)

For Call Pull, if the established session is a post ECT session Call Pull request should be rejected.

2.11.3.18 3-Party Session (3PTY)

For Call Pull, if the established session is 3PTY initiator session Call Pull request should be rejected.



2.11.4 Configuration

- On/Off on node level
- Configured and provisioned on user level with subscriber data

2.11.5 Performance management

The following performance counters are provided by MTAS to evaluate the usage and quality of service:

- The number of STOD invocations
- The number of successful STOD invocations

2.11.6 Fault management

For information on the alarm, refer alarm OPI [58]

2.12 Operator Controlled Transfer

2.12.1 Description

The Operator Controlled Transfer (OCT) service implements transferring of a call, upon request from the called party in the originating MMTel AS. The called party is an Operator Transferor. MTAS accept a SIP REFER request from a called party on an ongoing session when MMTel AS is aware of that it is coming from Operator Transferor and sends SIP INVITE to the Transfer Target.

2.12.2 Example call flow

The OCT service defines three different user roles:

- UA-A
- Transfer Target
- Operator Transferor

UA-A is the calling party to the operator transferor.

Transfer Target is the called party for the operator transferor call setup also called the target user for the transfer.

The Operator Transferor is the user that provides transfer assistance for the calling user.



2.12.2.1 Establishing call to Operator Transferor

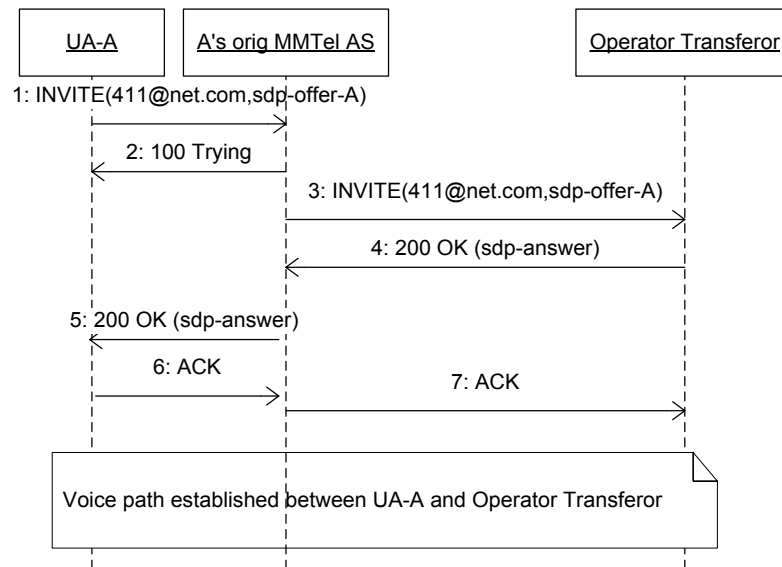


Figure 31 Establishing call towards Operator Transferor

Originating MMTel AS receives an INVITE with operator configured phone number. Originating MMTel AS recognizes that it is a call to an Operator Transferor by checking the phone number in the request Uri is included in configured list of valid phone number for OCT service and that the phone number is an Operator Service Number (OSN) or a Nationally Significant Number (NSN).

The Operator Transferor accepts the call.

2.12.2.2 Transfer accepted

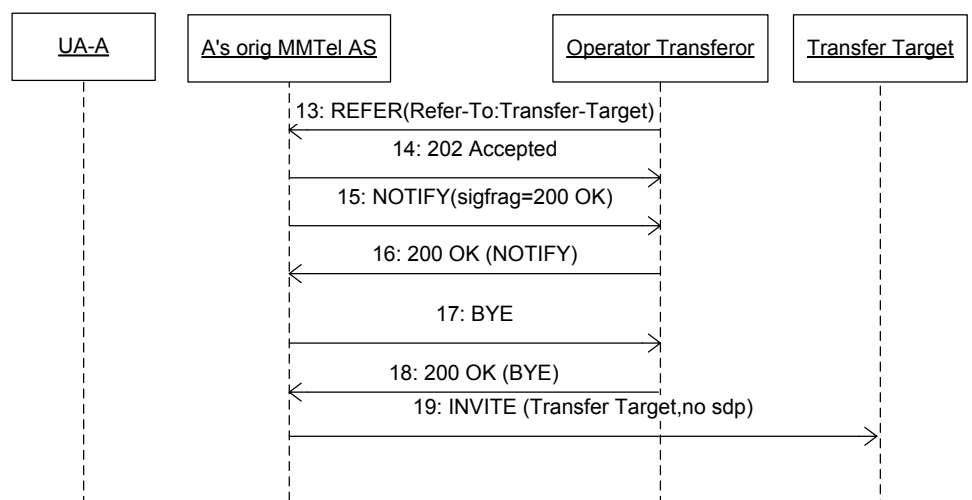




Figure 32 REFER accepted and INVITE sent to the transfer target

Originating MMTel AS accepts the REFER request when it is coming from the Operator Transferor. MMTel AS notifies the Operator Transferor about the transfer and disconnects the Operator Transferor. MTAS initiates the call setup to the Transfer Target.

2.12.2.3 Ringing before answer from target user

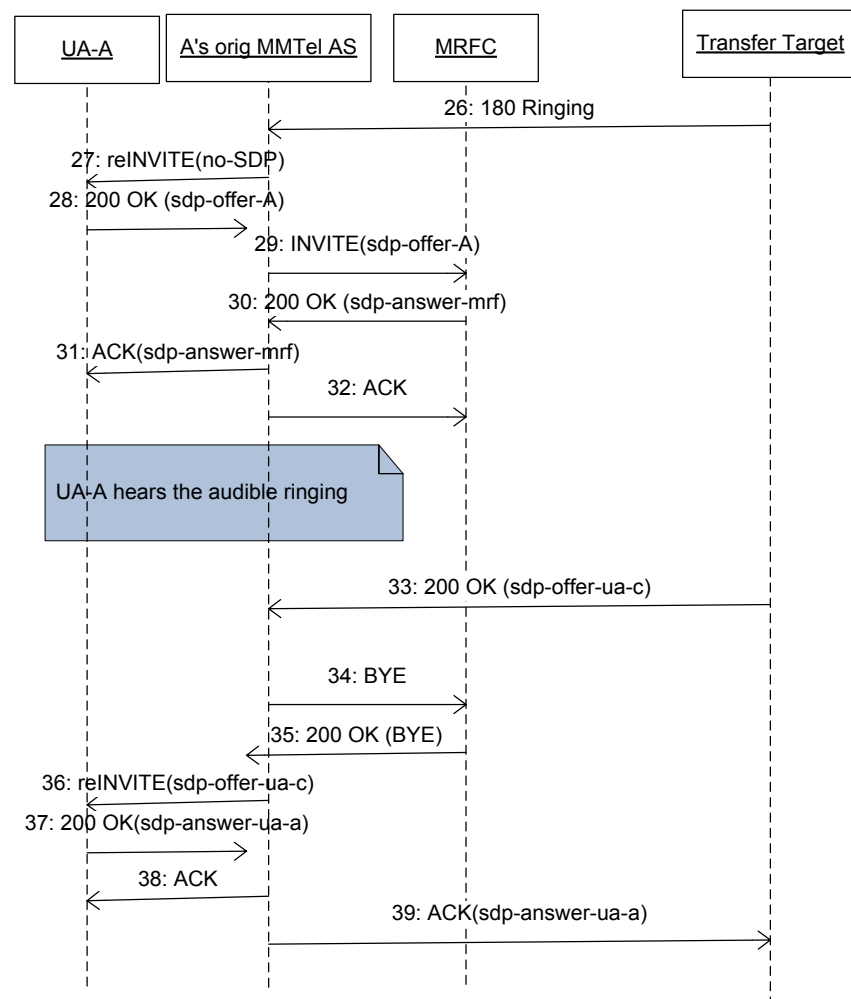


Figure 33 Ringing before answer from target user



The first provisional response from the target user is 180 Ringing. MMTel AS uses the MRFC to play the ring back tone to user A. When the Transfer Target answers the call MMTel AS informs the MRFC to stop the ring back tone and user A will hear the media from the target user.

2.12.2.4 Early media before answer from the target user

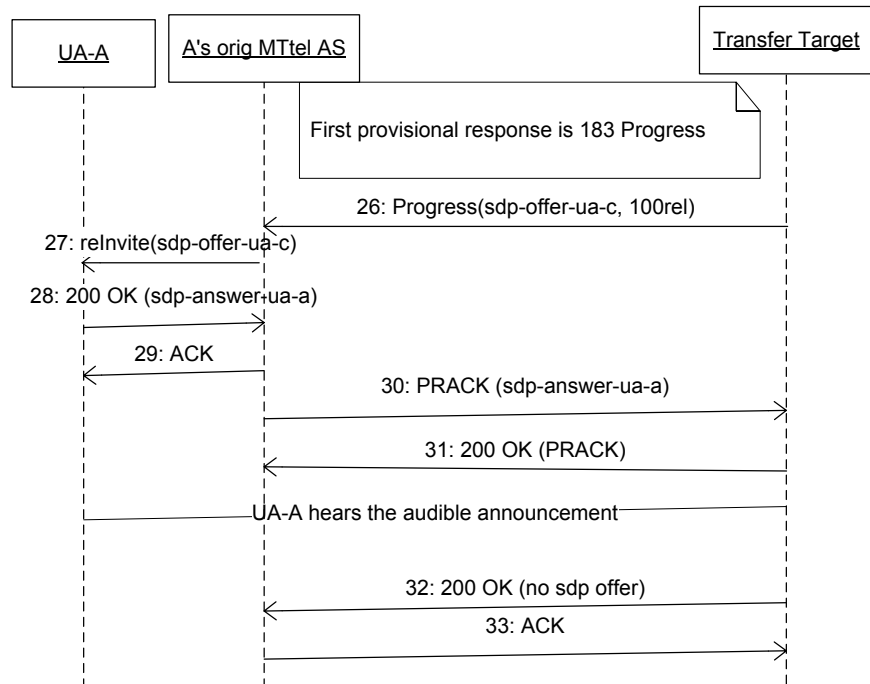


Figure 34 Early media before answer from the target user

The first provisional response from the target user is 183 Progress with an SDP offer. After media negotiation user A will hear the early media from the Transfer Target. When the target user answers the call with no changes in the media the call is established.



2.12.2.5 Target user is busy

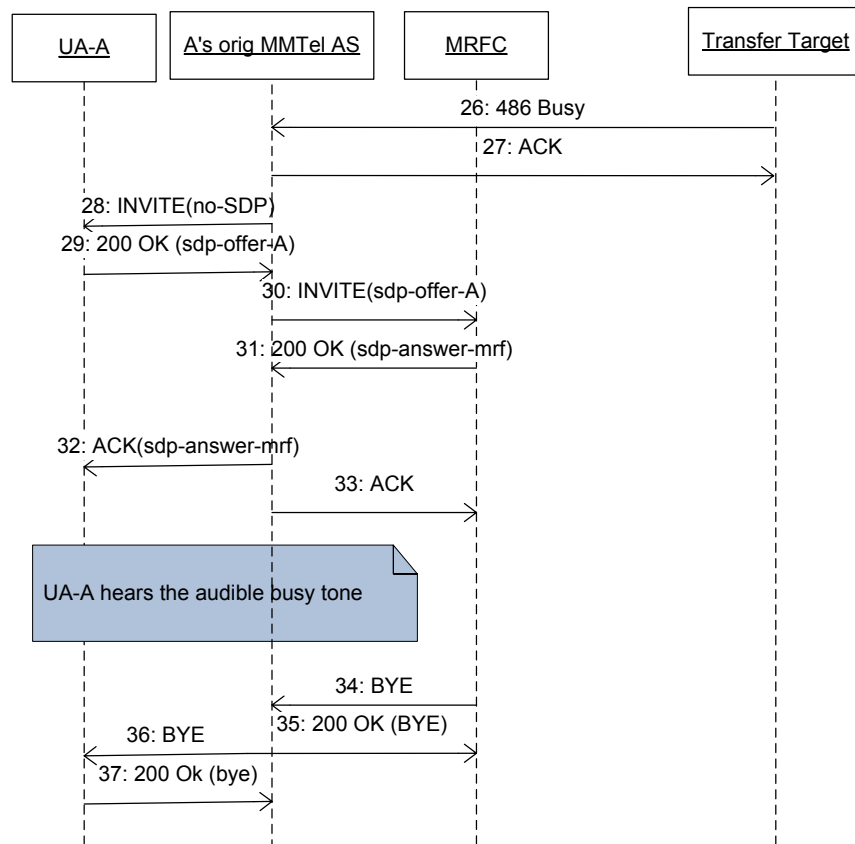


Figure 35 Target user is busy

The first response from the target user is 486 Busy. MMTel AS uses the MRFC to play the busy tone to user A. When the configured busy tone expires, MMTel AS sends BYE to user A.

2.12.3 Charging

MMTel AS sends charging information for the call between the calling user A and the Operator Transferor and for the call between the calling user A and the Transfer Target when charging apply for the calling user A.

2.12.4 Service interactions

2.12.4.1 MMTel Basic Call

A call to the target user is subject to number normalization, Outgoing Call Barring, Carrier Select and Identity Presentation.

The established call between calling user A and the Transfer Target is handled as any 2-party session, so any in-session service can be invoked.



2.12.4.2 Communication Diversion

It is not possible to make an Operator Controlled Transfer when the call is established due to communication diversion.

2.12.4.3 Ad-hoc and Scheduled Conference

It is not possible to make an Operator Controlled Transfer when the call is included in a multi-party call.

2.12.4.4 Three Party Call (3PTY)

It is not possible to make an Operator Controlled Transfer when the call is included in a multi-party session.

2.12.4.5 Operator Controlled Transfer

It is not possible to make an Operator Controlled Transfer when the call is established due to Operator Controlled Transfer.

2.12.5 Configuration

- Administrative state (enable/disable)
- A list of phone numbers to the Operator Transferor
- A ring back tone announcement
- A busy tone announcement
- A failure announcement

2.12.6 Performance management

The following performance counters are provided by MTAS to evaluate the usage and quality of service:

- The number of call to an Operator Transferor
- The number of call transfers from an Operator Transferor

2.13 Call return

2.13.1 Description

The Call Return service informs an end user about the last incoming call via a Supplementary Service Code (SSC) and then provides the possibility to make a call back.



2.13.1.1 Save Last Incoming call

Call Return has two configurable modes on node level for identification of the last incoming call, either last unanswered call or last incoming call independent on whether it was answered or not. The last incoming call is identified by that a 180 Ringing is received and for unanswered calls no 200 OK is received at the terminating MMTel AS from the called user.

The saved last incoming call will be kept by MMTel AS until either a new incoming call is received or the user is de-registered. During this time the user can use Call Return service to call back the caller at any time.

2.13.1.2 Invoke Call Return

The served user can make a call back to the caller of the last incoming call by using the SSC in the initial INVITE.

- The served user dial the supplementary code for CR (it is configurable),
 - One SSC for automatic call back
 - One SSC for announcement prompt is an interactive mode, where the user is prompted for confirmation
- When the SSC is for automatic call back the call back will be setup immediately when no announcement is configured. When an announcement for automatic call back is configured the served user can listen for an announcement about the last incoming call containing date/time when the call was received and the phone number. The call back will be automatically setup after the announcement is finished and the configured call back timer expires. The timer starts after the announcement is finished.
- When the SSC is for announcement prompt served user can listen for an announcement about the last incoming call date/time, phone number and the configured call return code to be used for confirmation. The served user presses the call return code for call back.
- A call attempt is made to the caller of the last saved incoming call
- When there is no last incoming call or the last incoming call includes a privacy header, the served user will be informed through an announcement about it and no call back will be performed,

2.13.2 Example Call Flow

2.13.2.1 Save last incoming call

The data to store is P-Asserted-Identity header(s), if TEL URI, or a SIP URI with embedded TEL available or a SIP URI, Privacy, and the date and time when the call is received

The following preconditions must be fulfilled for triggering the saving of last incoming call.



- The served user is provided with a CR subscription.
- The administrative state is unlocked for CR service
- The license is valid for the CR service

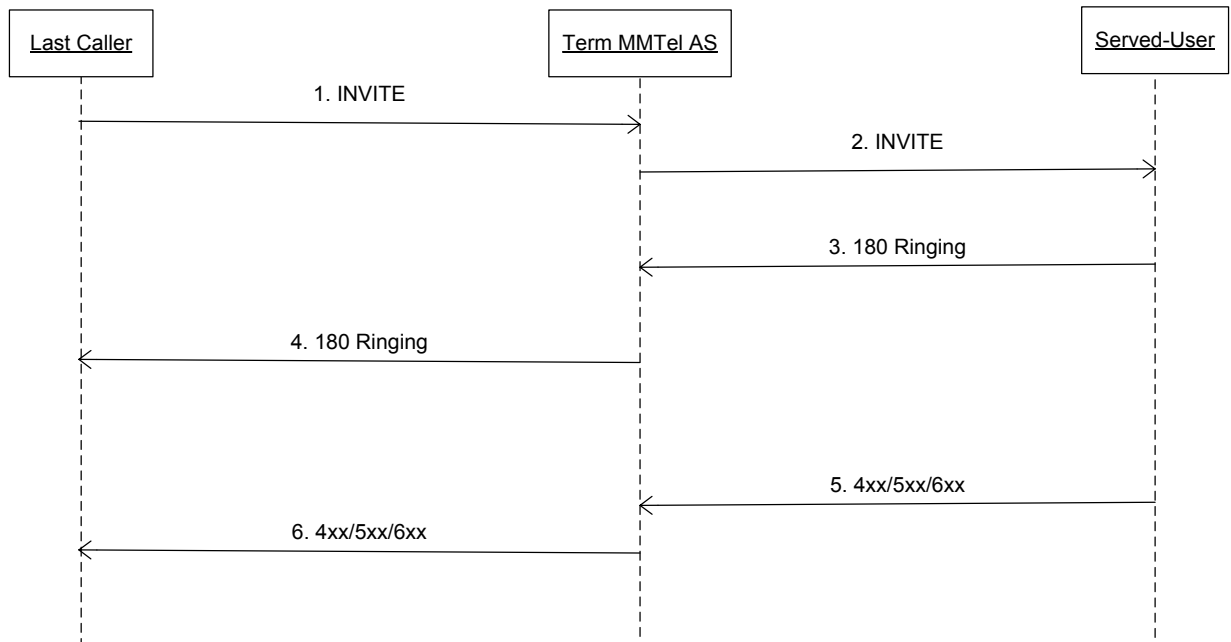


Figure 36: Save last incoming call

1. MMTel AS receives the INVITE message.
2. The served user receives the INVITE message.
3. MMTel AS receives 180 Ringing.
 - The served user must have been at least alerted before the call can be saved as the last incoming call. If the CR service is configured to save data for any incoming call it is saved at this stage. .
4. 180 Ringing is sent to the last caller.
5. MMTel AS receives 3xx/4xx/5xx/6xx. If the CR service is configured to save data for the last unanswered incoming call only, the data is saved now
6. 4xx/5xx/6xx is sent to the last caller.



2.13.2.2 Invoke Call Return for automatic call back

Pre-Conditions

- The subscriber has the CR subscription provisioned.
- The administrative state is unlocked for the CR service.
- The license is valid for the CR service.
- The CR SSC for automatic call back is configured and the announcement for automatic call back is configured to contain phone number, date and time. The time is included in the announcement when the user invoke the service the same day as the last incoming call was saved, otherwise the announcement includes the date of when the last incoming call was saved.

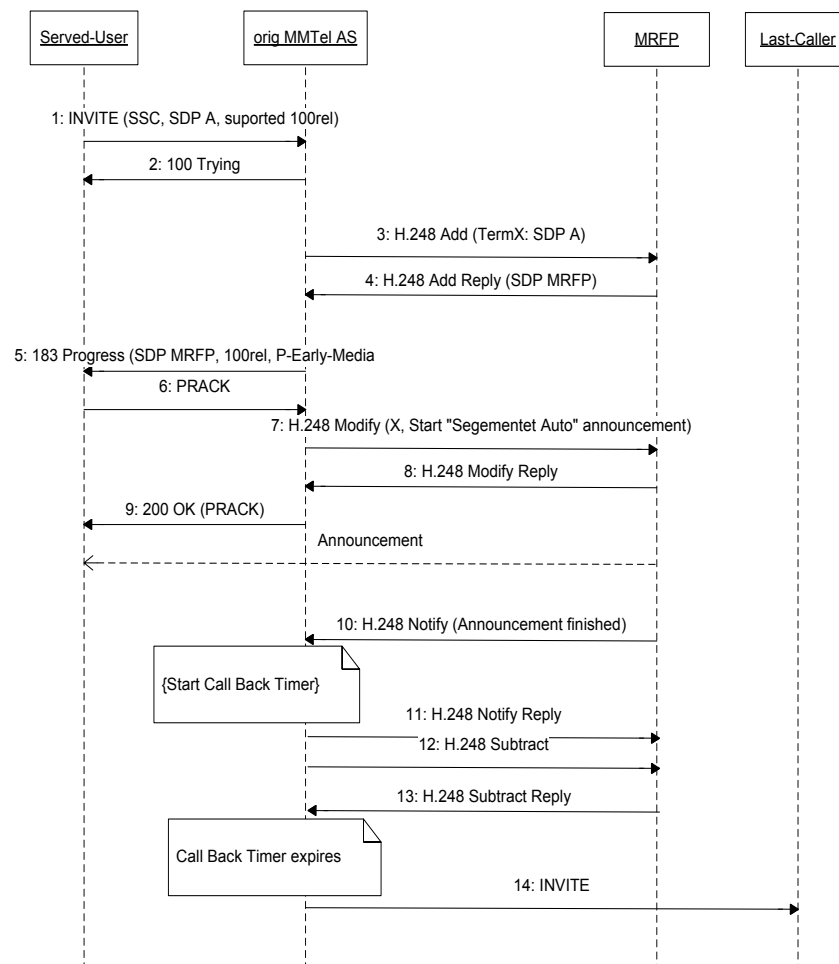


Figure 37 Call Return invoked with SSC for automatic call back

1. Served User sends INVITE with SSC for automatic call back including an SDP offer



2. MMTel AS responds with 100 Trying.
3. MMTel AS detects that Call Return service for automatic call back is invoked and an announcement name for automatic call back is configured. Resources are allocated in MRFP to play announcement.
4. Add Reply with MRFP SDP.
5. A provisional response 183 with MRFP SDP is sent to Served User.
6. PRACK received from Served User.
7. MMTel AS orders MRFP to play the announcement with the date/time and the phone number of the last saved incoming call. The date and time is in the served user home time zone with compensation for daylight saving time, when the user is provided with the time-zone-area. When the time-zone-area is not provisioned and the utc-offset is provided, the date and time is in the served user time zone, Otherwise the date and time is in the time zone where MMTel AS is installed with compensation for daylight saving time
8. Reply from MRFP that announcement is played.
9. 200 OK on PRACK is sent to Served User.
10. Notification from MRFP that announcement is finished. MMTel AS starts the configured call back timer.
11. Reply.
12. Release resources at MRFP.
13. Reply.
14. When the call back timer expires, MMTel AS updates the INVITE by replacing the request URI with the saved PAI and replacing the TO header with the saved PAI. The INVITE is sent to Last Caller.

2.13.3 Configuration

Examples of node-level configuration parameters related to the Call return service are:

- Administrative state (enable/disable)
- Call state of last saved call (last incoming call/last unanswered call)
- Announcements configuration
- Automatic call back timer
- SSC codes



2.13.4 Provisioning

MMTel AS operator can provide the user with the Call Return service.

2.13.5 Performance management

The following performance counters are provided by MMTel AS for the service:

- Successful attempt. Increment when INVITE sent by CR without announcement prompt.
- Unsuccessful attempt, external error. Increment when INVITE sent by CR without announcement prompt
- Unsuccessful attempt, internal error. Increment when INVITE sent by CR without announcement prompt.
- Successful attempt. Increment when INVITE sent by CR with announcement prompt.
- Unsuccessful attempt, external error. Increment when INVITE sent by CR with announcement prompt.
- Unsuccessful attempt, internal error. Increment when INVITE sent by CR with announcement prompt.

2.13.6 Fault management

For information on the alarm, refer alarm OPI [59].

2.14 Hotline

2.14.1 Description

Hotline service enables users to call hotline number without dialing it.

MTAS supports three variants of the Hotline service:

- Unconditional Hotline
- Instant Hotline
- Delayed Hotline



2.14.1.1 Unconditional Hotline

The Unconditional Hotline service routes all outgoing calls of the served user to an operator predefined destination, e.g. automatic rerouting to customer care. The service is typically applied in case the served user has not paid subscription bills to the operator. All outgoing calls except emergency calls are automatically re-routed to the hotline destination independent of dialed number. The Request-URI of the initial INVITE is replaced with the predefined hotline number.

2.14.1.2 Instant Hotline

The Instant Hotline service allows the served user to reach a hotline destination by providing an operator predefined service code (e.g. "***") as dialed number. Once the service recognizes the service code, the Request-URI of the initial INVITE is replaced with an operator predefined hotline destination.

The service code is typically generated by an IAD serving the served user, either immediately at off-hook or in case no digits were provided within a predefined time limit following the off-hook event.

2.14.1.3 Delayed Hotline

The Delayed Hotline service is almost identical with the Instant Hotline service with the following differences:

- The served user can control activation/deactivation of the service as well as what hotline destination number to use by self-service provisioning.
- The served user may define the own voice mail function as the hotline destination.
- The service code used to trigger the Delayed Hotline service (i.e. the called number) can be defined per served user if wanted. If defined it is used instead of the configured node level value.



2.14.2 Example Call Flows

2.14.2.1 Unconditional Hotline

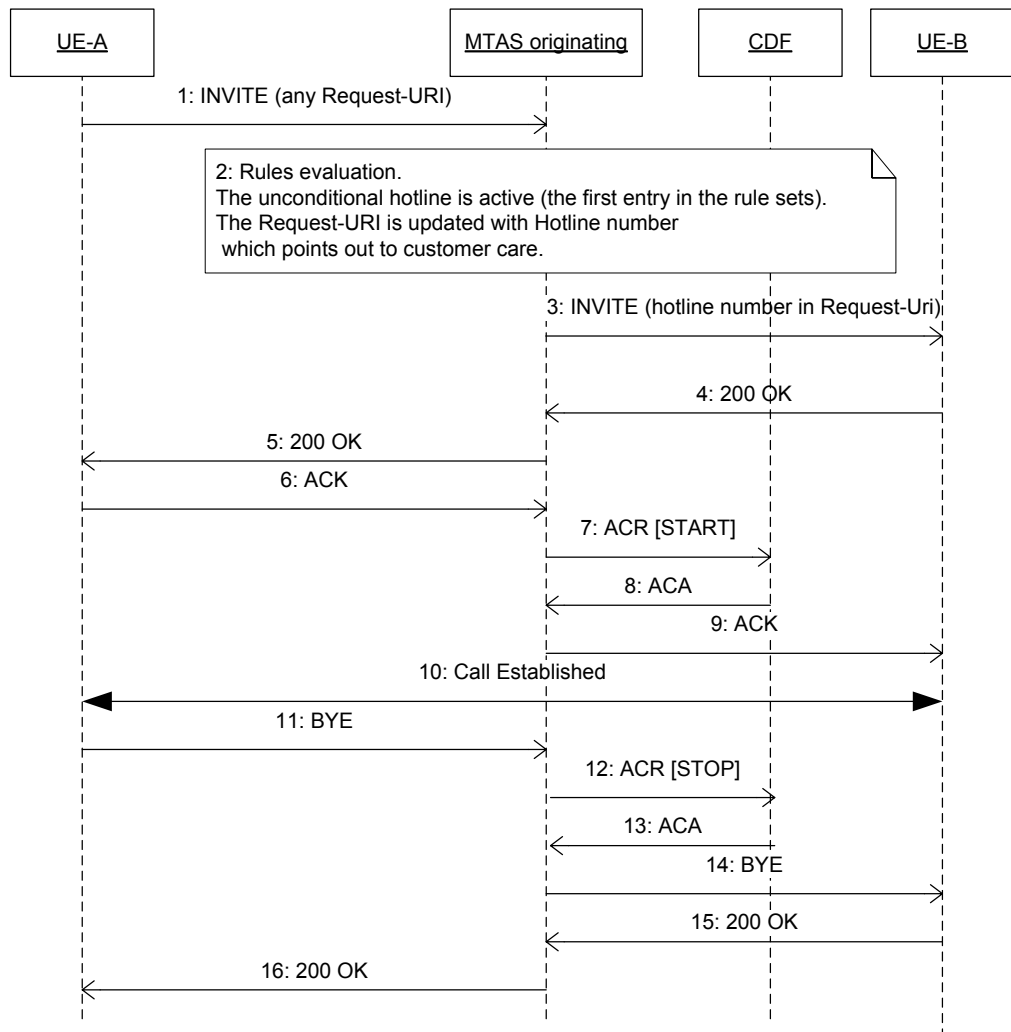


Figure 38 Unconditional Hotline

1. User A sends an INVITE request to a number which is not an emergency number.
2. MMTel AS checks the Hotline settings for user A. As Unconditional Hotline is active MMTel AS replaces the Request-URI of the INVITE with the hotline number provisioned by the operator for user A.
3. The INVITE is sent to user B (customer care).
4. MTAS receives 200 OK as the response to the SIP INVITE request.



5. MTAS forwards 200 OK towards user A.
6. MTAS receives ACK as the response to 200 OK.
7. If offline charging is active, then MTAS sends ACR Start request towards the CDF.
8. MTAS receives a response to the ACR Start request with successful indicator.
9. MTAS forwards ACK towards user B.
10. The call is established between user A and user B.
11. MTAS receives BYE.
12. MTAS sends ACR Stop request towards the CDF.
13. The charging session with the CDF is terminated.
14. The BYE is forwarded to user B.
15. MTAS receives 200 OK response to the BYE.
16. The 200 OK is forwarded to user A.

2.14.3 Charging

The use of Hotline is reported in charging messages generated during the setup of an MMTel session. The Hotline number is reported in the charging data record as called-party-id. The Request-URI replaced by Hotline is reported as requested-party-id.



2.14.4 Self administration

The end user can by itself administer the End user controlled Hotline via the Ut-interface and via SSCs.

- Activate/de-activate
- Set Hotline number
- Interrogate/Read service status

2.14.5 Provisioning

MTAS enables the operator to configure the service on user-level through the CAI3G interface. Possible settings are:

- Activate/de-activate Unconditional/Operator controlled/End user controlled Hotline
- Set Hotline number for each service type

2.14.6 Configuration

The node-level configuration parameters related to the Hotline services are:

- Enable and disable of the Hotline service (administrative state)
- The Operator/End user controlled Hotline service codes used in the Request-URI

2.14.7 Performance Management

Performance counters related to the Hotline service are:

- Number of successful invocations of Unconditional/Operator controlled/End user controlled Hotline
- Number of unsuccessful invocations of Unconditional/Operator controlled/End user controlled Hotline

2.14.8 Fault Management

For information on the alarm, refer alarm OPI[60] .

2.15 Dial Plan

The Nodal Dial Plan, the OTP-controlled per-VTP Dial Plan, and the VTP-controlled per-VTP Dial Plan are defined as MO attributes.

Each dial plan is defined by a list of allowed numbers and a list of excepted numbers, which apply to numerical Request URIs (i.e. SIP URIs with a user=phone parameter and tel URIs), and a list of allowed domains, which applies to non-numerical Request URIs.



The Nodal Dial Plan allows the operator to specify all the addresses that can be reached by the end-users supported by this MTAS.

The OTP-controlled per-VTP Dial Plan allows the operator to specify all the addresses that can be reached by the end-users belonging to each VTP.

The VTP-controlled per-VTP Dial Plan allows the VTP operator to specify all the addresses that can be reached by the end-users belonging to that VTP.

The lists are stored in active and standby tables. The lists in active tables are used in traffic and can only be read. The lists in standby can be managed by the operator and when ready they can be activated, direct or scheduled within two weeks time. This means the tables are switched.

2.16 Communication Barring

2.16.1 Description

The Communication Barring (CB) supplementary service enables a user to have barring of certain categories of communication. This service is standardized by 3GPP for communication session initiation scenarios, see ref [18].

For an initial INVITE, the sub-function Outgoing Communication Barring (OCB) is executed on the originating MMTel AS and the sub-function Incoming Communication Barring (ICB) is executed on the terminating MMTel AS. There is no difference in the execution due to the registration state.

The function is also executed on any SIP messages within a communication session that contain a media offer. The purpose of this functionality is to enforce CB media rules throughout the lifetime of a communication session.

The CB is a series of checks which determine if a communication shall be barred or allowed. The following sections cover the different types of checks that make up the CB service.

2.16.1.1 Rule Based Barring

The rules are built up by conditions and actions, which can be combined in many ways to express when a communication shall be barred or allowed.

The different conditions can be based on:

- Whether the identity match a user, domain, range of numbers, etc.
- Whether the calling user's identity is anonymous
- Time periods the rule is valid and not
- What media the communication contains
- Whether the communication has previously been diverted



- The carrier selected with on-demand (dialed) by the Carrier Select Rn service.
- Whether the user is roaming
- Whether the call is international
- Whether the call is international but to home country
- Whether the call is unconditional

In case the Multi Subscriber Number service is enabled also the served user's MSN identity may be used in the condition. For further details see chapter 2.36.

The different actions are:

- Allow or bar communication
- Bar incoming communication due to that Do Not Disturb is active
- Play generic or segmented announcement

In identity based rules, it is the served user's identity that is subjected to checking using the rule conditions.

It is possible to define valid time periods, invalid time periods and also to assemble a complex time condition based upon several calendar sub-conditions (times of day; days of the week; calendar months; calendar weeks; private and public holidays; daily, weekly and monthly repetitions etc.)

In media based outgoing and incoming communication barring (OCB Media and ICB Media) rules, the condition always refers to the newly added media in a media offer. For session initiation this means that all media lines are checked. For session updates, only new media lines are checked.

2.16.1.2 Incoming Communication Barring – Do Not Disturb (DNDCB)

When the end-user activates Do Not Disturb for CB all ICB rules with the Do Not Disturb action are active.

2.16.1.3 Anonymous Communication Rejection (ACR)

ACR is a particular case of Incoming Communication Barring, which bars anonymous callers. The incoming communication is rejected by a SIP response with result code 433 (Anonymity Disallowed).

2.16.1.4 Barring conditions for mobile UE

The incoming and outgoing communication barring based on roaming and the outgoing communication barring based on whether the call is international or international and not the home country are barring conditions for mobile UEs only. To verify any of these conditions, the UE's current location information in CS and/or PS network is fetched from HSS.



2.16.1.5 Rule definition

Examples of barring rules that can be defined in MTAS are:

- Bar all outgoing real-time text communication except to sip:+3611234567@ericsson.com;user=phone
(In this example the conditions are based on Media + Destination)
- Bar every type of communication to tel:+3611234567 except video between 08.00-11.00 2009-10-14
(In this example the conditions are based on Destination + Media + Time)
- Bar all calls to numbers in Coventry, UK
(In this example the conditions are based only on Destination)

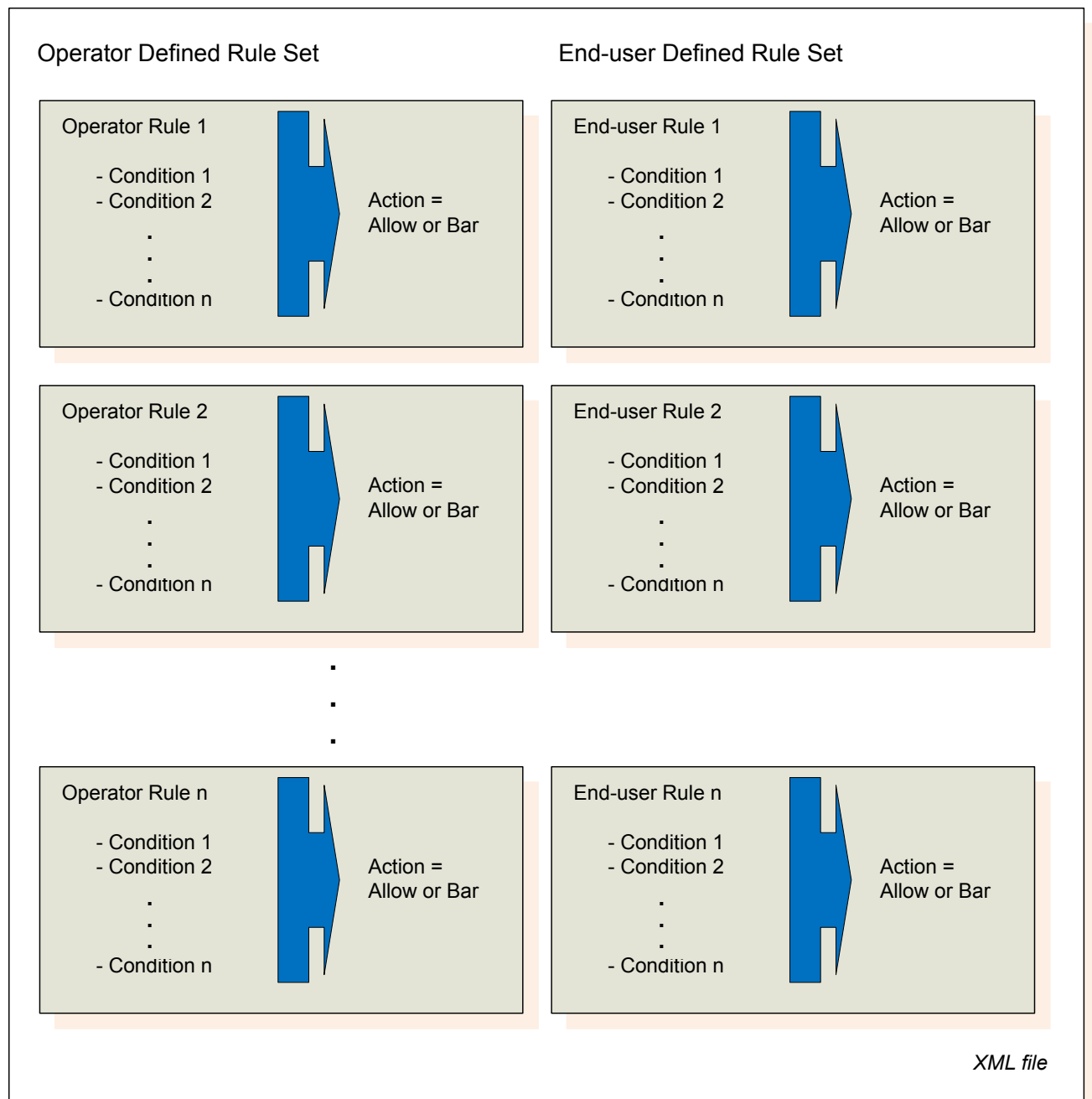




Figure 39 - Barring rule set

The rules are arranged in an OCB rule set and a ICB rule set. The evaluation method is characterized as follows:

- All the rules in the rule set are evaluated sequentially to test if their respective condition(s) are true.
- A rule is said to be matched if all conditions are evaluated as true.
- If exactly one rule matches within the rule set:
This rule's specified action is executed, i.e. if "Allow" then the call is allowed, if "Bar" then the call is barred.
- If more than one rule matches within the rule set:
The communication is barred only when all rule's action is Bar (i.e. "Allow" takes precedence over a "Bar").

It is possible to deactivate a rule without deleting the rule configuration.

2.16.1.6 Basic Barring - Barring Programs

This feature enables the operator to define Barring Programs and provides the end user with the possibility to select from the defined Barring Programs to bar his/her own outgoing communication. The outgoing communication restriction in Barring Programs is always based on destination.

Barring Programs are a set of Barring Categories that consist of a list of numbers to be barred and a list of numbers to be exempted from the barring.

The lists are stored in active and standby tables. The lists in active tables are used in traffic and can only be read. The lists in standby can be managed by the operator and when ready they can be activated, direct or scheduled within two weeks time. This means the tables are switched.

For Barring Programs the comparison is based on front-substring match.

For example:

The identity is tel: +3611234567 or
sip:+3611234567@ericsson.com;user=phone.

The front-substring comparison against a list entry of +361 would match, but a front-substring comparison against a list entry of +3615 would not match.

The end-user selects from the offered, node-level Barring Programs via Supplementary Service Codes (SSC) or via the XCAP based Ut-interface.

MTAS supports two Barring Program types (schemes).



In the first type scheme, known as Single scheme, the operator defines a relatively long list of relatively complex node-level Barring Programs from which the end-user can select only one. For example:

- 1 International
- 2 Mobile
- 3 Premium
- 4 Mobile and Premium
- 5 Mobile and International
- 6 Premium and International
- 7 Mobile, Premium, and International,

*33*1# might turn ON Barring Program 1 (International), and #33# might turn OFF the Barring Program.

In the second style of Barring Program (called Multiple scheme) the operator offers a shorter list of simpler, node-level Barring Programs. For example:

- 1 International
- 2 Mobile
- 3 Premium

In that case the end-user can select one or more programs at the same time and create his/her own outgoing barring plan by combining the offered programs.

2.16.1.7 Operator Black Lists

The operator can create a global ICB Black List to filter out the communication coming into the terminating MTAS and a global OCB Black List to filter out the communication originated from the originating MTAS and a global CDIV Black List to filter out communication diverted by MTAS (see chapter 2.4).

If the Wholesale function (see chapter 2.53) is enabled and Wholesale domains are defined on the node then the Black Lists can be configured in Operating Telephony Provider (OTP) and in Virtual Telephony Provider (VTP) level.

It means that the above set of Black Lists is replicated in the MOM by VTPs. One instance is controlled by the OTP operator and there are replications controlled by the corresponding VTP operators.

If the served user belongs to a VTP domain then both the OTP and the VTP Back List sets are applied on the communication. Served users not belonging to any VTP domain are OTP users, so that only the OTP Back Lists are applied on their communication.



Entries in the OCB Black List are matched with the “Request-URI” value of the INVITE message whereas in some use cases they are matched with the “Refer-To URI” values (see ref [27]).

Entries in the ICB Black List are matched with the “P-Asserted-Identity” value of the INVITE message whereas in some use cases they are matched with the “Referred-By URI” value (see ref [27]).

Entries in the CDIV Black List are matched with the “Request-URI” value of the diverted INVITE message.

The entry in the list can be part of a URI, it is not required to add a complete SIP URI or TEL URI to the list (i.e. the comparison is a sub-string match).

For example:

The following entries:

.se
+468
bob@example.com
spam

-will match the following URIs:

sven@operator.se
+468112233
bob@example.com
12345@good-spam.com

Matching is case sensitive and US-ASCII is used as the character set.

2.16.1.8 Operator Anonymous Communication Rejection

The operator can enable/disable the anonymous communication on node-level.

If the Operator ACR is enabled on the global level then all anonymous communications are rejected by the MTAS.

Please note, that Originating Identification Presentation (OIP) has an effect on Operator ACR. When OIP is disabled and ACR is set for a particular user all of his/her incoming communication will be rejected as the identity information will be removed from the SIP messages.

2.16.1.9 Operator Barring Programs, Operator Permitted Programs, and Operator Diversion Barring Programs

Operator Barring Programs, Operator Permitted Programs, and Operator Diversion Barring Programs are based on defining a set of Operator Barring Categories, which supplement the Barring Categories defined for user-accessible Barring Programs. An Operator Barring Category is defined by a list of number ranges to be included, a list of number ranges to be excluded, and a list of domains to be included.



A user's Operator Barring Program is defined in the operator part of the user data, as an alternative to the Operator Permitted Program. It contains a list of category names; the list can consist of Operator Barring Category Names, user Barring Category Names, and ten special categories: Local, Non Local, Allow Local, L_National, L_International, L_IntraLata, L_IntraLataToll, L_InterLata, L_NanpZone1, and L_Nanp.

When the user attempts to make an outgoing call, the Request URI is checked against the set of categories specified for the user, and the call is barred if the address is included in one of the categories.

A user's Operator Permitted Program is defined in the operator part of the user data as an alternative to the Operator Barring Program. It contains a list of category names; the list can consist of Operator Barring Category Names, user Barring Category Names, and nine special categories: Local, Non Local, L_National, L_International, L_IntraLata, L_IntraLataToll, L_InterLata, L_NanpZone1, and L_Nanp.

When the user attempts to make an outgoing call, the Request URI is checked against the set of categories specified for the user, and the call is barred if the address is not included in one of the categories.

A user's Operator Diversion Barring Program is defined in the operator part of the user data. It contains a list of category names; the list can consist of Operator Barring Category Names, user Barring Category Names, and ten special categories Local, Non Local, Allow Local, L_National, L_International, L_IntraLata, L_IntraLataToll, L_InterLata, L_NanpZone1, and L_Nanp.

When the user attempts to divert an incoming call, the number is checked against the set of categories specified for the user, and the call is barred if the address is included in one of the categories.

When a communication is barred by an Operator Barring Program or Operator Diversion Barring Program, a final SIP response 603 (Decline) is sent to the caller as an indication. In addition to the response an audio or video announcement can be played. Playing the announcements is configurable by the operator. The operator can choose to play an announcement associated with the (Operator, user, or special) Barring Category, or an announcement associated with Outgoing Communication Barring, or no announcement.

When a communication is barred by an Operator Permitted Program, a final SIP response 603 (Decline) is sent to the caller as an indication. In addition to the response an audio or video announcement can be played. Playing the announcements is configurable by the operator. The operator can choose to play an announcement associated with Outgoing Communication Barring, or no announcement.

The audio or video announcement is sent in-band using the early media session, if and only if the caller's SDP offer supports it.



2.16.1.10 Operator White Lists

It is possible for the operator to define one global ICB White List and one global OCB White List.

Both the ICB White List and the OCB White List are defined by 3 lists of strings. (Numbers Included, Numbers Excluded, Domains Included).

- The 'Numbers Included' list specifies the leftmost parts of the normalized numbers that are not to be barred by the global white list. This list is front-substring matched with Tel URIs and SIP URIs containing a telephone number.
- The 'Numbers Excluded' list specifies the leftmost parts of the normalized numbers that are to be barred by the global white list. This list is front-substring matched with Tel URIs and SIP URIs containing a telephone number.
- The 'Domains Included' list specifies the set of domains that are not to be barred by the global white List. This list is only compared with SIP URIs that does not contain a telephone number. Each entry in the list is a string which represents the host part of a URI. If the first character in the string is a '*' this is treated as a wildcard character and a rightmost match of the domain name from the URI will be performed with the rest of the characters in the string. If the first character in the string is not a '*' then the domain name from the URI must exactly match the included string.

Matching is case sensitive and US-ASCII is used as the character set.

If the Wholesale (see chapter 2.53) is enabled and Wholesale domains are defined on the node then the White Lists can be configured in Operating Telephony Provider (OTP) and in Virtual Telephony Provider (VTP) level.

It means that the above set of White Lists is replicated in the MOM by VTPs. One instance is controlled by the OTP operator and there are replications controlled by the corresponding VTP operators.

If the served user belongs to a VTP domain then both the OTP and the VTP White List sets are applied on the communication. Served users not belonging to any VTP domain are OTP users, so that only the OTP White Lists are applied on their communication.

2.16.1.11 Location Based OCB

It is possible for the operator to bar a call going to a specific destination depending on the mobile location the call was made from.

The operator can add areas from where the call is not allowed on the node-level.

The PANI-header (P-Access-Network-Info) is used to get the caller's location. Operator can add exceptions to allow subset of exempted number for outgoing communication.

Location related additional information can be played depending on the configuration of announcement



2.16.1.12 Play Announcement

When a communication is barred a final SIP response 603 (Decline) is sent to the caller as an indication. In addition to the response an audio or video announcement can be played. Playing the announcements is configurable by the operator. The operator can choose to play an announcement associated with the Barring Category, an announcement associated with Outgoing Communication Barring or to play no announcement.

The audio or video announcement is sent in-band using the early media session or established session, if and only if the caller's SDP offer supports it. The way of playing announcements is configurable by the operator.

2.16.1.13 Precedence order of barring rules

For session initiation and ICB, the following series of checks are performed (in precedence order):

1. ICB White List
2. Global Anonymous Communication Barring
3. ICB Black List
4. VTP ICB White List
5. VTP Global Anonymous Communication Barring
6. VTP ICB Black List
7. Operator incoming barring rules for subscriber
8. Subscriber incoming barring rules

The checks 4, 5 and 6 are performed only if the served user belongs to a VTP domain – see chapter 2.53 for more details about the wholesale.

For session initiation and OCB, the following series of checks are performed (in precedence order):

1. CDIV Black List
2. VTP CDIV Black List
3. OCB White List
4. Nodal Dial Plan
5. OCB Black List
6. OTP-controlled per-VTP Dial Plan
7. VTP OCB White List
8. VTP-controlled per-VTP Dial Plan



9. VTP OCB Black List
10. OTP Location Based OCB
11. VTP Location Based OCB
12. Operator Barring Program OR Operator Permitted Program
13. Operator Diversion Barring Program
14. Operator outgoing barring rules for subscriber
15. Barring Program
16. Subscriber outgoing barring rules

The checks 2, 6, 7, 8 and 9 are performed only if the served user belongs to a VTP domain – see chapter 2.53 for more details about the wholesale.

For session update and ICB, the following series of checks are performed (in precedence order):

1. Operator incoming barring rules for subscriber
2. Subscriber incoming barring rules

For session update and OCB, the following series of checks are performed (in precedence order):

1. Operator outgoing barring rules for subscriber
2. Barring Program

2.16.2 Service Interaction

CB interacts with following services:

2.16.2.1 Identity Presentation

Originating Identification Presentation (OIP) has an effect on ACR. When OIP is disabled and ACR is set for a particular user all of his/her incoming communication will be rejected as the identity information will be removed from the SIP messages.

If the end user has the OIP service active including OIR Override then this takes precedence over the ACR service. If the served user has the OIR Override service, no incoming request shall be treated as anonymous.

Communication barring is not invoked if CLIR interworking is active and the user calls with hidden identity.



2.16.2.2 Communication Diversion

When communication is diverted OCB of the diverted-by user is invoked on the transit MMTel AS.

2.16.2.3 Ad-hoc Conference

In interaction with the ad-hoc conferencing service the headers Refer-To and Referred-By are inspected.

2.16.2.4 Multi Subscriber Number (MSN)

When the MSN feature is enabled the CB conditions may also be specified per MSN identity.

2.16.3 Configuration

Examples of node-level configuration parameters related to the Communication Barring service are:

- Enable/disable all Communication Barring.
- Configure Operator White Lists and Operator Black Lists
- Configure Audio and Video Announcements separately for ICB, ACR, OCB, each Operator Barring Category, each user Barring Category, and each special Barring Category.
- Configure Barring Categories and Barring Programs

2.16.4 Performance Management

- Performance parameters for CB are updated.

2.16.5 Fault Management

For information on the alarm, refer alarm OPI [61].

2.17 Media Policy

2.17.1 Description

Operator can block media based on the media policy rules in the subscriber profile.

When MMTel AS receives a SIP message, containing an SDP offer or answer, it checks media policy rules in subscriber profile. If a matching rule is found with “allow” action set to “false” then ports of all streams where media type is same as in the rule is set to 0.

Rule matching continues until all rules are checked. After that, MMTel AS continues the call setup.



PM counters are stepped to indicate the number of blocked media streams. Usage of media policy function is indicated in charging messages.

2.17.2 Example Call Flows

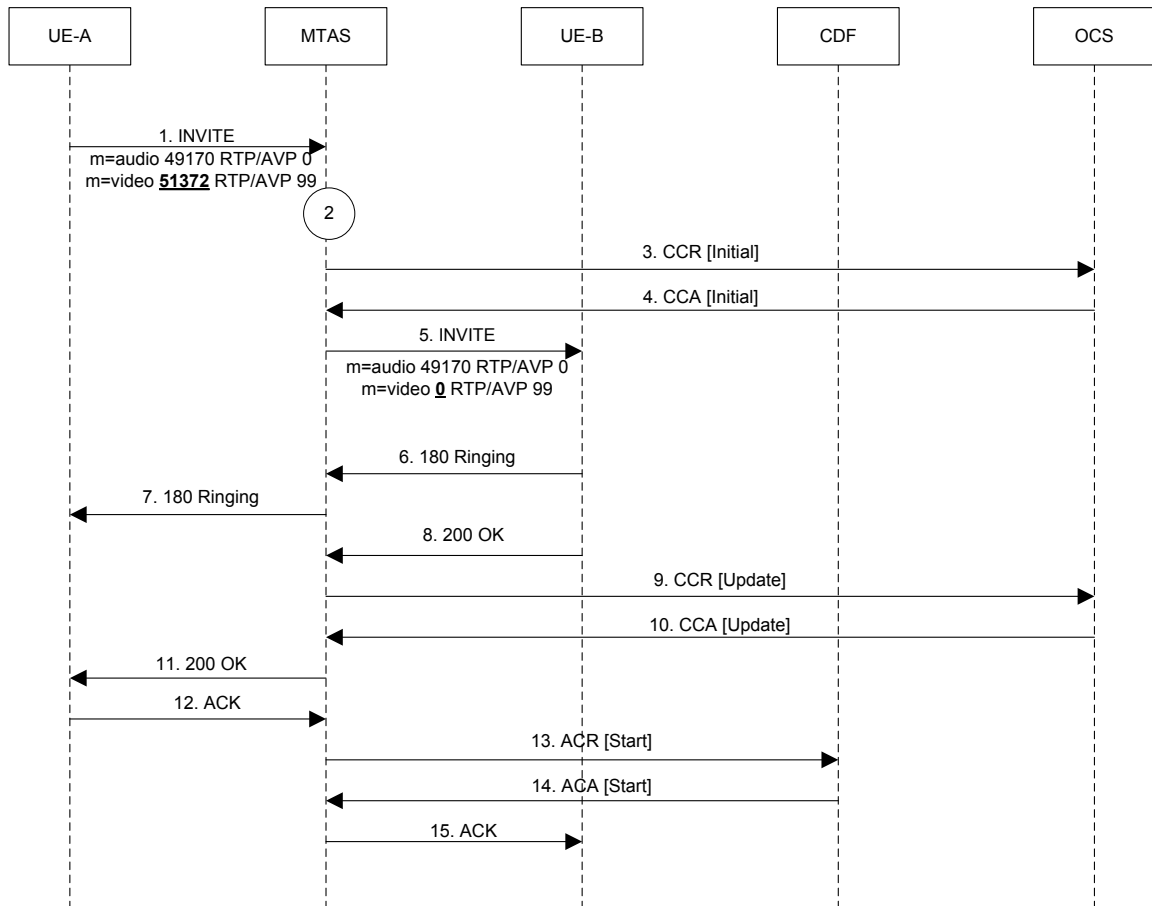


Figure 40 MMTel AS blocks a stream in an SDP offer in originating INVITE

2.17.3 Charging

Invoked Media Policy service information and blocked port is reported to the charging server.

2.17.4 Provisioning

MTAS enables the operator to configure the service on user-level through the CAI3G interface. Possible settings are:

- Create rules using XML structure
- Create rules using Binary Data Model



2.17.5 Example of Media Policy rule

```
<mmt-data:operator-configuration>
  <mmt-data:operator-service-data>
    <mmt-op:operator-media-policy activated="true">
      <cp:ruleset>
        <cp:rule id="VideoProhibited">
          <cp:conditions>
            <ss:media>video</ss:media>
          </cp:conditions>
          <cp:actions>
            <ss:allow>false</ss:allow>
          </cp:actions>
        </cp:rule>
      </cp:ruleset>
    </mmt-op:operator-media-policy>
  </mmt-data:operator-service-data>
</mmt-data:operator-configuration>
```

2.17.6 Performance Management

Performance counters related to the Media Policy service are:

- Number of streams blocked for remote user
- Number of streams blocked for local user

2.18 Malicious Communication Identification

MMTel AS conveys information about the malicious communication using Diameter messages to either the Communication Details Server (CDS) or as an ACR file stored on the local file system

MMTel AS uses information received in the initial INVITE request for identification purposes and does not support the generation of SIP INFO requests to obtain identification information that is not present in the INVITE request as this procedure is handled by the MGCF before forwarding the INVITE to MMTel AS. For temporary mode MCID, MMTel AS does not support the use of a SIP Re-INVITE request to register a communication as malicious.

The MCID service is executed at the terminating and originating MMTel AS node. The MCID service has two ways of reporting the MCID information:

- Report the MCID information to the Communication Details Server via the CommDetails interface.
- Store the MCID information locally on the file system as ACR files which can later be transferred to a remote host by the use of the File Transfer.

Only one of the report methods can be enabled at the same time.

The MCID service has two modes, permanent and temporary.



- Permanent mode. The details of all incoming communications to, or from the served user are reported.
- Temporary mode. The details of the last two incoming communications to the served user are stored. The served user can use an SSC to invoke the MCID service. Based on the SSC used the communication details held in either the “latest store” or the “previous store” are reported. Provisioning of temporary mode is available at terminating MTAS.

In case of forking, the communication details are reported for the first dialog to respond. The communication details are updated on receipt of the first 180 Ringing response to the INVITE request or the first 200 OK response in case no 180 Ringing responses have been received.



2.18.1 Example Call Flow

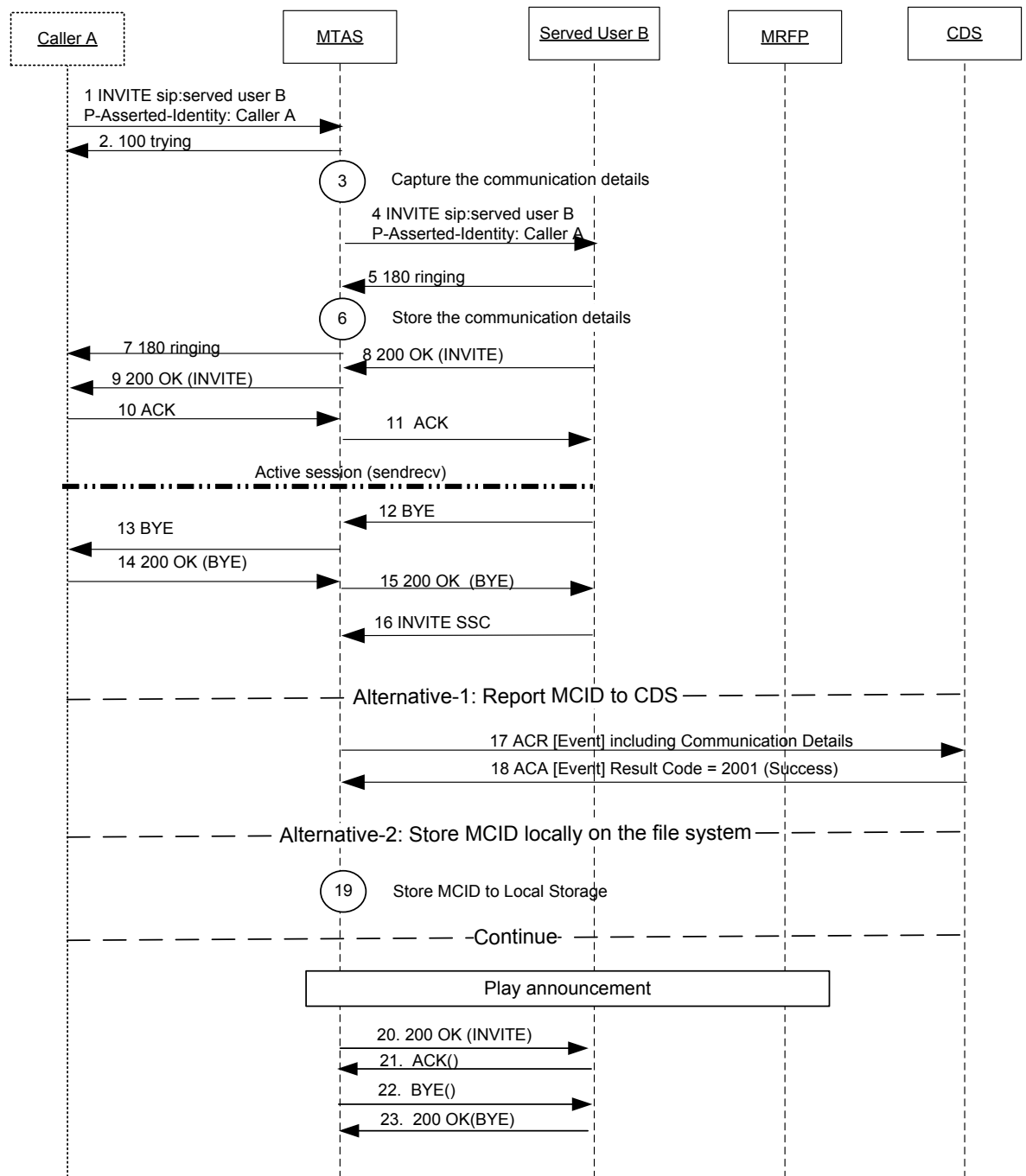


Figure 41 Temporary mode user registers communication as malicious

2.18.2 Service Interaction

MCID Permanent Mode has no service interaction.



MCID Temporary Mode has service interaction with

- Communication Completion Busy Subscriber (CCBS)
 - If a 180 Ringing response is received before the busy condition is identified, the stored communication details are updated.
- Communication Deflection
 - If Communication Deflection occurs during ringing, the stored communication details are updated.
- Communication Forwarding on Busy
 - If the user is alerted, the stored communication details are updated.
- Communication Forwarding No Reply
 - Communication details are updated on receipt of the first 180 Ringing response from the served user.
- Communication Waiting
 - Communication details are updated on receipt of the first 180 Ringing response from the served user.

2.18.3 Configuration

- Activation (active/ disable)
- Configuration of SSC
- Configuration of CDS peer information.
- Configuration of the Local Storage information

2.18.4 Performance Management

- Performance parameters for SSC are updated.
- Performance parameters for charging keyed by the CDS host or realm are updated.
- Performance parameters for the Local Storage information.

2.19 Malicious Communication Rejection

2.19.1 Description

Malicious Communication Rejection (MCR) is a combination of DBL and MCID, i.e. a service to store incoming identities in a black list to avoid unwanted communication as described in section 2.18 and to report details of the malicious communication to the Communication Details Server (CDS) as described in section 2.23.



2.19.2 Service Interaction

The service interaction is according to CB and SSC.

2.19.3 Configuration

- Configuration of SSCs.
- Configuration of CDS peer information.
- Configuration of CB.

2.19.4 Performance Management

- Performance parameters for MCR SSCs are updated.
- Performance parameters for charging keyed by the CDS host or realm are updated.
- Performance parameters for Barring are updated

2.20 Dynamic Black List

2.20.1 Description

Dynamic Black List (DBL) enables an end-user to store incoming identities in a black list to avoid unwanted communication. DBL is also applicable in case of anonymous communication.

Dynamic Black List (DBL) is a variant of the Incoming Communication Barring (ICB) feature, intended to work as an intelligent and selective filter for unwanted communication.

An end-user having the service active is able to store an unwanted incoming identity in the black list by dialing a feature code. The end-user can do this either during the conversation session, or by using the last or second last incoming number.

By supporting barring of selective anonymous calls, the DBL feature is very flexible in comparison with Anonymous Call Rejection (ACR) which bars all anonymous calls. When an anonymous call is stored in the black list, the number remains invisible to the end-user, thereby guaranteeing the anonymity of the calling user.

The black listed anonymous identity is stored in the operator part of the user provisioned data in order to be hidden from the user. There is another entry in the user part of the provisioning data displaying the time stamp for the black list entry in order to facilitate user removal of the entry in the DBL.

Future communication from user with its identity in the DBL will be subject to triggering of ICB.



2.20.2 Example Call Flow

The trigger for DBL has the call flow according to the SSC feature, see section 3.1.

Future communication from user with its identity in the DBL will be handled according to ICB with 603 Decline.

2.20.3 Service Interaction

The service interaction is according to CB and SSC.

2.20.4 Configuration

- Configuration of SSCs.
- Configuration of CB.

2.20.5 Performance Management

- Performance parameters for DBL SSCs are updated.
- Performance parameters for Barring are updated

2.21 Address Policing

2.21.1 Description

MTAS supports a number of address policy checks, as described below.

2.21.1.1 Long Distance Mobile Number Policing

This check is intended for use in numbering plans where mobile numbers are associated with geographic locations, and mobile numbers are distinguished from fixed numbers by the number of digits.

The check ensures that a call to a non-local mobile number is dialed using the National Dialing Prefix.

2.21.1.1.1 Configuration

- On/Off for this sub-service
- National Dialing Prefix
- Length of a normalized fixed number, including the leading “+”
- Length of an unnormalized mobile number



2.21.1.1.2 Performance management

The following performance counters are provided by MTAS:

- Number of calls rejected by Long Distance Mobile Number Policing

2.21.2 Example call flow

The call flow is common to all the sub-services of Address Policing.

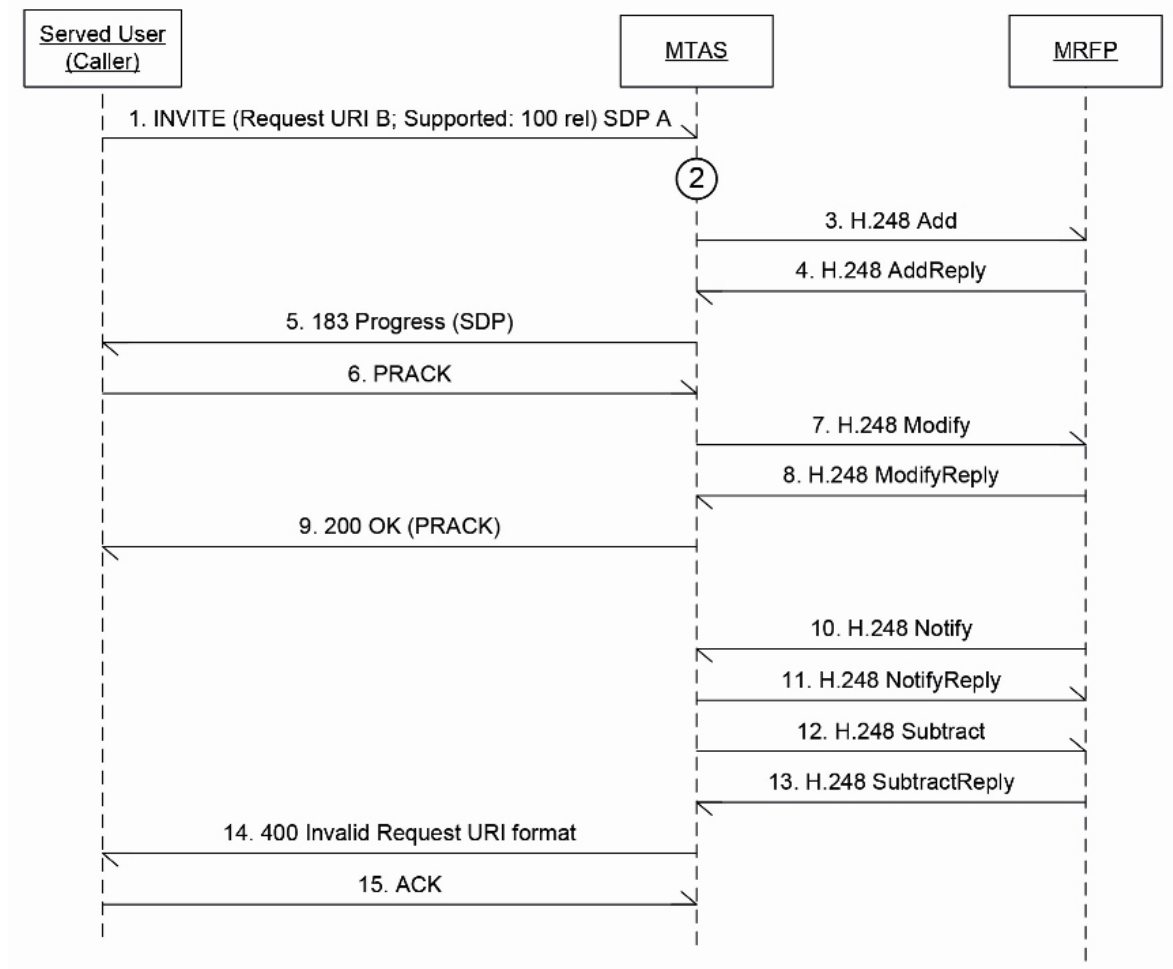


Figure 42 – Reject Malformed Address

- 54 The Served User (caller), A, sends an INVITE towards B. In this case the INVITE indicates support for reliable provisional responses and contains SDP in the body.
- 55 MTAS receives the INVITE, performs the format checks on Request URI and finds that the Request URI does not comply. MTAS finds that the announcement associated with the format check combined with the SDP received in the INVITE means that an announcement should be played.
- 56 MTAS sends an H.248 Add containing the SDP received in the INVITE to the MRFP to reserve resources.



- 57 The MRFP responds with an SDP answer identifying the allocated resources.
- 58 MTAS sends a 183 Progress reliable provisional response towards A, containing the SDP answer received from the MRFP, and a P-Early-Media header.
- 59 A sends a PRACK in response to the 183 Progress.
- 60 MTAS sends an H.248 Modify to the MRFP to get it to start playing the announcement.
- 61 The MRFP responds.
- 62 MTAS sends a 200 OK (PRACK) to A.
- 63 The MRFP sends an H.248 Notify to MTAS to indicate that the announcement has finished.
- 64 MTAS sends an H.248 NotifyReply to the MRFP.
- 65 MTAS sends an H.248 Subtract to the MRFP to release the resources.
- 66 The MRFP responds.
- 67 MTAS sends a 400 final response, with a Reason Phrase describing the problem.
- 68 A sends an Acknowledgement to the Final Response.

2.21.3 Configuration

- On/Off for all sub-services

2.22 Abbreviated Dialing

2.22.1 Description

The Abbreviated Dialing service enables a subscriber to call an assigned, stored number by dialing a short digit sequence.

It shall be noted that for Abbreviated Dialing, it is the end-user that is responsible for the definition and configuration of the abbreviated numbers within his/hers own subscriber profile, whereas in feature Short Number Dialing it is the operator that maintains the number plan.

Abbreviated Dialing is handled on the originating MTAS node.

The number called has an Abbreviated Number in a range between the numbers (0)0 and 99 (100 numbers in total) and this number is analyzed and changed to a stored number during an INVITE or REFER. Abbreviated numbers between 0-9 and 00-99 are supported.

A range of valid abbreviated numbers to use is defined with CM attributes with the possible values between (0)0 and 99.



The range for abbreviated number from 0-9 and 00-09 are stored to the same number range 0-9.

A list of stored New Destination (ND) numbers is stored in the subscriber data. The ND stored can optionally be prefixed with an ad-hoc identity presentation SSC or carrier select code in the format [<SSC>][<CSC>]<ND>.

Provisioning is done on the CAI3G interfaces. Update of the subscriber data is done on the CAI3G and on the Ut interface.

2.22.2 Example Call Flow

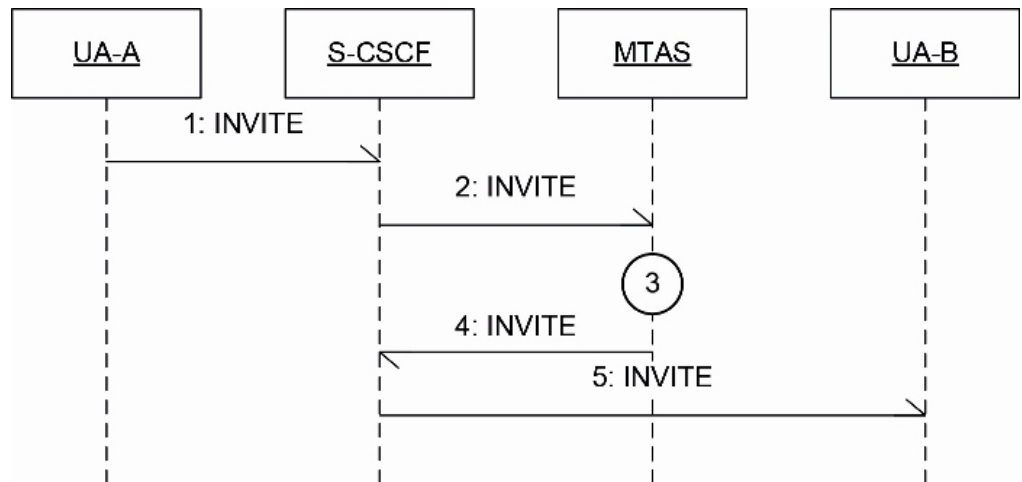


Figure 43 - Handle INVITE

1. The User-A sends an INVITE to establish a call.
2. The S-CSCF sends the INVITE to the MTAS.
3. MTAS analyzes the Request URI according to the command syntax. If it is a match then MTAS searches the subscriber data list of the abbreviated numbers for the user. If the abbreviated number exists then the MTAS updates the Request URI header with an Abbreviated Number to have a Request URI with a number and type (sip: or tel:) fetched from the user data list for Abbreviated Dialing.
4. The MTAS processes the INVITE normally and forwards it to the S-CSCF.
5. The INVITE is routed by the S-CSCF towards the User-B on the terminating side.

2.22.3 Service Interaction

The Abbreviated Dialing function has the following interactions with other services provided by MTAS.



2.22.3.1 Charging

The use of Abbreviated Dialing is reported in charging messages generated during the setup of an MMTel session. Expanded number saved in charging record for called party id. The requested-party-id holds the number input to MTAS.

2.22.3.2 Supplementary Service Codes

SSC code for ad-hoc identity presentation is supported in front of a normalizable number stored in the subscriber data.

2.22.3.3 Carrier Select

The carrier select code is supported in front of a normalizable number stored in the subscriber data

2.22.3.4 OCB

OCB service will act on the expanded number.

2.22.4 Configuration

- Activation (active/ disable)
- Configuration of SSC

2.22.5 Performance Management

- Count of the total number of successful invocation of the Abbreviated Dialing function.

2.23 Short Number Dialing

2.23.1 Description

The Short Number Dialing (SND) service provides the members in a group with the ability to call each other by means of short numbers common to all members of the group.

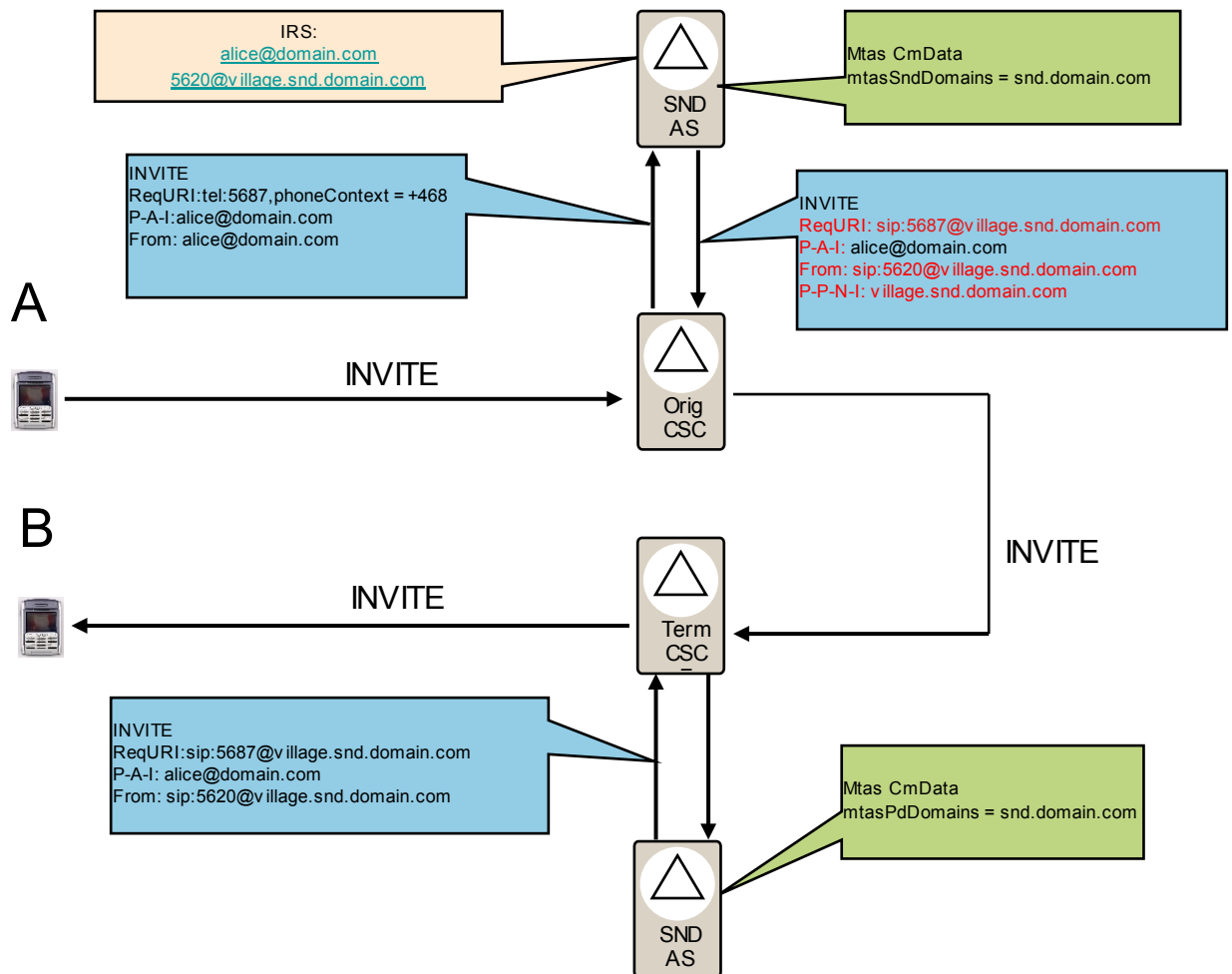


Figure 44 – SND service overview

The SND service impacts the routing, i.e. to route a call from the caller to the called party, based on the SND identity in the Request-URI. It also impacts the presentation and charging.

2.23.2 Example Call Flow

2.23.2.1 SND call where the Req URI is a SND identity

The scenario covers the case, where a SND user calls another SND user by its SND identity.

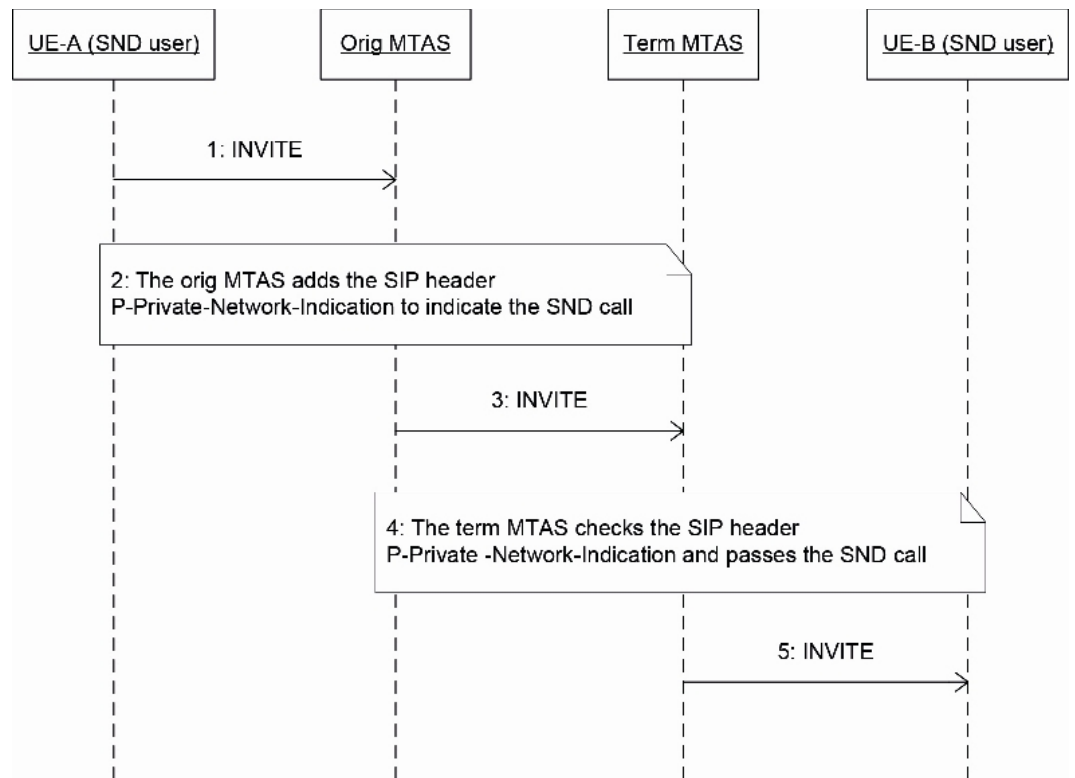


Figure 45 – SND call, SND user calls SND user by using SIP URI

69 INVITE is received at the originating MTAS.

```

INVITE sip:5622@village1.snd.domain.com
P-A-I: sip:alice@domain.com
From: sip:alice@domain.com
  
```

70 The originating MTAS checks whether it is a SND call.
This is a SND call, so the From header is changed to UE-A's SND identity.
The P-P-N-I header is added containing the SND domain.
The charging AVPs for SND are added.

71 INVITE is sent by the originating MTAS.

```

INVITE sip:5622@village1.snd.domain.com
P-A-I: sip:alice@domain.com
From: sip:5621@village1.snd.domain.com
P-P-N-I: village1.snd.domain.com
  
```

72 The terminating MTAS checks whether it is a SND call.
This is a SND call, so the P-P-N-I header is removed before sending the INVITE to the UE-B.
INVITE is sent by the terminating MTAS.

```

INVITE sip:5622@village1.snd.domain.com
P-A-I: sip:alice@domain.com
From: sip:5621@village1.snd.domain.com
  
```



2.23.2.2 SND call where the Req URI contains SND number

The scenario covers the case, where a SND user calls another SND user by its SND number.

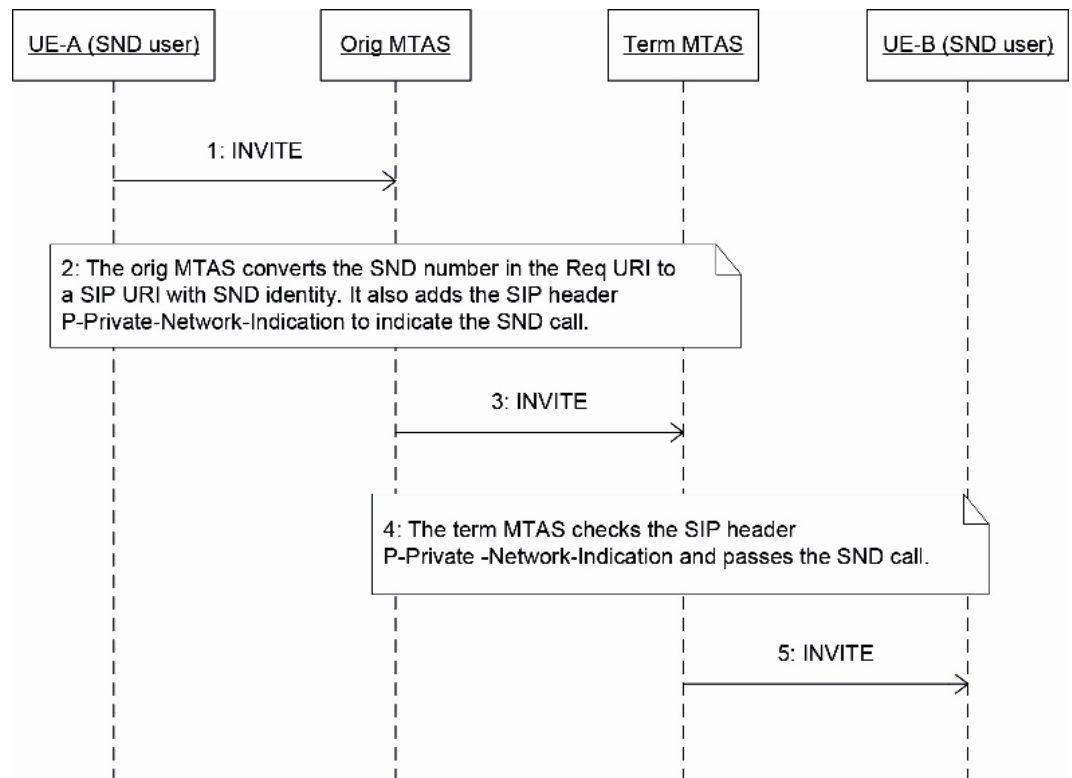


Figure 46 – SND call, SND user calls SND user by using tel URI

73 INVITE is received at the originating MTAS.

```

INVITE tel:5622; phone-context=+468
P-A-I: sip:alice@domain.com
From: sip:alice@domain.com
  
```

74 The originating MTAS checks whether it is a SND call.
This is a SND call, so the Req URI is converted to SIP URI with the UE-A's SND domain. The From header is changed to UE-A's SND identity. The P-P-N-I header is added containing the SND domain. The charging AVPs for SND are added.

75 INVITE is sent by the originating MTAS.

```

INVITE sip:5622@village1.snd.domain.com
P-A-I: sip:alice@domain.com
From: sip:5621@village1.snd.domain.com
P-P-N-I: village1.snd.domain.com
  
```

76 The terminating MTAS checks whether it is a SND call.
This is a SND call, so the P-P-N-I header is removed before sending the INVITE to the UE-B.

77 INVITE is sent by the terminating MTAS.



INVITE sip:5622@village1.snd.domain.com
P-A-I: sip:alice@domain.com
From: sip:5621@village1.snd.domain.com

2.23.3 Service Interaction

2.23.3.1 Communication Diversion

The SND service adds the P-P-N-I header to the outgoing INVITE if it is a SND case

Handling of SND in the CDIV target at provisioning

2.23.3.2 Conference

Handling the P-P-N-I header in REFER and outgoing INVITE if it is a SND to SND case

2.23.3.3 Flexible Communication Distribution

The SND service adds the P-P-N-I header to the outgoing INVITE if it is a SND case

Handling of SND in the FCD target at provisioning

2.23.3.4 Communication Barring

ICB is done towards the From header for barring of the short number as the P-A-I does not contain the short number

2.23.3.5 Carrier Select

CS does not apply to short numbers

If CIC or CSC prefix is used it will still not trigger CS

The SND dialing plan must not clash with the CIC/CSC prefix

2.23.4 Configuration

Short Numbers are configured as valid Public User Identities to the alias IRS of the user.



2.24 Number Normalization

2.24.1 Description

SIP URI and tel URI [28] can be presented to the MTAS via a number of interfaces, these being:

- Ut-interface
- ISC interface from the S-CSCF
- CAI3G interface

The MTAS Number Normalization feature is capable of normalizing SIP and tel URI by using contexts from:

- Request URI context, or
- P-Asserted-Identity context, or
- <userIdentity> element on CAI3G, or
- Ut input with an X-3GPP-Asserted-Identity.

MTAS is capable of inserting a user=phone parameter if parameter is missing in the SIP URI.

MTAS is also capable of detecting a “name” in SIP URI and not acting upon it, returning it unchanged.

After number normalization MTAS updates the input-URI with the normalized number or Null if normalization is not possible.

The number normalization output may be a:

- Global E.164[36] format number in SIP or tel format
- Nationally Significant Number (NSN) with a Country Code (CC) context or domain name context
- Operator Service Number (OSN) with a Country Code (CC) or domain name context of the OSN operator

2.24.2 Configuration

MTAS provides the operator a number of node-level configuration parameters which affects the behavior of the Number Normalization. The parameters are managed by LDAP. Examples are:

- Profile name in string for which the Number Normalization data will be defined.
By default the profile should be a country name.
- Warning text string that defines the nature of number Normalization failure



- String of the rules context consisting of digits or domain name.

2.24.3 Performance Management

The following performance counters are provided by MTAS to evaluate the usage and quality of service:

- Number of input string which is not recognized as a valid tel or SIP URI or is deemed to be a malformed tel or SIP URI
- Number of failed attempts to normalize a NSN, OSN, or normal tel or SIP URI number because no rules contexts could be found to match against the contexts provided

2.25 Dialed Number Mapping

2.25.1 Description

The Dialed Number Mapping (DNM) service main functionality is to support Local format and National format dialing for subscribers within the North American Numbering Plan (NANP) and to support toll-free and short-code dialing services for the subscribers within a generic Numbering Plan. The DNM service can be configured to modify the Phone-Context of the dialed number.

The DNM function serves both fixed and mobile terminals.

The term “location” will have different meaning for fixed and mobile terminals. For mobile terminals, the “location” represents the geographical location of the terminal at the time of placing the call, derived from the P-Access Network Information SIP header. For fixed terminals, the “location” represents an area code (from the main identity set in Implicit Registration Set (IRS)).

The Local format dialing service enables a subscriber to call local number by dialing only the local format part, which is Local formats length in NANP. The DNM service, when enabled, will complete the dialed number by adding the country code for North America, followed by subscriber's Numbering Plan Areas (NPA) prefix for obtaining global E.164 number. The Local format dial attempt from outside home area network will be rejected by the DNM service. The Local format dialing service can be restricted to certain areas within the home network, allowing this type of dialing method for some areas and rejecting it for others.

The National format dialing service enables a subscriber to call a national number by dialing the NPA and Local number. The DNM service, when enabled will determine call type of the call for National format dialed number. Call type means what that call is classified as, for example international, long-distance, local etc. Based on the policy configuration, DNM service can allow or reject the call. In both cases it is also possible to play an announcement for the calling user before the call is rejected or let to continue. Location related additional information can be played depending on the configuration of announcement. The National format processing can be enabled or disabled for Local format dialed calls.



The toll-free service enables a subscriber to call a number (for example: 1-800-xxxxxxx or 310-xxxx) free of charge. The DNM service, when enabled, will add a prefix in front of dialed number, which will ensure further routing in the network. The added prefix is dependent on the location where the subscriber makes the call attempt.

The short-code dialing service enables a subscriber to call an assigned short number (short-code). The DNM service, when enabled, will translate the short-code into the actual URI dependent on the location where the subscriber makes a call attempt. The short-code can start with any number including “#” and “*” characters. The actual URI can be either tel URI, embedded tel URI, or SIP URI.

The DNM Phone-Context modification service modifies the Phone-Context in the request-URI dependent on the location where the subscriber makes a call attempt. Then the modified Phone-Context can be used by other services (e.g. Number Normalization service). The phone-context modification service can be used together with one of the local, national, short-code or toll-free services.

2.25.2 Example call flow

There are five main call flows for the Dialed Number Mapping service:

- Processing Local format Number Call Flow
- Processing Short-Code Call Flow
- Processing National format Number Call Flow
- Processing Toll-Free Number Call Flow
- Processing phone-context modification Call Flow

When processing a Local format number from mobile device, the Dialed Number Mapping (DNM) service uses the value in the P-Access-Network-Info (PANI) and P-Asserted-Identity (PAI) headers enclosed in the received SIP INVITE to determine whether the caller is dialing from the home area network or not. If the caller is dialing from the home area network, the call attempt will proceed after +1NPA prefix is added into the phone number in the Request-URI header.

When processing a Short-code number, the DNM service uses the value in the P-Access-Network-Info (PANI), for Mobile devices, and P-Asserted-Identity (PAI), for Fixed devices, enclosed in the received SIP INVITE to obtain the location information of served user as the caller. The DNM service translates the short-code into the actual URI dependent on the location where the served user makes a call attempt.



When processing a National format number, the DNM service uses the value in the P-Access-Network-Info (PANI) for Mobile devices, and P-Asserted-Identity (PAI), for Fixed devices, enclosed in the received SIP INVITE to obtain the location information of served user as the caller. The DNM service obtains the localness of the call using the Number Analysis CMCO. Based on the configured policy, an announcement can be played and the call can be accepted or rejected.

When processing a Toll-free number, the DNM service uses the value in the P-Access-Network-Info (PANI) header enclosed in the received SIP INVITE for mobile, and IRS main identity for fixed to obtain the location information of served user as the caller. The DNM service obtains a string value specific for the location where the served user makes a call attempt. This string value is added as a prefix to the dialed number. This makes it possible to achieve location specific routing for the toll-free number.

When processing a phone-context modification, the DNM service uses the value in the P-Access-Network-Info (PANI) header enclosed in the received SIP INVITE for mobile to obtain the location information of served user as the caller. The DNM service obtains a modification rule specific for the location where the served user makes a call attempt. This modification rule is applied to the phone-context of the dialed number. This makes it possible to achieve location specific phone-context modification.

2.25.2.1 Processing Local format Number Call Flow for Mobile

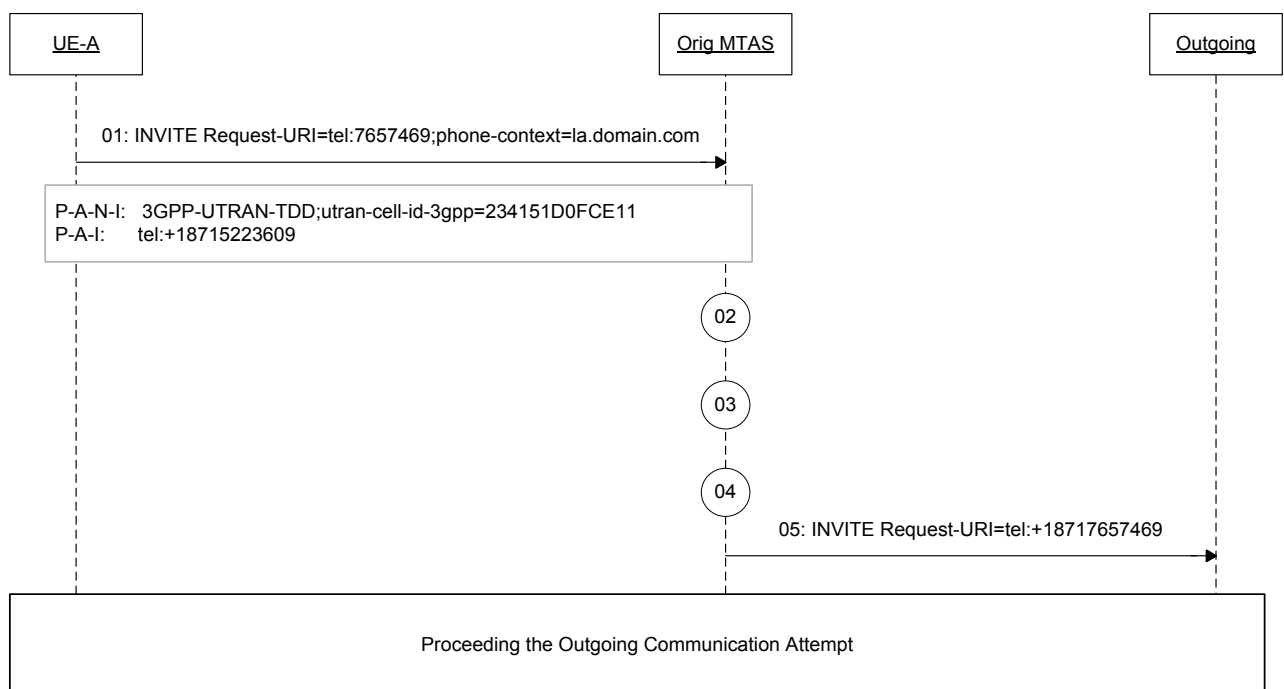


Figure 47 Successful Local format Dialing from Home Area Network for Mobile device



1. UE-A sends an INVITE request towards `tel:7657469;phone-context=la.domain.com`. The value of the P-Access-Network-Info header is `3GPP-UTRAN-TDD;utran-cell-id-3gpp=234151D0FCE11`, and the value of the P-Asserted-Identity header is `tel:+18715223609`.
2. Based on configuration the DNM service decides that Redirect Server is not used and also that it must apply group centric dataset logic. The DNM service using the Number Translation service evaluates the dialed number and determines that this is Local format dialing call attempt.
3. The DNM service performs wildcarding NPA information lookup by using as the input the concatenation of access type and access info type value separated by & character (`3GPP-UTRAN-TDD&utran-cell-id-3gpp`) contained in the P-A-N-I header to find the instance of `MtasCommonDataPaniTranslationProfile`. Then the `mtasCommonDataPaniTranslationRule(10:/(.*)/3GPP-UTRAN-FDD&\1/)` is applied on the access info value (`234151D0FCE11`). The output (`3GPP-UTRAN-TDD&234151D0FCE11`) is used as a key to find the instance of `MtasCommonDataAccNetwTypeAccInfo`. The data lookup obtains the list of NPAs information. In this case, the NPA values are 871, 722, and 781.
4. The DNM service fetches the NPA part of the telephone number contained in the P-Asserted-Identity header. In this example, the NPA indicated in the P-A-I header is 871 (taken from `+1-871-5223609`). Since the NPA is matched with one of the obtained NPA values from the data lookup, the DNM service modifies the telephone number in the Request-URI header by adding `+1871` prefix. The modified Request-URI header now is `tel:+18717657469` (the phone-context in the Request-URI is then removed since now the number is in the global E.164 format).
5. MTAS steps up the counter for successful mobile local number format invocation and then forwards the SIP INVITE.



2.25.2.2 Processing Local format Number Call Flow for Fixed

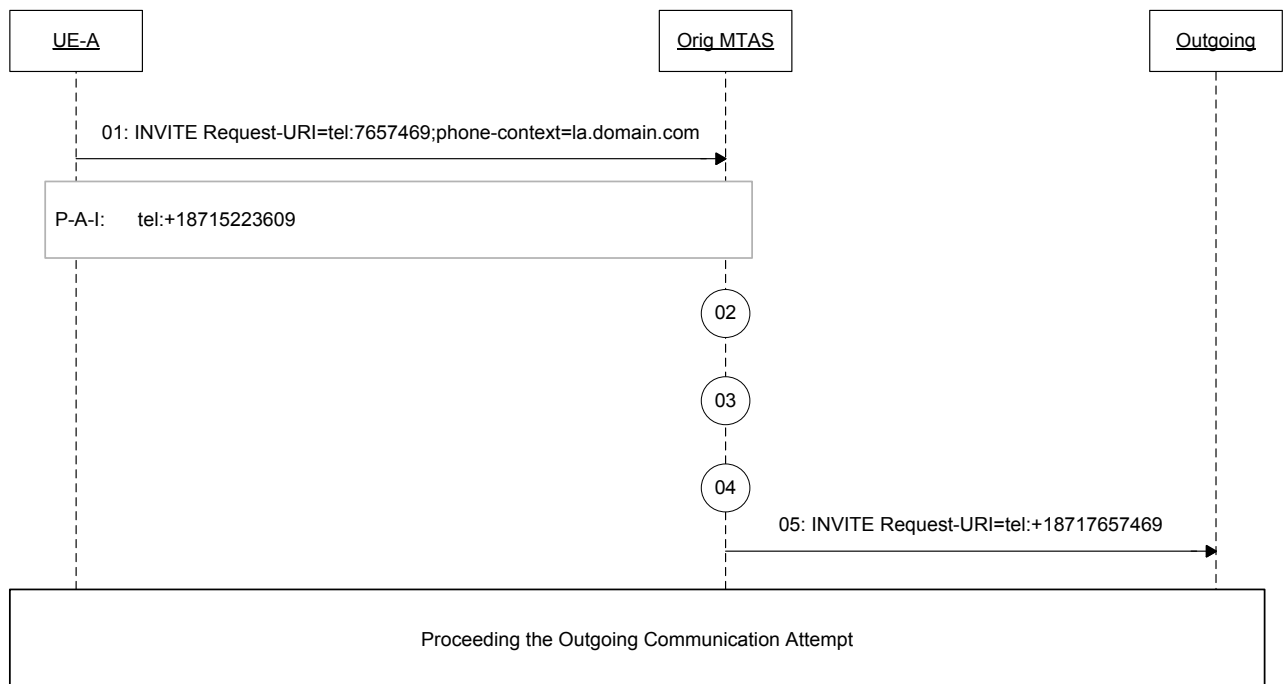


Figure 48 Successful Local format Dialing from Home Area Network for Fixed Device

1. UE-A sends an INVITE request towards `tel:7657469;phone-context=la.domain.com`. The value of the P-Asserted-Identity header is `tel:+18715223609`.
2. The DNM service evaluates the length of the dialed number and determines that this is Local format dialing call attempt from the fixed device.
3. DNM service fetches the Area code from the subscriber data. The area code is 871
4. The DNM service modifies the telephone number in the Request-URI header by adding +1871 prefix. The modified Request-URI header now is `tel:+18717657469` (the phone-context in the Request-URI is then removed since now the number is in the global E.164 format).
5. MTAS steps up the counter for successful fixed local number format invocation and then forwards the SIP INVITE.



2.25.2.3 Processing Short-Code Call Flow for Mobile

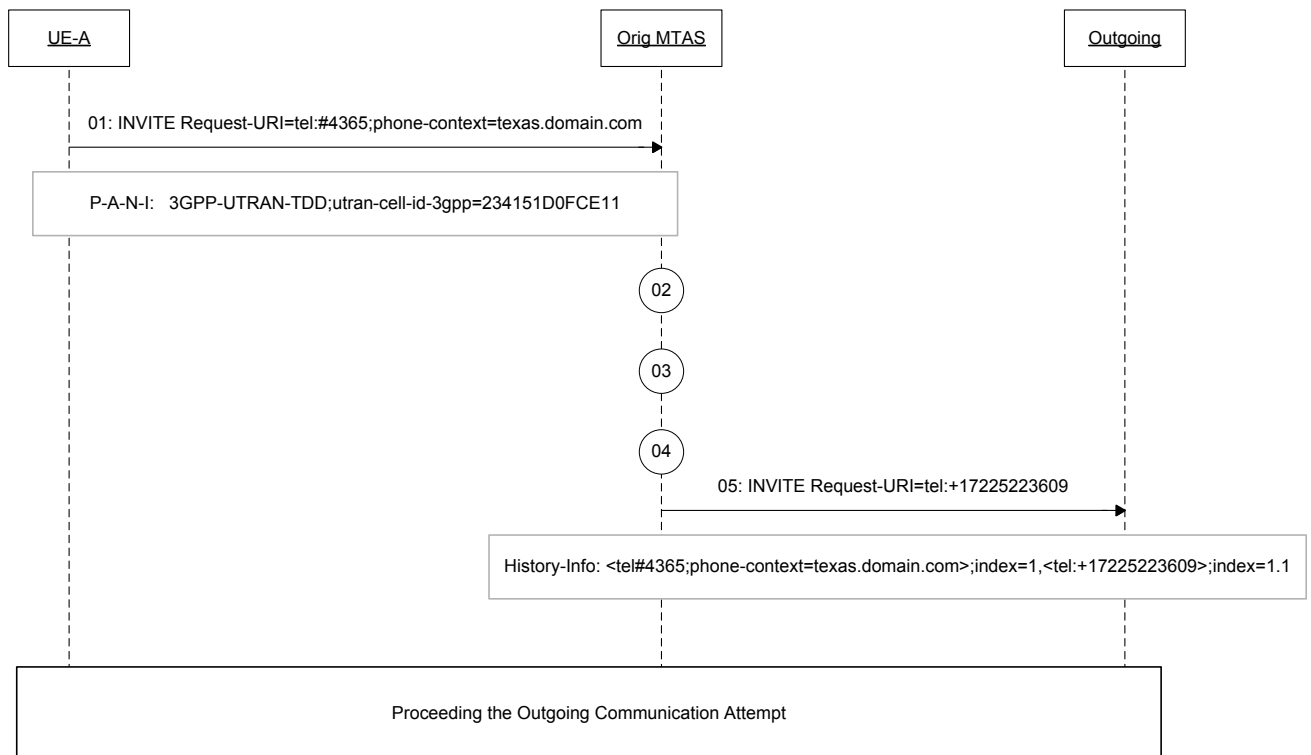


Figure 49 Successful Short-Code Dialing for Mobile

1. UE-A sends an INVITE request towards `tel:#4365;phone-context=texas@domain.com`. The value of the P-Access-Network-Info header is `3GPP-UTRAN-TDD;utran-cell-id-3gpp=234151D0FCE11`.
2. Based on configuration the DNM service decides that Redirect Server is not used and also that it must apply group centric dataset logic. The DNM service using the Number Translation service evaluates the length of the dialed number and determines that this is short-code dialing call attempt.
3. The DNM service performs wildcarding actual URI information lookup. After several lookup processes, the DNM obtains the actual URI value that is `tel:+17225223609`.
4. The DNM service replaces the short-code in the Request-URI header with the URI resulted from the data lookup. The modified Request-URI header now is `tel:+17225223609`. The History-Info header is added with the following value:

`<tel:#4365;phone-context=texas.domain.com>;index=1,<tel:+17225223609>;index=1.1`
5. MTAS steps up the counter for successful mobile short-code invocation and then forwards the SIP INVITE.



2.25.2.4 Processing Short-Code Call Flow for Fixed

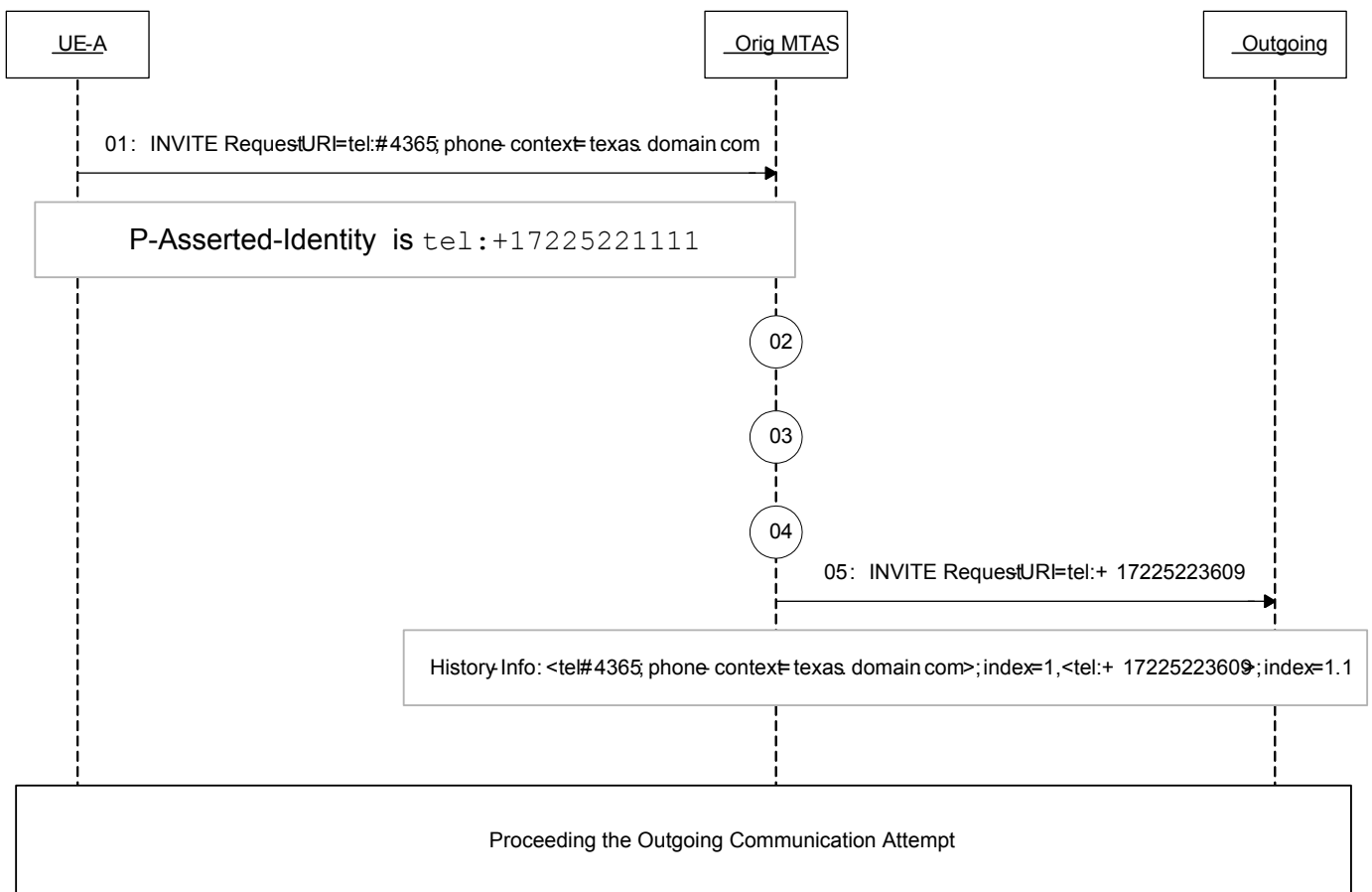


Figure 50 Successful Short-Code Dialing for Fixed

1. UE-A sends an INVITE request towards `tel:#4365;phone-context=texas.domain.com`. The value of the default IMPU is <tel:+17225221111>.
2. Based on the configuration the DNM service decides that Redirect Server is not used. DNM service also decides that it is a call from Fixed Device (due to absence of PANI header). The DNM service using the Number Translation service evaluates the length of the dialed number and determines that this is short-code dialing call attempt.
3. The DNM service performs actual URI information lookup, using the NPA, for an instance of `MtasCommonDataGroupOfCells`, followed by instance lookup of `MtasCommonDataGroupOfCellsAndCode`. After lookup, the DNM obtains the actual URI value that is `tel:+17225223609`.
4. The DNM service replaces the short-code in the Request-URI header with the URI resulted from the data lookup. The modified Request-URI header now is `tel:+17225223609`.
5. MTAS steps up the counter for successful mobile short-code invocation and then forwards the SIP INVITE.



2.25.2.5 Processing National format Number Call Flow for Mobile

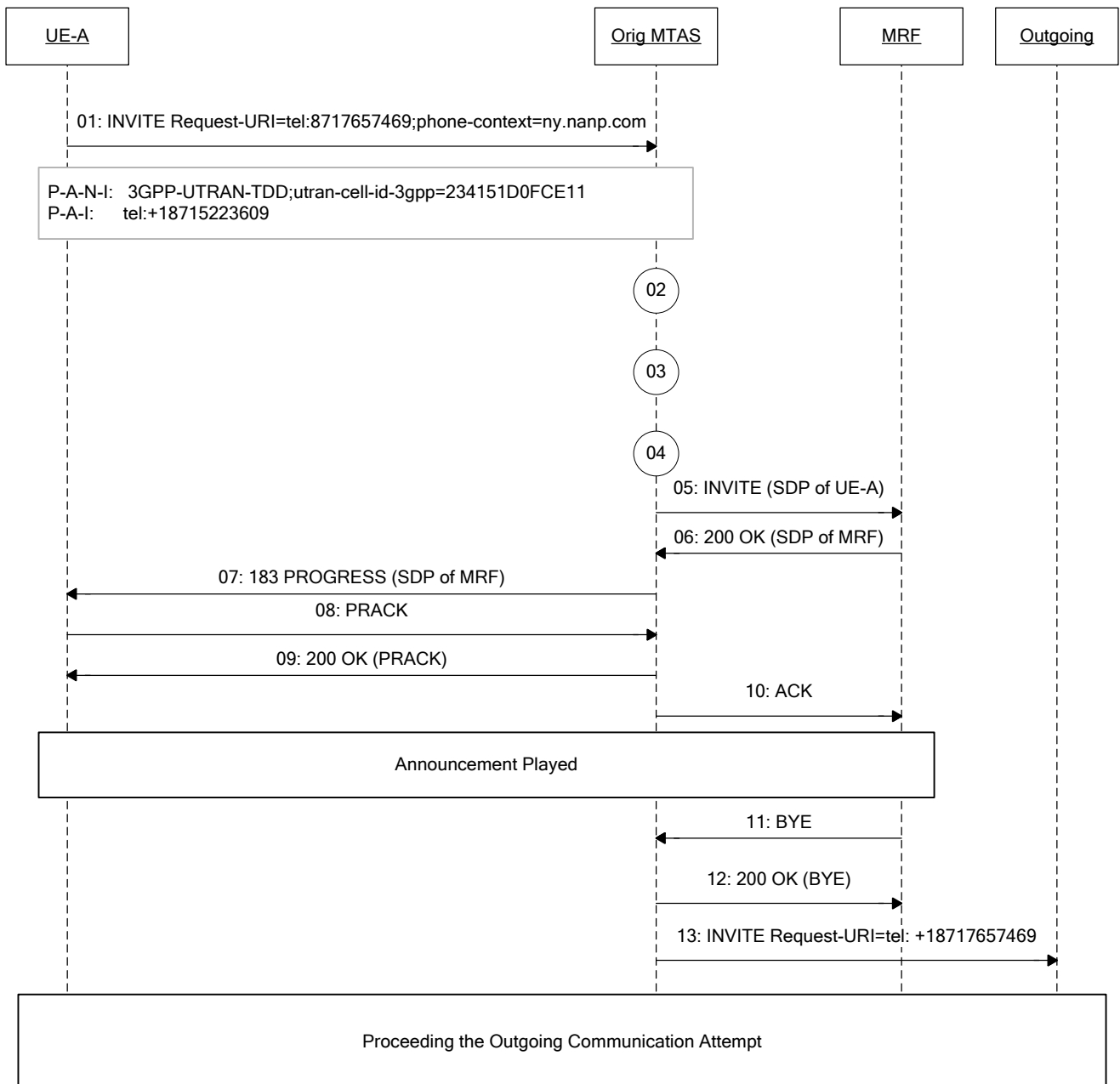


Figure 51: Successful National format dialing for mobile, without Redirect Server

1. UE-A sends an INVITE request towards `tel:8717657469;phone-context=ny.nanp.com`. The value of the P-Access-Network-Info header is `3GPP-UTRAN-TDD;utran-cell-id-3gpp=234151D0FCE11`.



2. Based on configuration the DNM service decides that Redirect Server is not used. The DNM service using the Number Translation service evaluates the length of the dialed number and determines that this is National format dialing call attempt.
3. The DNM service performs wildcarding Mobile Cell information lookup by using the concatenation of access type and access info type value separated by & character (3GPP-UTRAN-TDD&utran-cell-id-3gpp), contained in the P-A-N-I header, as input to find the instance of `MtasCommonDataPaniTranslationProfile`. Then the `mtasCommonDataPaniTranslationRule(10:/(.*)/3GPP-UTRAN-FDD&\1/)` is applied on the access info value (234151D0FCE11). The output (3GPP-UTRAN-TDD&234151D0FCE11) is used as a key to find the instance of `MtasCommonDataAccNetwTypeAccInfo`. The lookup gives the location based Rating Center for determination of call localness.
4. The DNM service modifies the telephone number in the Request-URI header by adding +1 prefix. The modified Request-URI header now is `tel:+18717657469` (the phone-context in the Request-URI is then removed since now the number is in the global E.164 format). DNM service calls the Number Analysis to get the call type. MTAS steps up the `MtasDnmNationalFormatNbrDialOk` counter. Based on configured rules, DNM services determine that it need to play announcement. DNM then prepares to send the announcement informing the caller about the type of call.
5. MTAS sends INVITE request with an SDP offer towards MRF.
6. MRF responds the INVITE with 200 OK containing the SDP of MRF.
7. MTAS sends 183 PROGRESS containing SDP generated by MRF to UE-A.
8. UE-A responds the 183 PROGRESS with PRACK.
9. MTAS responds the PRACK with 200 OK.
10. MTAS sends ACK to MRF ordering it to play announcement informing about dialing out of the home area and then the announcement is played towards UA-E.
11. After the announcement is played, MRF immediately sends BYE request to release the call attempt.
12. MTAS sends 200 OK to MRF.
13. MTAS forwards the INVITE.



2.25.2.6 Processing National format Number Call Flow for Fixed

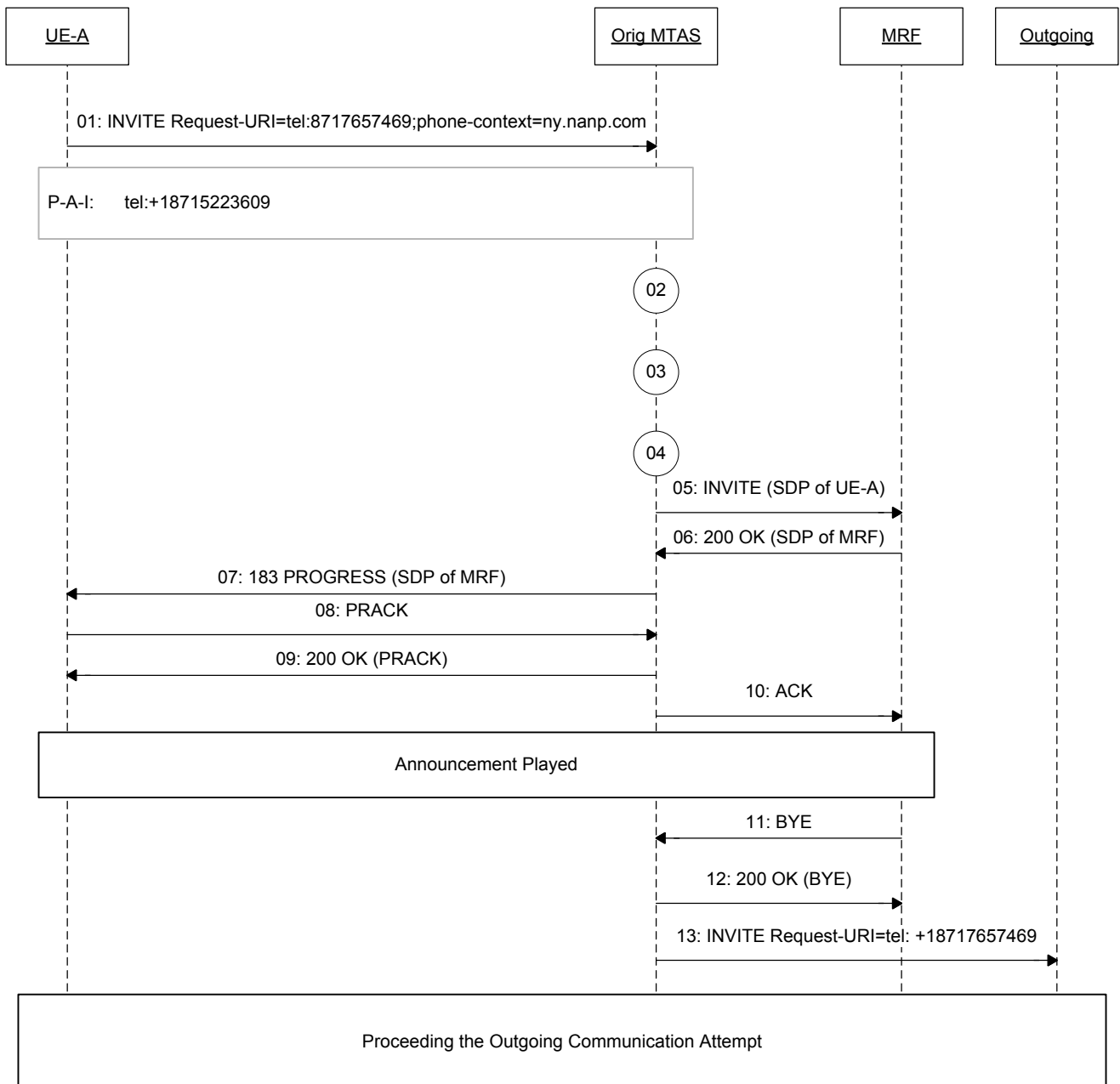


Figure 52: Successful National format dialing for fixed without Redirect Server

1. UE-A sends an INVITE request towards `tel:8717657469;phone-context=ny.nanp.com`.
2. The DNM service using the Number Translation service evaluates the length of the dialed number and determines that this is National format dialing call attempt.



3. The DNM service obtains the IMPU of the calling subscriber, and use that as calling party number.
4. The DNM service modifies the telephone number in the Request-URI header by adding +1 prefix. The modified Request-URI header now is `tel:+18717657469` (the phone-context in the Request-URI is then removed since now the number is in the global E.164 format). DNM service calls the Number Analysis to get the call type. MTAS steps up the `MtasDnmNationalFormatNbrDialOk` counter. Based on configured rules, DNM services determine that it need to play announcement. DNM then prepares to send the announcement informing the caller about the type of call.
5. MTAS sends INVITE request with an SDP offer towards MRF.
6. MRF responds the INVITE with 200 OK containing the SDP of MRF.
7. MTAS sends 183 PROGRESS containing SDP generated by MRF to UE-A.
8. UE-A responds the 183 PROGRESS with PRACK.
9. MTAS responds the PRACK with 200 OK.
10. MTAS sends ACK to MRF ordering it to play announcement informing about dialing out of the home area and then the announcement is played towards UA-E.
11. After the announcement is played, MRF immediately sends BYE request to release the call attempt.
12. MTAS sends 200 OK to MRF.
13. MTAS forwards the INVITE.



2.25.2.7 Processing Toll-Free Number Call Flow for Mobile

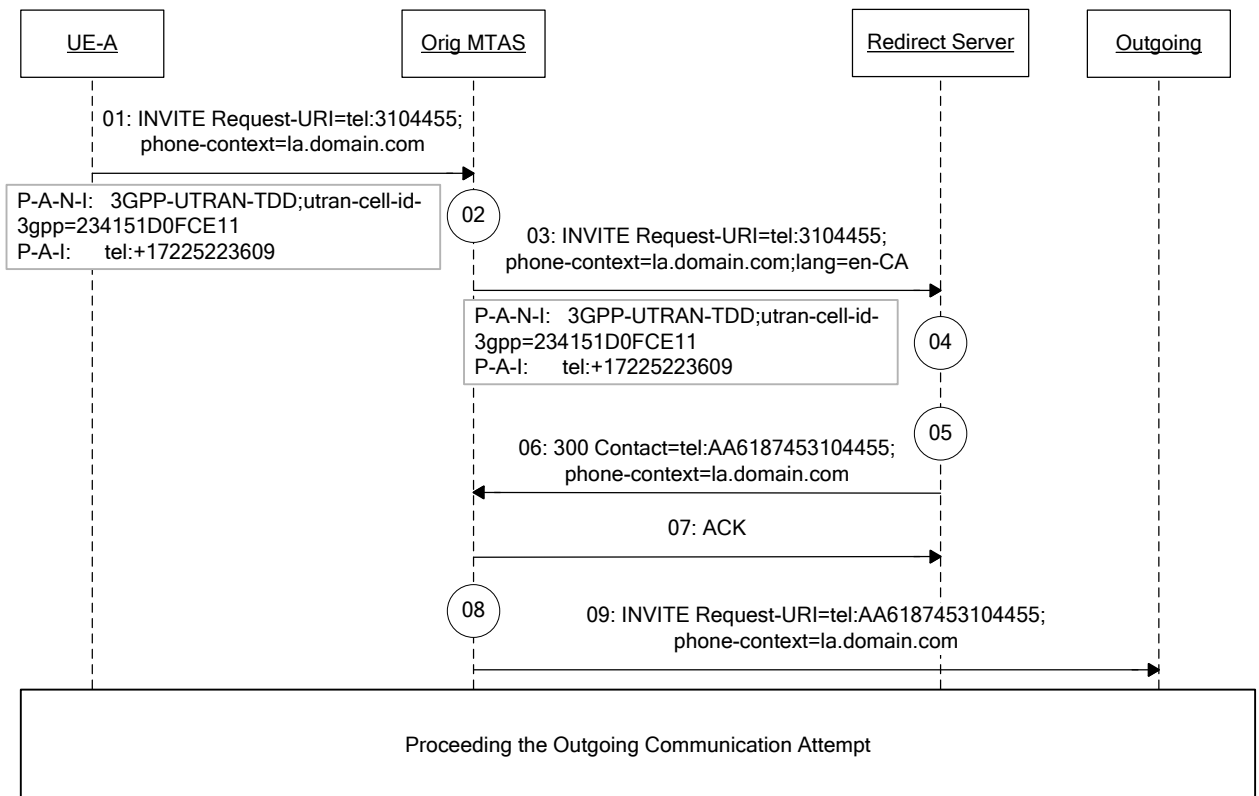


Figure 53: Successful toll-free dialing, group centric dataset with Redirect Server

1. UE-A sends an INVITE request towards `tel:3104455;phone-context=la.domain.com`. The value of the P-Access-Network-Info header is `3GPP-UTRAN-TDD;utran-cell-id-3gpp=234151D0FCE11`.
2. Based on configuration the DNM service decides that Redirect Server should be used.
3. INVITE is forwarded to the Redirect Server.



4. Based on configuration the Redirect Server decides that it must apply group centric dataset logic. The Redirect Server using the Number Translation service evaluates the dialed number and determines that this is toll-free dialing call attempt. The Redirect Server performs wildcarding location specific string lookup by using the concatenation of access type and access info type value separated by & character (3GPP-UTRAN-TDD&utran-cell-id-3gpp) , contained in the P-A-N-I header, as input to find the instance of `MtasCommonDataPaniTranslationProfile`. Then the `mtasCommonDataPaniTranslationRule` (10:/(.*)/3GPP-UTRAN-FDD&\1/) is applied on the access info value (234151D0FCE11). The output (3GPP-UTRAN-TDD&234151D0FCE11) is used as a key to find the instance of `MtasCommonDataAccNetwTypeAccInfo`. The data lookup obtains the "AA618745" string.
5. The Redirect Server then adds this string as a prefix to the dialed number from the Request URI header. The modified URI now is `tel:AA6187453104455;phone-context=la.domain.com`.
6. Redirect Server responds with 300 Multiple Choices with the Contact header containing the modified URI.
7. ACK is sent to Redirect Server.
8. The DNM service replaces the URI in the Request-URI header with the URI resulted from the Contact header from the response. MTAS steps up the `MtasDnmTollFreeFormatNbrDialOk` counter.
9. MTAS forwards the INVITE.



2.25.2.8 Processing Toll-Free Number Call Flow for Fixed

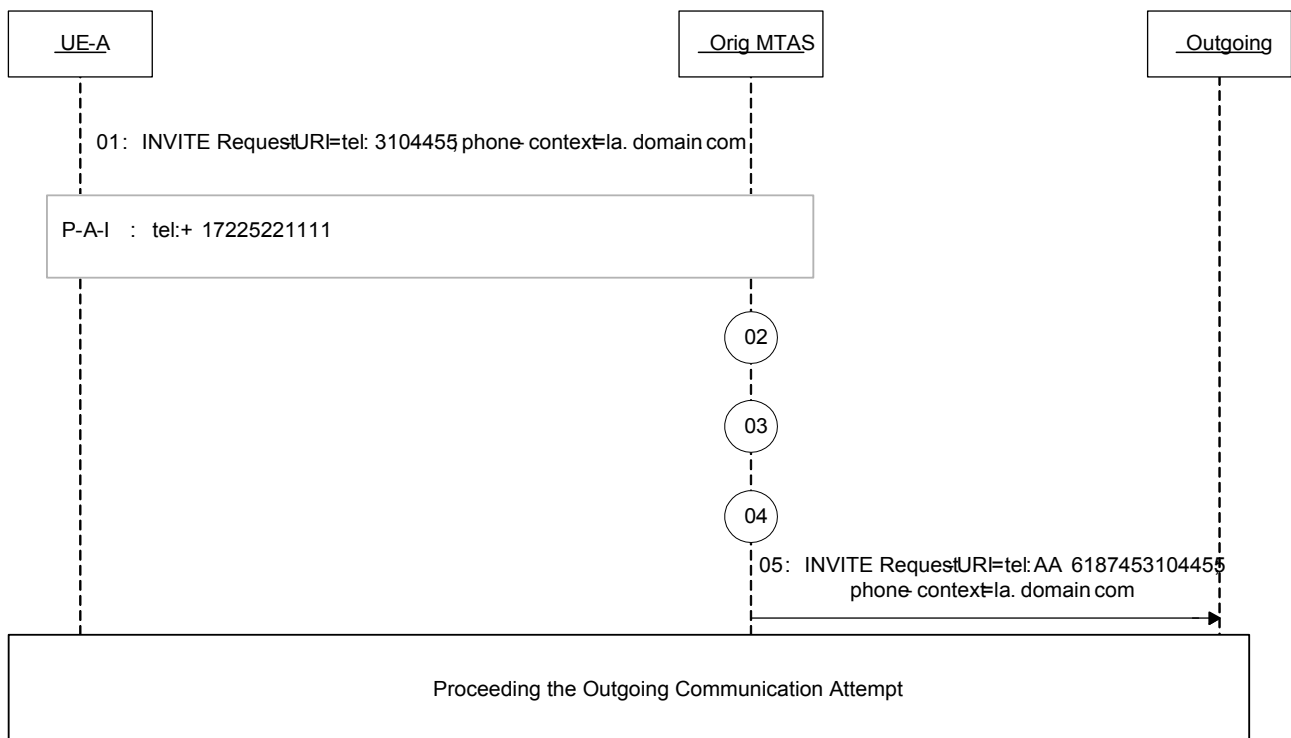


Figure 54: Successful toll-free dialing for fixed, without Redirect Server

1. UE-A sends an INVITE request towards `tel:3104455;phone-context=la.domain.com`. The value of the default IMPU is <tel:+17225221111>.
2. Based on the configuration, the DNM service decides that Redirect Server is not used. The DNM service using the Number Translation service evaluates the dialed number and determines that this is toll-free dialing call attempt.
3. The DNM service performs location specific (Based on Area code). The data lookup obtains the "AA618745" string.
4. The DNM service then adds this string as a prefix to the dialed number. The modified Request-URI header now is `tel:AA6187453104455`. MTAS steps up the `MtasDnmTollFreeFormatNbrDialOk` counter.
5. MTAS forwards the INVITE.



2.25.2.9 Processing Phone-context modification

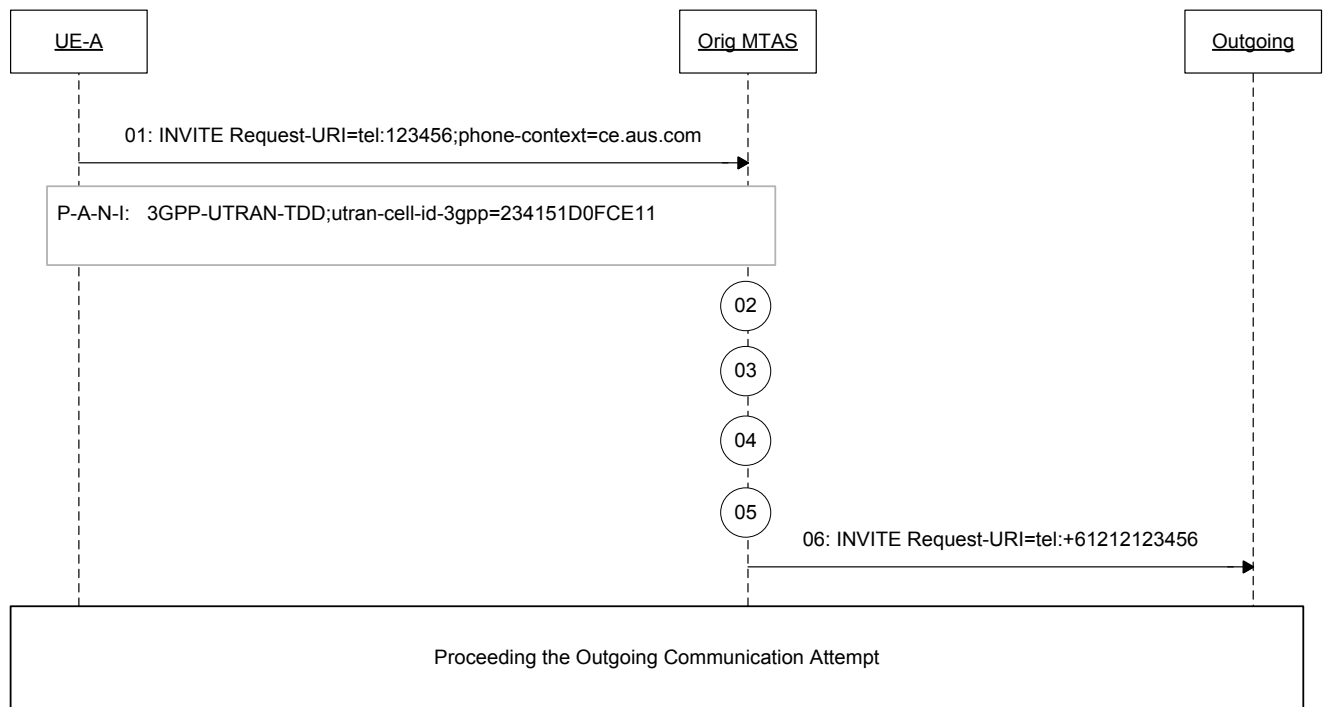


Figure 55: Successful toll-free dialing for fixed, without Redirect Server

1. UE-A sends an INVITE request towards `tel:123456;phone-context=ce.aus.com`. The value of the P-Access-Network-Info header is `3GPP-UTRAN-TDD;utran-cell-id-3gpp=234151D0FCE11`.
2. Number Translation is performed on dialed number using Number Translation rules (`10:/^([0-9]{6})$/\1|$PCM/:TRUE` in this case). The translated number is `"123456$PCM"`
3. The DNM service determines that Phone-Context Modification should be applied and performs wildcarding Modification Rule lookup by using the information contained in P-A-N-I header and `MtasCommonDataPaniTranslationProfiles` table as input to find the instance of `MtasCommonDataAccNetwTypeAccInfo`. The data lookup obtains the list of Phone-Context modification rule:
`PCM&/^(.*)$/location1.\1/`
4. The DNM service modifies the Phone-Context using the modification rule. The modified Phone-Context is `phone-context=location1.ce.aus.com`.
5. DNM determines that none of the DNM scenarios (toll-free, short-code, local number, national number) are applicable but the call is not rejected (because non-NANP) and then the Number Normalization is called to normalize number according the configuration: [tel:+61212123456](#).



6. MTAS steps up the counter for successful phone-context modification and then forwards the SIP INVITE.

2.25.3 Service interactions

2.25.3.1 Short Number Dialing

The DNM service shall not be provisioned if the Short Number Dialing (SND) service is provisioned. Thus, the `mtasDnmAdministrativeState` and `mtasSndAdministrativeState` attributes can not be both unlocked state at the same time.

2.25.3.2 Number Normalization

If the DNM service is unlocked and the license is valid, the Number Normalization service will be invoked two times, before and after DNM service. The first invocation is to perform the “user equal to phone” error correction, to process the Carrier Select Code (CSC), and to determine whether the dialed number is OSN/NSN number. If the result from Number Normalization first invocation is that the dialed number is OSN/NSN, the DNM service will simply forward the SIP INVITE and leave it untouched. However, if the dialed number is neither OSN nor NSN, the DNM service will be triggered and the dialed number length evaluation process is performed. The DNM service may change or modify the phone-context parameter if presented in the Request-URI.

The Request-URI in the outgoing SIP INVITE following the DNM service process must be again normalized. Therefore, the Number Normalization service will be invoked for the second time to normalize the URI.

2.25.3.3 Carrier Select and Carrier Select Rn

The DNM service is invoked before Carrier Select or Carrier Select Rn service. The Request-URI containing Carrier Select Code (CSC) in the SIP INVITE will be processed by Dialed String Analysis (DSA) in the first invocation of Number Normalization service. After the first Number Normalization service invocation, the Request-URI will contain `cic` parameter if the Carrier Select service is provisioned or `rn` parameter if the Carrier Select Rn service is provisioned. The `dai="presub-unkwn-da"` parameter will also be added into the Request-URI. Those parameters will be kept by the DNM service.

When a mobile subscriber from home area network dials a Local format number and provides CSC, the DNM service will complete the number by adding +1NPA prefix and keep the `cic` or `rn`, and `dai="presub-unkwn-da"` parameters in the URI.

When a mobile subscriber dials a short-code and provides CSC, the DNM service will replace the number with the actual URI and keep the `cic` or `rn`, and `dai="presub-unkwn-da"` parameters in the URI.



When a mobile subscriber from home area network dials a National format number and provides CSC, the DNM service will complete the number by adding +1 prefix and keep the `cic` or `rn`, and `dai="presub-unkwn-da"` parameters in the URI.

When a mobile subscriber dials a toll-free number and provides CSC, the DNM service will prefix the number with a location specific prefix and keep the `cic` or `rn`, and `dai="presub-unkwn-da"` parameters in the URI.

Since the Carrier Select/Carrier Select Rn service is only applicable on tel URI or Embedded tel URI, therefore when the dialed short-code is replaced with the actual URI that is in SIP URI by the DNM service the Carrier Select/Carrier Select Rn service will not be triggered.

2.25.3.4 Carrier Pre-select and Carrier Pre-select Rn

The DNM service is invoked before Carrier Pre-select or Carrier Pre-select Rn service.

When processing the Local format number the DNM service will complete the number by adding +1NPA prefix and keep the `cic` or `rn`, and `dai="presub-unkwn-da"` parameters in the URI.

When processing the short-code the DNM service will replace the number with the actual URI and keep the `cic` or `rn`, and `dai="presub-unkwn-da"` parameters in the URI.

When a mobile subscriber from home area network dials a National format number and provides CSC, the DNM service will complete the number by adding +1 prefix and keep the `cic` or `rn`, and `dai="presub-unkwn-da"` parameters in the URI.

When a mobile subscriber dials a toll-free number and provides CSC, the DNM service will prefix the number with a location specific prefix and keep the `cic` or `rn`, and `dai="presub-unkwn-da"` parameters in the URI.

Since the Carrier Pre-select/Carrier Pre-select Rn service only applies on tel URI or Embedded tel URI, therefore when the dialed short-code is replaced with the actual URI that is in SIP URI by the DNM service the Carrier Pre-select/Carrier Pre-select Rn service will not be triggered.

2.25.3.5 Communication Diversion

The DNM service will not be invoked after an incoming communication is diverted by Communication Diversion (CDIV) service. Local format, National format, toll-free and short-code numbers may be used as diversion targets for the CDIV service, but the DNM service is not applicable on the target number.



2.25.3.6 Flexible Communication Distributions

The DNM service will not be invoked after an incoming communication is distributed by Flexible Communication Distributions (FCD) service. Local format, National format, toll-free and short-code numbers may be used as the target of FCD service or as the target of FCD Divert Primary (FCDDP) but the DNM service is not applicable on the target number.

2.25.3.7 Session Transfer to Own Device

The DNM service will not be invoked after Session Transfer to Own Device (STOD) is invoked and distributes the call. Thus, the Local format, National format, toll-free and short-code number should never been used as the target of STOD service.

2.25.3.8 Scheduled Conferencing

The DNM service is not invoked when a dial-out call is made from a Scheduled Conference service.

2.25.3.9 Ad-hoc Conferencing

The DNM service is not invoked when a dial-out call is made from a Scheduled Conference service.

2.25.3.10 OCB

The DNM stores the localness value of the call. This value is used by OCB feature, in case localness barring is required.

2.25.4 Configuration

Node level configuration parameters for service functionalities:

- Activation (active / disable)
- Negative announcements
- Access information
- Short-code related information
- Wildcarding profiles

2.25.5 Performance management

The following performance counters are provided by MTAS to evaluate the usage and quality of service:



- The number of successful modification of the dialed number in local format
- The number of unsuccessful Local format dial attempts distinguished by the reasons
- The number of successful replacement of the dialed number in short-code format
- The number of unsuccessful short-code dial attempts
- The number of successful phone-context modifications
- The number of unsuccessful phone-context modification attempts

2.26 Number translation service

2.26.1 Description

By means of the Number Translation service, a substitution rule-set using regular expressions can be applied to the number part of a Tel URI or a number based SIP URI (i.e. embedded Tel URI) received in:

- Request-URI of INVITE
- Refer-To header of REFER

Number Translation is active when its administrative state is unlocked on originating side and in transit mode on terminating side. Number Translation is executed after Abbreviated Dialing and after handling of Supplementary Service Codes but before Dial Number Mapping, Number Normalization, and Short Number Dialing.

The translation rules can be organized in up to 1000 profiles. Each profile has an associated pattern matching the prefix of the numbers to be translated. Each prefix is shorter than 16 characters. Different profiles have different prefixes. There can be up to 100 substitution rules in a profile.

Finding and executing the applicable translation rules:

- 1 The profile with the longest matching prefix is selected. An empty prefix matches any number. If there is no matching profile the input number is returned unchanged.
- 2 Within the selected profile the first matching translation rule is applied. If the rule is terminal, the result is returned directly. Otherwise, the process continues with looking for the next matching rule. If there are no further matching rules the result is returned.



Before selecting and applying the translation rules, the visual separators and the Carrier Select Codes are removed from the input number.

2.27 Carrier Select

2.27.1 Description

2.27.1.1 Carrier Pre-Select

The Carrier Pre-Select service allows calls from each served user to be handled by a carrier other than the default carrier, depending on the call type (e.g. Local or Non-Local). The operator can specify which sets of contiguous phone numbers constitute a local call for each local area, by setting common service configuration data. The operator can specify which sets of contiguous phone numbers constitute each non-local call type, by setting configuration data.

MTAS determines the type of call relevant for carrier pre-Selection. It checks only for valid call types that have been provisioned in an end-user's XML file. The function returns two parameters: 1) Determined Call Type, 2) Call Type Status. The Call Type Status has three values defined:

DETERMINED:

The Carrier Pre-Select call type has been determined for which there is a provisioned Carrier Code against the user. The Determined Call Type parameter contains the call type.

INDETERMINATE_LOCAL:

This indicates that it cannot be determined whether the call type is Local or Non_local. The Determined Call Type parameter is <empty>.

INDETERMINATE_OTHER:

This indicates that no matching call type has been found for which there is a provisioned Carrier Code and the call type is Non_Local. The Determined Call Type parameter is <empty>.

If the Request URI contains a name rather than a phone number, then the Carrier Pre-Select service is not applicable and will not be applied to the call.

2.27.1.2 Carrier Select

The Carrier Select service allows an end-user to choose which carrier to use for a particular call. Carrier Select overrides Carrier Pre-Select.



The Carrier Select function is responsible for validating carrier identification codes that appear in a cic parameter of the Request URI. The Carrier Select function is not directly responsible for analyzing dialed string Carrier Select Codes, but depends on the Dialed String Analysis function to do this. The Dialed String Analysis function analyzes the dialed digits for Carrier Select Codes and translates them into a cic parameter on the Request URI. The cic parameter is accompanied by a dai parameter (Dial Around Indicator), which is added at the same time as the cic. See [37] and [38] for more detail on CIC and dai parameters.

The main function of the Carrier Select service is to:

- check that a user is allowed to use a Carrier identified either from a dialed Carrier Select Code prefixed to the dialed B number; or a cic parameter added directly to the Request URI
- check the Carrier type and take the appropriate action

The checks performed by the Carrier Select function are described in more detail below.

The MTAS checks that the destination number is allowed to have calls to it and use a carrier other than the default carrier. If not, the MTAS takes one of three actions, depending on the destination number:

- Reject call
- Continue with default carrier routing

MTAS checks whether the call type is local and takes one of 4 actions, depending on network operator policy:

- Release call
- Continue with default carrier routing
- Continue with the chosen carrier

The operator can specify which sets of contiguous phone numbers constitute a local call for each local area, by setting configuration data.



2.27.2 Example Call Flow

2.27.2.1 Carrier Pre-Select

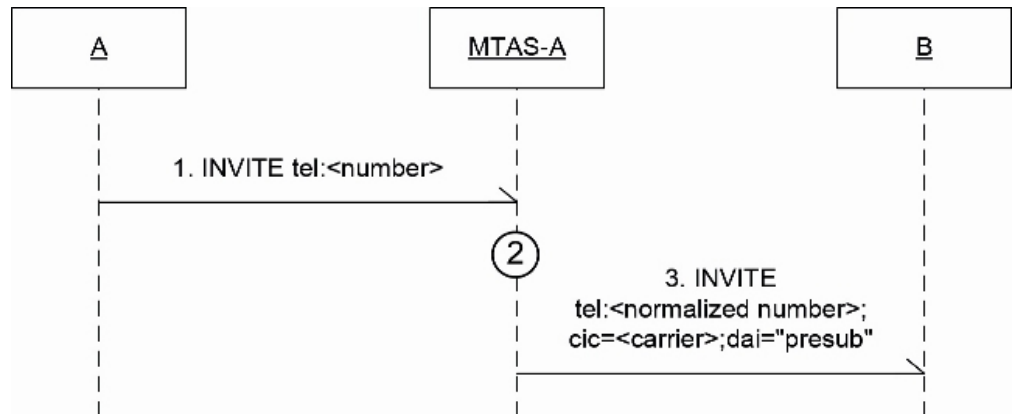


Figure 56 - Carrier Pre-Select

1. An initial INVITE is received. The Request URI contains a tel URI. Note this could equally be a sip URI.
2. The MTAS normalizes the phone number in the Request URI if it is not a global number. MTAS determine Carrier Pre-Select Call Type, from the input parameters - the calling and called party numbers, and the list of call types defined for pre-select from the end-user's XML file. The calling party phone number is obtained from the P-Asserted-Identity SIP header or the first phone number in the IRS in that order.

The function to determine the Carrier Pre-Select call type returns a `call_type_status` of DETERMINED a `call_type` set to one of the following values: "Local", "Non Local", or one of the names configured. MTAS looks up the Carrier code against the determined call type, in the end-user's XML file. This is the Carrier code used for Carrier Pre-selection.

3. The MTAS passes on the INVITE towards the destination, with the Carrier Code associated with the selected carrier in the `cic` parameter on the Request URI. The format of the generated `cic` parameter is dependant on configuration. A `dai` parameter is also added to the Request URI with a value of "presub" which indicates that the CIC contains the caller's presubscribed carrier and the caller did not dial a Carrier Identification Code.



2.27.2.2 Carrier Select

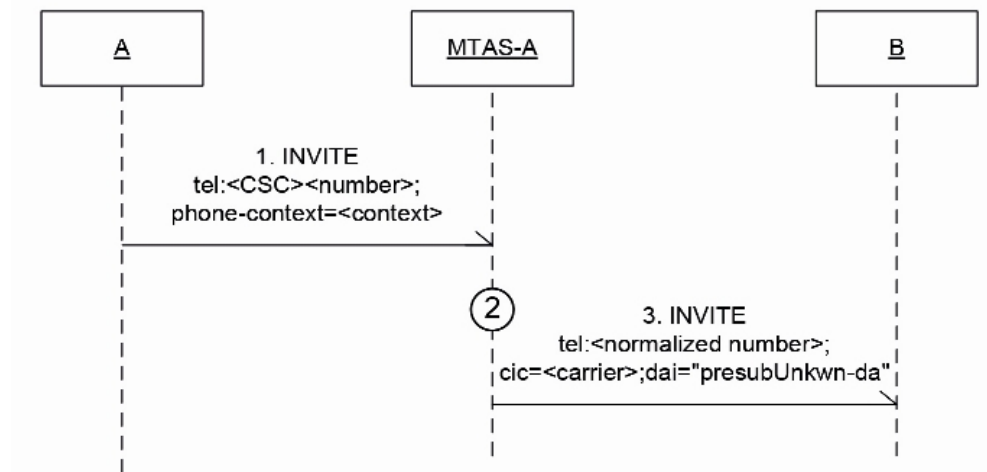


Figure 57 - Carrier Select

- 3 An initial INVITE is received. The Request URI contains a non global number, which includes a Carrier Select Code (CSC).
- 4 Before invoking the Carrier Select function, MTAS invokes the Dialed String Analysis function. Dialed String Analysis acts on the Request URI and translates the dialed Carrier Select Code to a cic parameter. Once CSC translation has been performed, the phone number is passed to the number normalization function to normalize the rest of the phone number. MTAS then checks the call type. In this case, the return value is "Non Local".
- 5 The MTAS passes on the INVITE towards the destination, with the Carrier Code associated with the selected carrier in the cic parameter on the Request URI. A dai parameter is also added, which contains information about how the cic value relates to any pre-selected carrier code. In this scenario, the dai parameter is set to the value "presub-unkwn-da" which indicates that the CIC contains the caller's dialed carrier-identification-code and it is not known if the selected carrier is the caller's presubscribed carrier.

2.27.3 Service Interaction

2.27.3.1 Carrier Select

Carrier Select overrides Carrier Pre-Select.

2.27.3.2 Conference

For conference, Carrier Select will apply to each initiated dial out leg.

2.27.4 Configuration

- Carrier Pre-Select Activation (active/ disable)
- Carrier Select (active/ disable)



- Local Call Type Table, includes the list of leftmost part of normalized global numbers which constitute the local call type
- Carrier Pre-Select Call Type (other type than Local or Non-local), includes the list of leftmost part of global numbers or local numbers which constitute (or exempted from) the call type
- Dialed String Analysis Table, includes the mapping table between Carrier Select Code and CIC

2.28 Identity Presentation

2.28.1 Description

MTAS offers the following identity presentation services based on the 3GPP standard ref [2], [15] and [16]:

- Originating Identification Presentation (OIP)
- Originating Identification Restriction (OIR)
- Terminating Identification Presentation (TIP)
- Terminating Identification Restriction (TIR)

Additionally MTAS offers the following non-standard services:

- Calling Name Identity Presentation (CNIP/OCNIP)
- Flexible Identity Presentation (FIP)

The services enable presentation of the identities of participants in a communication to the other participants and enable a participant to withhold his identity information from the other participants. The restriction may be overridden by the OIP and TIP services for participants which have the override option.

The Identity presentation and restriction is based on the SIP Privacy header information which is defined in RFC 3323 (ref [24]) and RFC 3325 (ref [25]). The values on the Privacy header specify for what other headers privacy is requested. The values that are of interest for the MTAS are:



Table 1 - Privacy header value

Privacy header value	Affected Headers
none	No privacy is required.
id	'Network asserted user identity' privacy requested.(i.e. P-Asserted-Identity, P-Preferred-Identity headers)
user	'Headers added by the user' privacy (i.e. From, Subject, Call-Info, Organization, User-Agent, Reply-To, In-Reply-To, Server, Warning, Referred-By headers).
header	'Headers added by the network' privacy.(i.e. Via, Contact, Record-Route headers)

When MTAS executes OIP and TIP services it will update headers based on the level of privacy required, "header" or "user". The "id" level of privacy is normally handled by the CSCF but in case the served user doesn't subscribe to the OIP/TIP service, MTAS will update also the "id" level headers.

Other values of the Privacy header than the values included in Table 1 are handled transparently.

2.28.1.1 Originating Identity Presentation (OIP)

OIP enables presentation of the originating user's identity to the terminating users. MTAS responsibility is to remove identity information and privacy headers in case that the terminating user does not have the OIP service or if restriction is requested. The OIP service is executed on behalf of the terminating user and in the terminating MTAS.

The main case for OIP is that the terminating user has OIP and then no actions are performed in MTAS.

If the served user does not have OIP enabled then:

- The From header is anonymized.
- The P-Asserted-Identity and P-Preferred-Identity headers and Privacy header values "none", "user", "header" and "id" are removed.

In case restriction is requested with the Privacy header MTAS will always remove identity information from the messages as listed in Table 1. The only exemption is if the served user has OIR Override.

OIR Override

OIR Override makes it possible for the terminating user to see the identity information of the originating user even though the originating user has requested the identity to not be shown. This part of the OIP function is usually only enabled for specific users.

If the served user has OIR Override, Privacy header values "none", "user", "header" and "id" are removed but no other headers are modified.



OIP Global Identity Presentation Restriction list

Global Identity Presentation Restriction is a sub-function of OIP, which allows the operator to specify ranges of numbers which are never presented to end-users.

When configuring the ranges to restrict it is also possible to specify sub-ranges within the restricted number ranges that are not to be restricted.

In the cases an INVITE includes an identity matching the Global Identity Presentation Restriction List OIP behaves as if OIP was not active i.e. anonymize the From header and remove the P-Asserted-Identity for users with OIP active but no OIR Override active.

A match in the Global Identity Presentation Restriction List is not considered to be a match with an anonymous condition by Communication Diversion or Communication Barring.

A match in the Global Identity Presentation Restriction List does not prevent the identity from being included as a target identity by CDIV or FCD.

If there is no match in the Global Identity Presentation Restriction List, the OIP function behaves as normal.

From header de-normalization

If the From header is not anonymized, it can be presented in a locally dialable format. If From header de-normalization is enabled it will be de-normalized if the country code and area code are configured in the user's common data.

2.28.1.2 Originating Identity Restriction (OIR)

OIR enables to restrict the originating user's identity from being presented to the terminating users. MTAS responsibility is to indicate in the signaling that the originating user wants to restrict the presentation of the identity. There are two different modes – temporary mode and permanent mode. The OIR service is executed on behalf of the originating user and in the originating MTAS.

The main case for OIR is that the originating user has OIR active, either in permanent mode or in temporary mode, with the service settings indicating the level of restriction. The originating MTAS will in this case add Privacy header values in accordance with the originating user's service settings.

MTAS supports identity restriction levels:

- 'Restrict asserted identity' corresponds to 'id' + 'user' privacy in Table 1
- 'Restrict all private information' corresponds to 'id' + 'user' + 'header' privacy in Table 1

In permanent mode MTAS updates the Privacy header in each call.



In temporary mode OIR can be set on a per call basis by using Supplementary Service Codes (SSCs). The SSC can be interpreted either as “Restriction” or “No Restriction”,

e.g.: *31*DN# will indicate “Restriction” and

#31*DN# will indicate “No Restriction”.

(DN = Directory Number of the called party)

2.28.1.3 Terminating Identity Presentation (TIP)

TIP enables presentation of the terminating user’s identity to the originating user. The MTAS responsibility is to remove identity information and privacy headers in case that the originating user does not have the TIP service or if restriction is requested. The TIP service is executed on behalf of the originating user and in the originating MTAS.

The main case for TIP is that the originating user has TIP and then no actions are performed in MTAS.

In the alternative case where the originating user does not have TIP the AS will remove identity information from the messages.

If the served user does not have TIP enabled then:

- The P-Asserted-Identity and P-Preferred-Identity headers and Privacy header values “none”, “user”, “header” and “id” are removed.

In case restriction is requested with the Privacy header MTAS will always remove identity information from the messages as listed in Table 1. The only exception is if the served user has TIR Override.

TIR Override

TIR Override makes it possible for the originating user to see the identity information of the terminating user even though the terminating user has requested the identity to not be shown. This part of the TIP function is usually only enabled for specific users.

If the served user has TIR Override, Privacy header values “none”, “user”, “header” and “id” are removed and no other headers are modified.

2.28.1.4 Terminating Identity Restriction (TIR)

TIR enables to restrict the terminating user’s identity from being presented to the originating user. The MTAS responsibility is to indicate in the signaling that the terminating user wants to restrict the presentation of the identity. There are two different modes – temporary mode and permanent mode. The TIR service is executed on behalf of the terminating user and in the terminating MTAS.

The main case for TIR is that the terminating user has TIR active. The terminating MTAS will in that case add the Privacy header field ‘user; id’.



In permanent mode MTAS updates the Privacy header in each call. In temporary mode TIR can be set on a per call basis by using Supplementary Service Codes (SSCs).

2.28.1.5 Calling Name Identity Presentation (CNIP)

This feature adds the caller's display name to the communication signaling.

The feature exists both on the originating MMTel AS (referred as Originating CNIP or OCNIP) and the terminating MMTel AS (CNIP). In the CNIP Service, the terminating MMTel AS retrieves the display name from the external Calling Name Server using SOAP/HTTP protocol.

In the OCNIP service, the callers display name is provisioned in subscriber data.

Display-name is added in the display name portion of the P-Asserted-Identity and the From header. The CNIP/OCNIP service has two modes, "always" and "interrogate-on-unavailability". In "interrogate-on-unavailability" the identity information is only retrieved and added to the SIP headers if the display name is missing in both the P-Asserted-Identity and From headers of the SIP request from the originating user. In "always" mode, the identity information is always retrieved and overwritten if already present in the originating user's request.

The operator has the possibility to configure a default display-name to be used in MTAS when the calling name server cannot provide the identity information due to a timeout, error or when there is no display-name present in the incoming INVITE and the subscriber does not have display name provisioned.

2.28.1.6 Flexible Identity Presentation

The FIP service is replacing the identity of the caller with a predefined identity also called FIP identity, when an outgoing communication is made from the served user. For the called party the call will behave as if it was placed by a user with the FIP identity. Any terminating service dependent on the calling identity (for example Incoming Communication Barring) will work based on the FIP identity. Also charging in the terminating domain will only 'see' the FIP identity. After the call leaves the originating IMS domain the original calling identity is lost and will not be available for other IMS domains or other networks.

It's possible to suppress the FIP identity in case of toll-free call types.

2.28.2 Example Call Flows

2.28.2.1 OIP and OIR

Following diagram shows an example call flow of OIP and OIR.

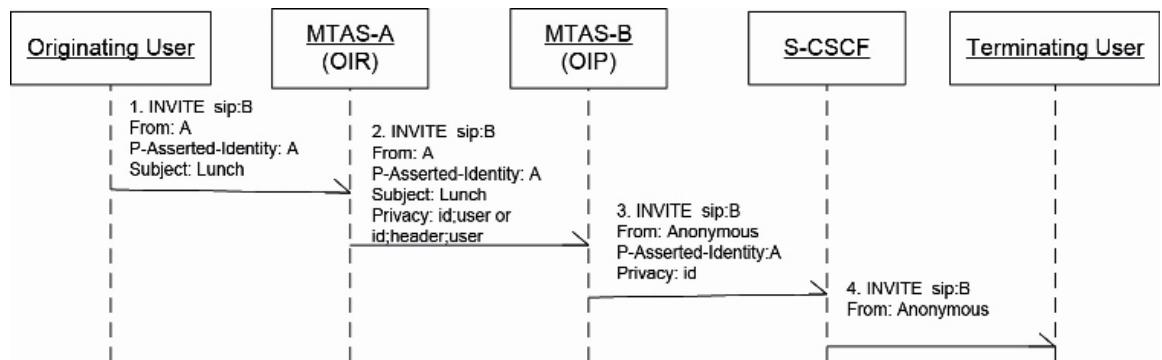


Figure 58 - OIP and OIR, both active, successful case

- 1 User A sends an INVITE request to user B.
- 2 The OIR service is triggered in MTAS-A. If OIR is active then the MTAS-A ensures that a Privacy header is set to “id; user” or “id; header; user” and is part of the request before forwarding it. The level of identity restriction decides which of the two values to use. The “user” value in the Privacy header field is a MTAS implemented network option to ensure “user” level identity restriction.
- 3 The OIP service is triggered in the terminating MTAS. If a Privacy header is present then the MTAS-B ensures that restriction is applied to all relevant headers.
- 4 CSCF handles the “id” level Privacy.

2.28.2.2 CNIP on terminating MMTel AS via SOAP

The served user has both the OIP and CNIP services enabled. The MTAS serving the user will change headers based on the information retrieved from the Calling Name server. The case that will be shown here is the case where a global number identity of the originating user can be extracted from the message headers, the mode is “always” and the Calling Name server successfully answers back with a calling name.

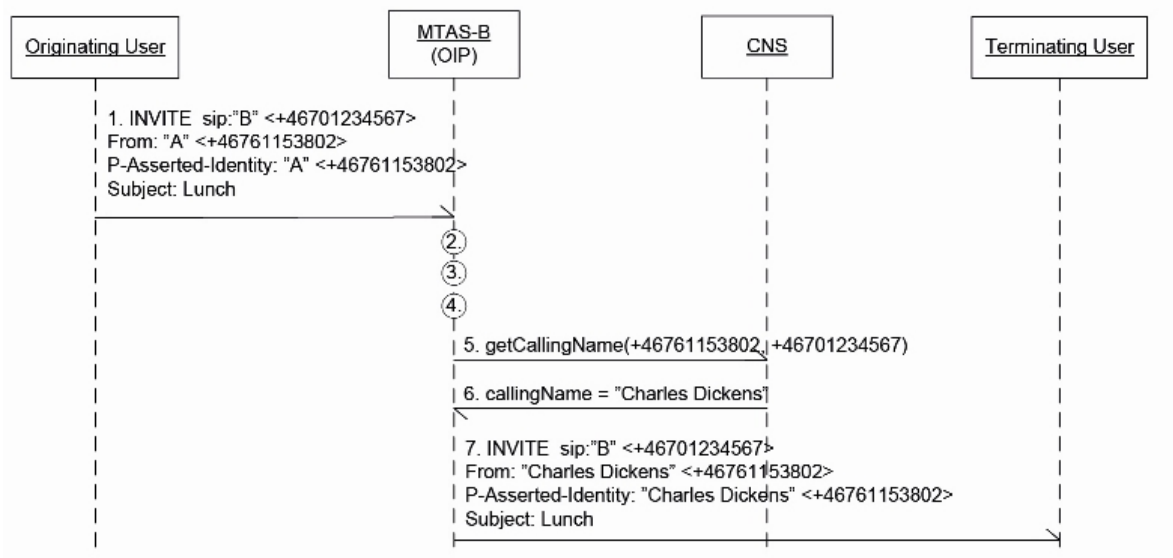


Figure 59 – CNIP in “always” mode



1. User A sends an INVITE request to user B.
2. The OIP service is triggered in the terminating MTAS.
3. If the OIP services does not anonymize the From header or delete the P-Asserted-Identity header the CNIP service is triggered in the terminating MTAS.
4. The global number identity of the originating user is extracted from the P-Asserted-Identity header. If it is missing there, then it is extracted from the From header. If no global number can be obtained from any of those headers then CNIP function will make no further attempt to modify the message.
5. A request is generated toward the Calling Name server containing the following information
 - a. callingPartyId: global number obtained from either P-Asserted-Identity header or From header.
 - b. calledPartyId: global number if available (optional)
6. A response is received containing a calling name.
7. The display-name portion of the P-Asserted-Identity and From headers are overwritten with the calling name received from Calling Name server in case the display-name was already present or it is added to the header if it was not present.

2.28.2.3 CNIP on originating MMTel AS (OCNIP)

The display name of Served user is provisioned in the subscriber data. Originating MMTel AS serving the user will change headers based on the display name retrieved from the subscriber data. The case that is shown here describes where a global number identity of the originating user can be extracted from the message headers, the mode is “always” and originating MMTel AS successfully retrieves display name from subscriber data.

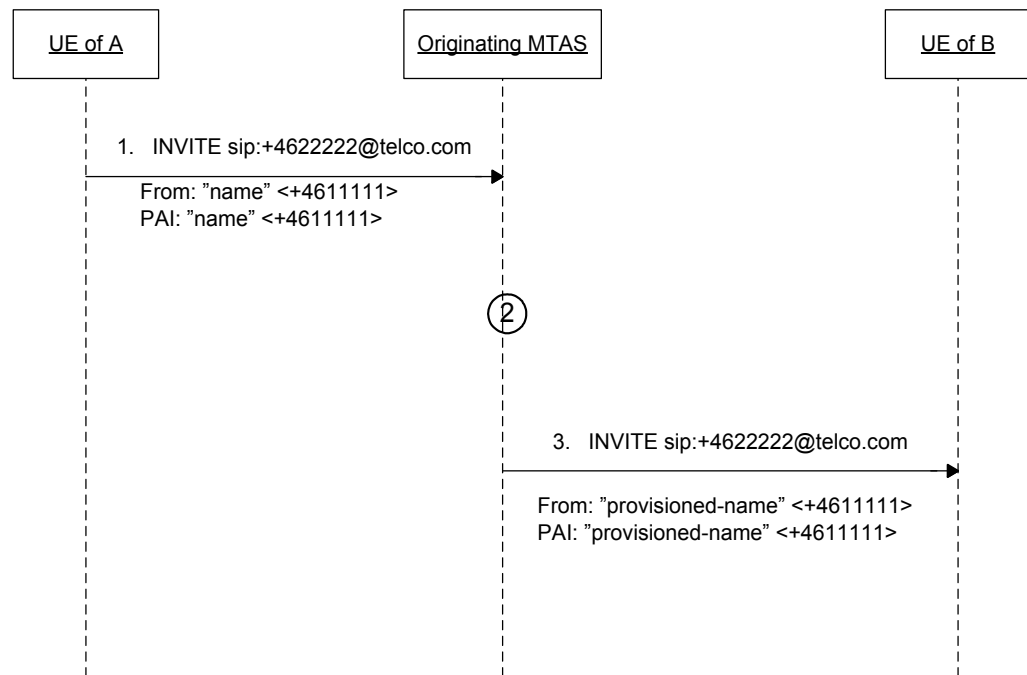


Figure 60 Call with OCNIP Service

- 1 User A makes an outgoing call. The SIP INVITE message will reach MTAS on the originating side containing amongst others the From and P-Asserted-Identity (PAI) headers.
- 2 The OCNIP service within MTAS inserts/replaces the display name of the PAI and From header with the display name provisioned in the subscriber data.
- 3 MTAS sends the message towards CSCF and the final destination. The INVITE will reach the called user's UE with the From and PAI headers set to the display name added by OCNIP service in Originating MTAS.



2.28.2.4 FIP

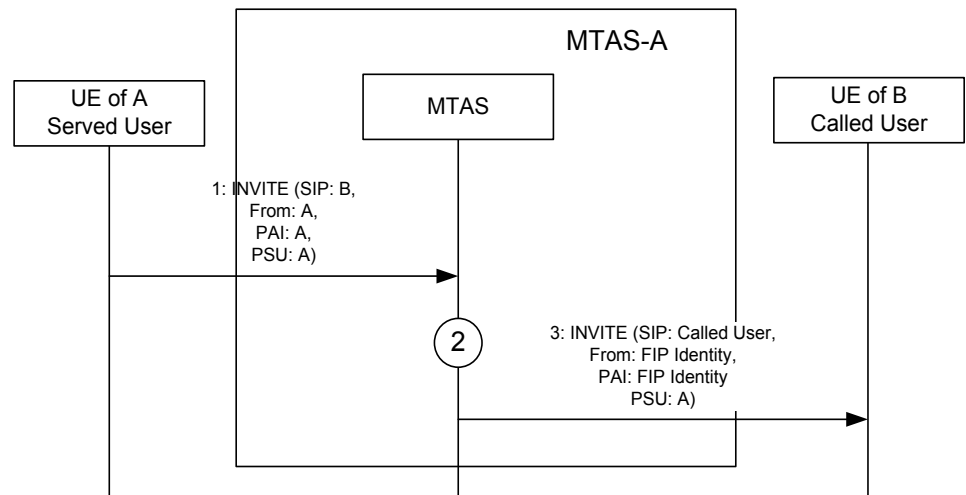


Figure 61 - Call with the FIP service

- 1 User A makes an outgoing call. The SIP INVITE message will reach MTAS on the originating side containing amongst others the From and P-Asserted-Identity (PAI) headers.
- 2 The FIP service within MTAS replaces the content of the PAI and From header with the identity defined in the FIP service. The P-Served-User (PSU) header is not altered.
- 3 MTAS sends the message towards CSCF and the final destination. The INVITE will reach the called user's UE with the From and PAI headers set to the FIP identity. The PSU header will be removed by the originating IMS network.

2.28.3 Service Interaction

Identity Presentation interacts with CDIV and CB. See chapter 2.4.3.

1. Served user identity is hidden and FIP identity is presented when a call completion recall is made towards the called user.
2. Served user identity is hidden and FIP identity is presented when a call is made towards the user registered with the FIP identity.
3. Served user identity is hidden and FIP identity is presented when a collocated Ad-hoc Conference is created by the served user.
4. Served user identity is hidden when the served user forwards the call due to the Call Diversion service.



2.28.4 Self administration

The end user can by itself administer his/her services with the following options via the Ut interface or SSCs:

- Set each of OIP, OIR, TIP, TIR, CNIP and FIP to activated or de-activated.

Via Ut only the end user can also:

- Set the default behavior of OIR and TIR when in temporary mode to *presentation restricted* or *presentation not restricted*

2.28.5 Provisioning

MTAS enables the operator to configure the service on user-level through the CAI3G interface. Possible settings are:

- OIP, OIR Override, OIR, TIP, TIR Override, TIR, CNIP and FIP activation (activate/deactivate)
- OIR and TIR mode (permanent mode/temporary mode)
- OIR Restriction Level (restrict asserted identity / restrict all private information)
- FIP Identity, FIP suppression

2.28.6 Configuration

Examples of node-level configuration parameters related to the Identity Presentation services are:

- Enable and disable of all Identity Presentation services (i.e. OIP, OIR, TIP, TIR, CNIP, OCNIP and FIP)
- OIP Global Restriction list
- CNIP Mode (always/ interrogate-on-unavailability)
- Default Display Name

2.28.7 Performance Management

Examples of performance counters related to the Identity Presentation services are:

- Number of successful invocations with OIR Override
- Number of sessions where the user, with OIR in temporary mode, decides on a per call basis to allow presentation
- Number of successful invocations of FIP



2.28.8 Fault Management

For information on the alarm, refer alarm OPI [62], [63] .

2.29 Advice of Charge (AOC)

Three AOC service types are available with the service.

- The Advice of Charge at Communication Set-up (AOC-S) service type enables a user to receive information about the applicable charging tariff at communication set-up and whenever the applicable tariff changes during the communication.
- The Advice of Charge, During the Communication (AOC-D) service type enables a user to receive information on the cost incurred for a communication upon session establishment, at periodic intervals throughout the communication and when the communication is terminated.
- The Advice of Charge at the End of the Communication (AOC-E) service type enables a user to receive information on the cost incurred for a communication when the communication is terminated.

Users can be provisioned with any combination of service types. A user provisioned with both the AOC-D and AOC-E service types will only receive information once on the recorded cost for the communication when the communication is terminated.



2.29.1 Example Call Flow (AOC-S)

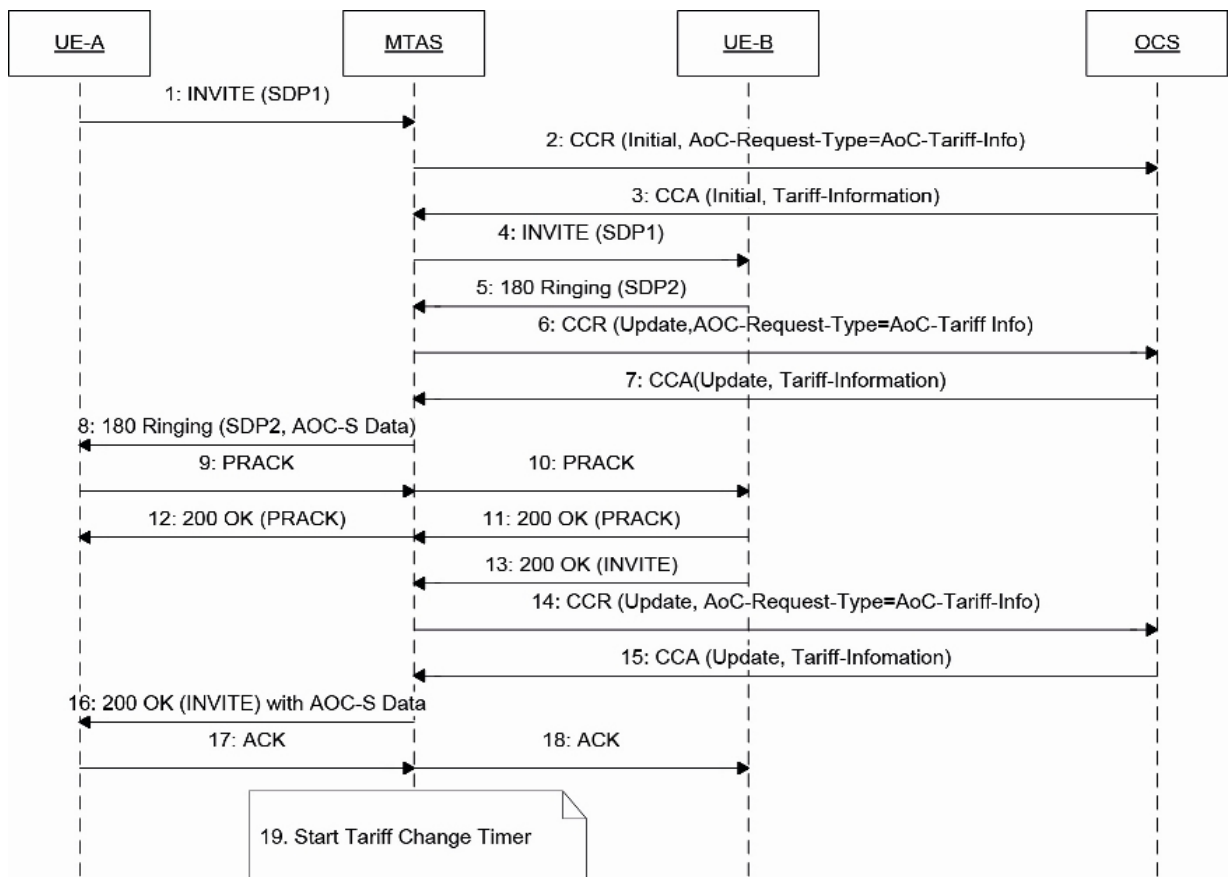


Figure 62 - INVITE with SDP offer and SDP answer in 180 Ringing

- 1 UE-A sends an INVITE with an SDP offer. The INVITE must indicate either "Require: 100rel" or "Supported: 100rel".
- 2 MTAS read the user's service data. User A has AOC-S service type.
MTAS sends a Credit Control Request (Initial Request) to the OCS.
- 3 The OCS responds with a Credit Control Answer including the tariff information. The tariff information received in this message is not used. MTAS stores the Charging Server Host Identity from the Origin-Host AVP.
- 4 MTAS sends INVITE to UE-B.
- 5 UE-B sends 180 Ringing.
- 6 As this is the 1st 1XX response received for this session, MTAS sends a Credit Control Request (Update) as shown in
- 7 The OCS responds with the Credit Control Answer including the tariff
- 8 MTAS forwards 180 Ringing to UE-A with an AOC MIME body containing an early indication of the AOC-S data.

The tariff information received is mapped to an AOC MIME body depending on the chosen AOC MIME Type.



- 9 UE-A sends a PRACK to MTAS.
- 10 MTAS forwards PRACK to UE-B.
- 11 UE-B responds to PRACK with 200 OK.
- 12 MTAS forwards 200 OK to UE-A.
- 13 UE-B sends a 200 OK response to the INVITE.
- 14 MTAS sends a Credit Control Request (Update Request) to the OCS.
- 15 The OCS responds with a Credit Control Answer including the tariff information.
- 16 MTAS sends the 200 OK response to the INVITE to UE-A with an AOC MIME body containing AOC-S data.

The tariff information received is mapped to an AOC MIME body depending on the chosen AOC MIME Type.
- 17 UE-A sends an ACK to MTAS.
- 18 MTAS forwards ACK to UE-B.
- 19 If the tariff information provided by the OCS contains the 'next tariff' as well as the 'current tariff', MTAS starts a timer which will expire when the 'tariff-time-change' is reached.

If the 'tariff-time-change' has already passed, MTAS takes a new action.

2.29.2 Service Interaction

2.29.2.1 Advice of Charge

Advice of Charge is provided for successfully established communication sessions only and therefore there is no interaction with any services that prevent communication establishment (e.g. Communication Barring).

Advice of Charge for communication sessions that are established by MTAS for CCBS purposes is not supported in the current product version. MTAS does not initiate an Advice of Charge session with the OCS for the MTAS-to-B leg.

2.29.3 Configuration

- Activation (active/ disabled)
- Timer
- OCS Address and relation data
- Behavior for session when AoC can not be provided



2.29.4 Performance Management

Counters for successful and unsuccessful –internal and external

- AOC-D
- AOC-E
- AOC-S

2.30 Number Portability

2.30.1 Description

The Number Portability (NP) service main functionality is to identify whether the other party URI is within the original/donor network or has been ported to other network. If the other party URI has been ported to other network, the NP service conveys this NP-related information to the charging system.

The NP service is applicable only if the other party URI is either in tel URI or in Embedded tel URI format. In the case of outgoing communication the other party is the Called party, while in the case of incoming communication, the other party is either the Calling party or the party that diverted or distributed the communication before reaching the served user.

As a sub-function of the NP service for outgoing communication, MTAS plays an announcement to notify the caller that the called party is ported. The rationale behind the NP announcement is that the caller might get different charging rates for this particular call. The NP announcement is played in basic MMTel calls and in Ad-hoc Conference scenarios.

MTAS plays NP Announcement using Generic Announcement. Announcements are configured with Routing Number (RN) Prefix. If there is no matching Generic Announcement for a RN, no announcement will be played.

NP announcements shall not be configured for home RN (i.e. network served by MTAS) and thus no announcement is played when the user is ported within the network (see virtual RN numbers) or when called party is a ported in subscriber. However, NP announcements shall be configured for all other relevant RNs and thus when the called party is ported to a different network the appropriate announcement is played.

The main use cases of this service are:

- NP service for outgoing communication
- NP service for incoming communication

2.30.2 Example call flow

There are two main call flows for the Number Portability service:



- Successful NP Lookup for Outgoing Communication where called-party is ported
- Successful NP Lookup for Regular Incoming Communication where Caller is ported

2.30.2.1 Successful NP Lookup for Outgoing Communication where called-party is ported

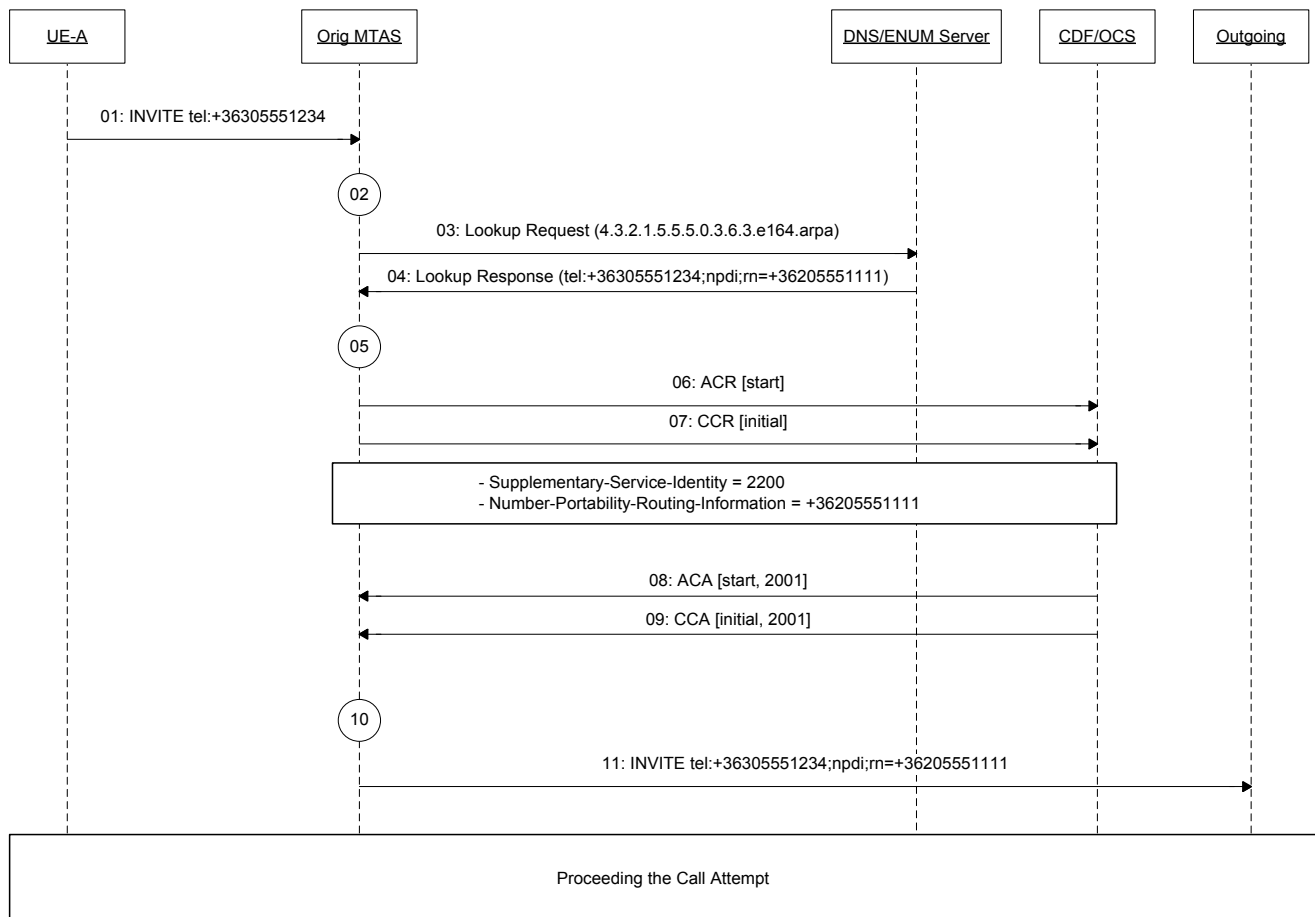


Figure 63 Successful NP Lookup for Outgoing Communication where called-party is ported

1. UE-A sends SIP INVITE request towards `tel:+36305551234`.
2. The SIP INVITE arrives at the originating MTAS. The NP service is triggered, and after ensuring that the Request-URI contains no NP parameter (rn and npdi) it determines to perform an NP lookup.
3. MTAS sends NP Lookup request towards ENUM/DNS Server by using processed phone number of the Request-URI.
4. MTAS receives NP Lookup response where the result indicates that the called party number has been ported out from the Donor network. In this case, the `rn` parameter contains `+36205551111` as the value.



5. MTAS conveys the obtained NP-related information towards Charging System.
6. If offline charging is active, then MTAS sends ACR Start request towards the CDF..
7. If online charging is active, then MTAS sends CCR Initial request towards the OCS..
8. MTAS receives a response to the ACR request with successful indicator (code 2001).
9. MTAS receives a response to the CCR request with successful indicator (code 2001).
10. MTAS modifies the Request-URI header by inserting `npdi` parameter and `rn` parameter set to `+36205551111`.
11. MTAS forwards the SIP INVITE to the next node and the call attempt proceeds as in the normal MMtel.



2.30.2.2 Successful NP Lookup for Regular Incoming Communication where Caller is ported

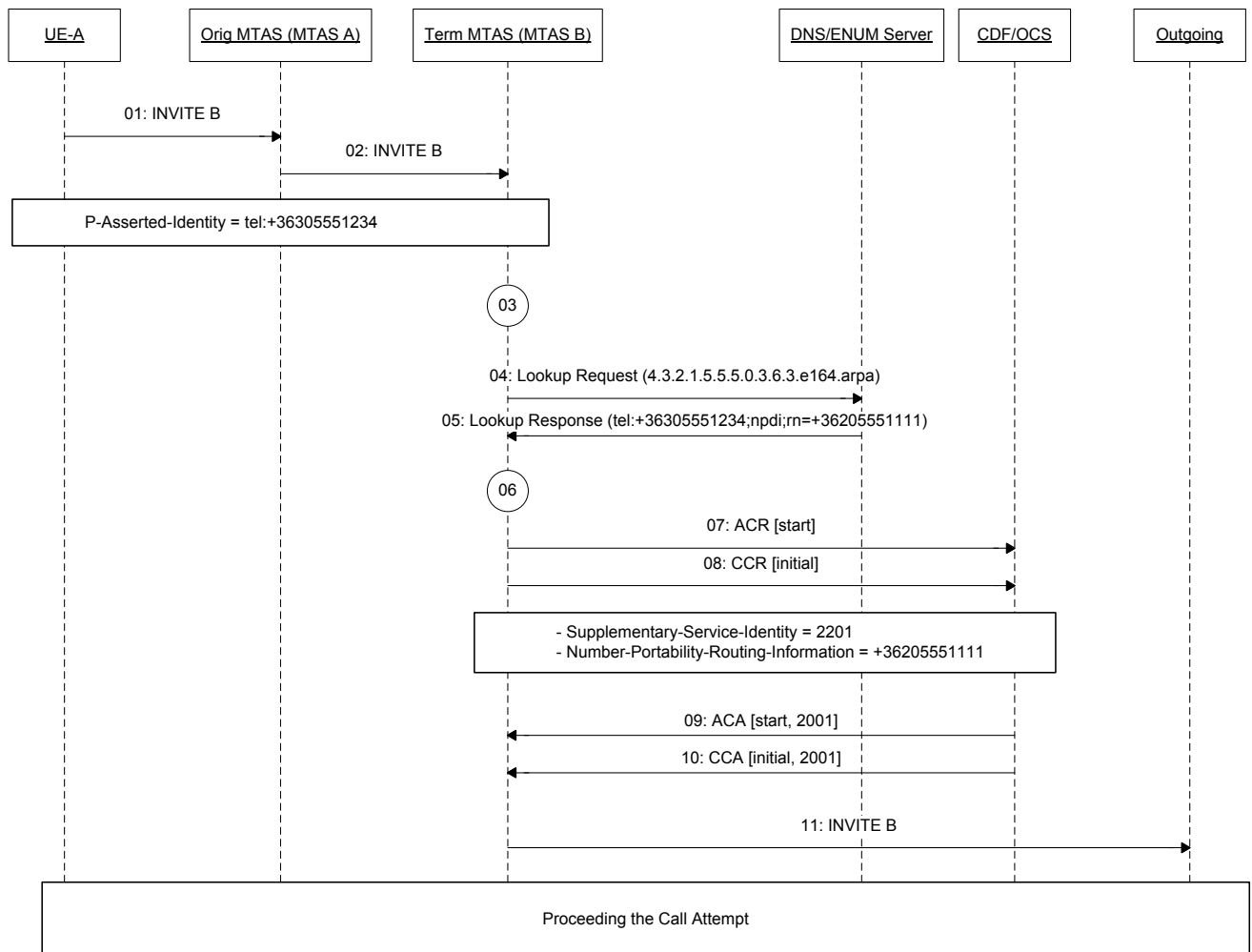


Figure 64 Successful NP Lookup for Regular Incoming Communication where Caller is ported

1. UE-A sends SIP INVITE request towards B. The P-Asserted-Identity header contains tel:+36305551234.
2. The SIP INVITE arrives at the originating MTAS and then forwarded to the terminating MTAS.
3. The SIP INVITE arrives at the terminating MTAS. The NP service is triggered and determines that the SIP INVITE comes directly from the Caller after evaluating the History-Info header. MTAS determines to perform an NP lookup.
4. MTAS sends NP Lookup request towards ENUM/DNS Server by using processed phone number of the P-Asserted-Identity.



5. MTAS receives NP Lookup response where the result indicates that the phone number has been ported out from the Donor network. In this case, the `rn` parameter contains +36205551111 as the value.
6. MTAS steps up the `MtasNpEnumResponse` counter, keyed with `RnNpdi`, and then determines to convey the obtained NP-related information towards Charging System.
7. If offline charging is active, then MTAS sends ACR Start request towards the CDF..
8. If online charging is active, then MTAS sends CCR Initial request towards the OCS..
9. MTAS receives a response to the ACR request with successful indicator (code 2001).
10. MTAS receives a response to the CCR request with successful indicator (code 2001).
11. MTAS forwards the SIP INVITE to the next node and the call attempt proceeds as in the normal MMTel.

2.30.3 Charging

When the NP service is triggered by outgoing communication attempt and followed by successful Number Portability lookup resulting in receiving information that the called party is ported, it generates charging message for Originating Charging. The charging message includes a service specific AVP identifying the type of NP service (in this case the NP service for outgoing communication), and specific AVP containing the routing number information.

When the NP service is triggered by incoming communication attempt and followed by successful Number Portability lookup resulting in receiving information that the caller is ported, or the party that diverts or distributes the communication is ported, it generates charging message for Terminating Charging. The charging message includes a service specific AVP identifying the type of NP service (in this case the NP service for incoming communication), and specific AVP containing the routing number information.

2.30.4 Service interactions

2.30.4.1 Communication Barring

In the Originating MTAS, the NP service is triggered before Outgoing Communication Barring (OCB) service. Therefore, the NP service when triggered by an outgoing communication may perform an NP lookup although the Called-Party is barred by OCB.

If the called party is ported out subscriber, then the calling party may hear NP Announcement, followed by OCB announcements (if any).



On the other hand, in the Terminating MTAS, the NP service is triggered before Incoming Communication Barring (ICB) service. Therefore, the NP service when triggered by an incoming communication may perform an NP lookup although the Caller is barred by ICB.

2.30.4.2 Communication Diversion

In the Transit MTAS, when an incoming communication is diverted by any type of Communication Diversion (CDIV) service, the generated SIP INVITE towards new target will invoke NP service if the `mtasNpControl` attribute is set to either 1 or 3. If invoked, the NP service will perform an NP lookup by using as the input the new target URI if it is either in tel URI or in Embedded tel URI format.

2.30.4.3 Flexible Communication Distributions

In the Transit MTAS, when an incoming communication is distributed by Flexible Communication Distribution (FCD) service, every generated SIP INVITE will invoke NP service if the `mtasNpControl` attribute is set to either 1 or 3. If invoked, the NP service will perform an NP lookup for every distributed target that has either tel URI or embedded tel URI format. Every successful NP lookup will be followed by NP-related information conveyance towards the Charging System.

2.30.4.4 Carrier Select and Carrier Select Rn

The NP service for outgoing communication is invoked before Carrier Select or Carrier Select Rn service. However, The Request-URI containing Carrier Select Code (CSC) in the SIP INVITE will be processed by Dialed String Analysis (DSA) in the Number Normalization service, which is invoked before the NP service.

After the Dialed String Analysis process, the Request-URI header may contain `cic` parameter if the Carrier Select service is provisioned or `rn` parameter if the Carrier Select Rn service is provisioned.

2.30.4.5 Carrier Pre-Select and Carrier Pre-Select Rn

The NP service for outgoing communication is invoked before Carrier Pre-Select or Carrier Pre-Select Rn service.

When the Request-URI of an initial SIP INVITE contains `rn` parameter inserted to be processed by Carrier Pre-Select Rn service and it does not contain `npdi` parameter, the NP service checks the `mtasNpRnBeforeNpLookup` attribute value.

When the Request-URI of an initial SIP INVITE contains `cic` parameter inserted to be processed by Carrier Pre-Select service and it does not contain `npdi` parameter, the NP service will perform an NP lookup.



However, if the `mtasNpRnBeforeNpLookup` attribute is set to 0, the NP service will remove the `rn` parameter enclosed in the Request-URI of the initial SIP INVITE and then performs NP lookup..

When the Request-URI of an initial SIP INVITE contains `cic` parameter inserted to be processed by Carrier Select service and it does not contain `npdi` parameter, the NP service will perform an NP lookup..

2.30.4.6 Ad-hoc Conferencing

In the Conference Focus the NP service is invoked after Conference service. When the Conference service receives an invitation attempt towards participants, after processing and validating the invitation it generates new SIP INVITE-s addressed to invited participants. Consequently, every generated SIP INVITE will invoke NP service if the `mtasNpControl` attribute is set to either 1 or 3. If invoked, the NP service will perform an NP lookup for every distributed SIP INVITE message towards participant that has either tel URI or embedded tel URI format. Every successful NP lookup will be followed by NP-related information conveyance towards the Charging System.

2.30.4.7 Number Normalization

In the Originating MTAS, the NP service is invoked after the Number Normalization service. Therefore, the Request-URI header of the SIP INVITE that triggers NP service has a normalized URI.

2.30.5 Configuration

Node level configuration parameters for service functionalities:

- Activation (active / disable)
- Use of NP service (in originating MTAS only, in terminating MTAS only, or in both)
- Whether to perform NP Lookup in cases where NP related information is already present in Request URI.
- Generic Announcement with Routing Number Prefixes.

2.30.6 Performance management

The following performance counters are provided by MTAS to evaluate the usage and quality of service:

- The number of NP lookup attempts
- The number of ENUM queries performed resulting in NP related information



2.31 Account activation

2.31.1 Description

Account activation is an online charging feature enabling the operator to prompt the end user to provide a pin for activation of its online charging account at the first call attempt.

The prompt and collect functionality of MMTel AS is initiated by the OCS when the end user makes the first call attempt. OCS requests MMTel AS to play an announcement and collect, via DTMF, a pin code from the end user. MMTel AS sends the pin to OCS which validates the pin for account activation.

2.32 Administration of user's language preference

Administration of user's language preference is an online charging feature enabling the operator to initiate an update of the user's preferred language stored in HSS.

When MMTel AS receives a language indicator from OCS in a CCA response, MMTel AS compares the received language indicator with the preferred language present in the subscriber's transparent data. If the two are different, MMTel AS updates any internally stored preferred language as well as the preferred language stored in the subscriber's transparent data on the HSS.

2.32.1 Configuration

Node level configuration parameters for the Administration of user's language preference:

- Activation (active / disable)
- Ro language mapping table (mapping the Ro language code to HSS language code and vice versa)

2.32.2 Performance management

Performance counter related to the Administration of user's language preference:

The number of mapping keys not found when searching for a key in the Ro language mapping table

2.33 Network Provided Location Information (NPLI)

2.33.1 Description

The NPLI feature consists of a procedure for Location Information retrieval from HSS and the inclusion of resulting Network Provided Location Information (NPLI) in the PANI for originating and terminating calls.



If NPLI retrieval is enabled in the MMTel AS for the call case and if no valid network PANI is received in the SIP, location information will be fetched from HSS over Sh interface on originating INVITE (originating case) or on 180/200 response (terminating case), it also can be configured to be fetched on INVITE for terminating case. The access domain used in the retrieval is based on the subscriber's registration domain (CS, PS MME, PS SGSN) or on the NPLI configuration. The location information is populated into a network provided PANI which is inserted/updated in the SIP signaling.

In addition, NPLI can also be triggered from the Japanese Charging service and Communication Barring service for ICB on the terminating INVITE. In this case an active location update is requested.

To reduce the number of Sh requests when multiple services want to access the same Location Information, a Location Information cache mechanism is used.

2.33.2 Example Call Flow

See Japanese Charging service call flows (2.34.2) where the NPLI is used.

2.33.3 Service Interactions

2.33.3.1 Communication Barring

Location information as retrieved by the NPLI feature is used by OCB for originating case when barring rules contains roaming condition.

Location information with active location update is retrieved on terminating INVITE by ICB when barring rules contains roaming condition. This information is stored with the session and later used by NPLI feature on 180/200 response and inserted in PANI

2.33.3.2 Japanese Charging

Location information using NPLI feature with active location update is retrieved on terminating INVITE by JC service when CFU applies to the call.

2.33.4 Self administration

N/A

2.33.5 Provisioning

N/A.

2.33.6 Configuration

The node-level configuration attributes related to the NPLI feature are:

- Disable or Enable originating NPLI retrieval in MMTel AS.



- Disable or Enable terminating NPLI retrieval in MMTel AS.

2.33.7 Performance Management

N/A

2.33.8 Fault Management

N/A

2.34 Japanese Charging

2.34.1 Description

The Japanese Charging (JC) service provides Interconnection Charge Billing System (ICBS) and Flexible Charging (FCH) functionality defined by TTC, the Japanese standardization group, on MMTel AS.

MMTel AS transfer the interconnection charging information per call basis in signaling messages over the Session Initiation Protocol (SIP) and stores it to be sent in the Accounting Request (ACR) start messages when offline charging is applicable. Interconnection charging information is based on the subscriber's location information as made available by the NPLI feature (see 2.33) and it is divided into accounting and charging information.

- Accounting information is transferred in forward (originating side to terminating side) and backward (terminating to originating side) direction in a call chain (e.g. ICBS data).
- Charging information is transferred from charging determination point towards the charging point i.e. only in backward direction (e.g. FCH data). The backward direction is from the terminating network to the originating network.

The JC service also handles FCH data for Telephone Directory Service (TDS).

The purpose is to provide carriers with a flexible system for billing interconnection charges in interconnection patterns involving more than two carriers.

CM parameter `mtasJcBehaviorType` specifies how ICBS and FCH information is sent over the network. It can be set to 'headers' (0) or 'parameters' (1), which is referred as JC service behavior 'headers' or 'parameters' respectively in below sections.



2.34.1.1 ICBS and FCH for originating MMTel AS

For every initial INVITE message received without ICBS data, MMTel AS will add the forward ICBS data. MMTel AS determines the Charge Area (CA) and Carrier Code (CC) based on the originating subscriber's location information. The ICBS data, Additional User Category (AUC), is read from configuration and other ICBS data are fixed. When JC service behavior is set as 'parameters', for every initial INVITE message received with ICBS data, MMTel AS will just forward ICBS data.

2.34.1.2 ICBS and FCH for terminating MMTel AS

For 18x/200 OK (INVITE) messages received without ICBS data, MMTel AS will add the backward ICBS data. MMTel AS determines the Charge Area (CA) and Carrier Code (CC) based on the terminating subscriber's location information. The ICBS data, Additional User Category (AUC), is read from configuration and other ICBS data are fixed. When JC service behavior is set as 'parameters', for 18x/200 OK (INVITE) message received with backward ICBS and FCH data, MMTel AS will just forward ICBS and FCH data, and MMTel AS will also determine and add FCH data as well as ICBS data.

2.34.1.2.1 Forward ICBS data for terminating MMTel AS

In case of communication diversion the forward ICBS data will be set in the INVITE sent towards the redirection target. Same CA, CC and AUC values are used as for backward ICBS data but some of the fixed ICBS data is set differently for the forward ICBS data.

2.34.1.3 TDS and FCH for originating MMTel AS

The number inquiry assistance service is offered by NTT PSTN networks in Japan. TDS can be inquired by VoLTE users through dialing 104. Each TDS query is verbally asked to the assistance and results is sent to the MMTel AS in a SIP INFO message. When FCH data is present in the received SIP INFO message, MMTel AS terminates the SIP INFO message by sending a 200 OK (INFO) as response.



2.34.2 Example Call Flows

2.34.2.1 ICBS and FCH for originating MMTel AS

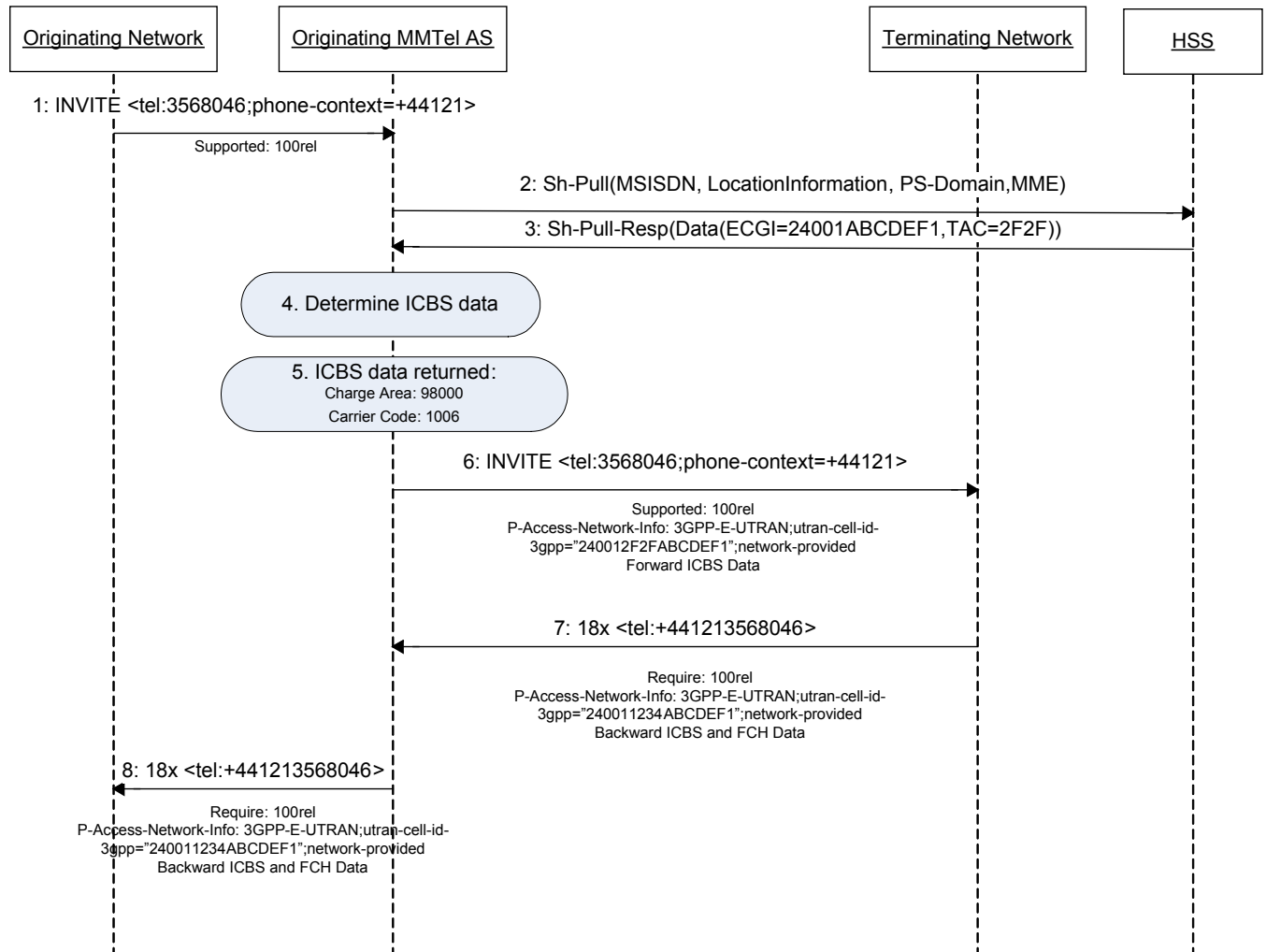


Figure 65 ICBS and FCH for originating MMTel AS

1. MMTel AS receives initial INVITE without a network provided PANI header and without forward ICBS data.
2. MMTel AS sends Sh-Pull request to HSS to fetch Network Provided Location Information (NPLI) for the originating subscriber, when NPLI fetching is configured on node level.
3. MMTel AS receives Sh-Pull response with E-UTRAN Cell Identity (ECGI) and Tracking Area Code (TAC) from the HSS.
4. Based on the NPLI, MMTel AS determines the forward ICBS data, CA and CC.



5. MMTel AS adds the CA and CC, the configured AUC value and the fixed ICBS data to the initial INVITE. MMTel AS stores the forward ICBS data to be included later in ACR(start). The P-Access-Network-Info header is added to the INVITE with the NPLI data fetched from HSS.
6. MMTel AS sends the INVITE with the added forward ICBS data.
7. MMTel AS receives 18x with backward ICBS and FCH data and stores the data to be included later in ACR(start).
8. MMTel AS forwards the 18x and call setup continues.

2.34.2.2 ICBS and FCH for terminating MMTel AS

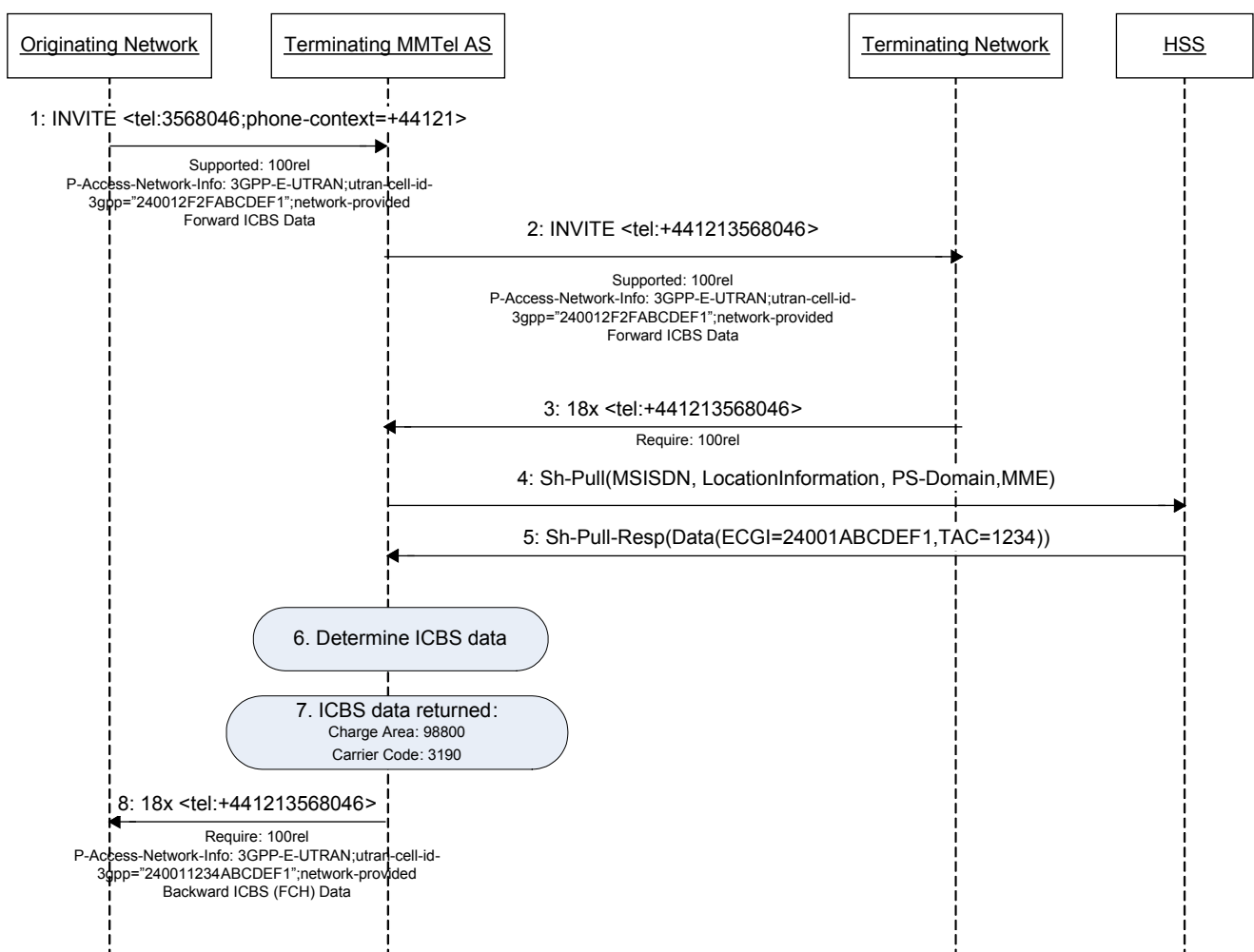


Figure 66 ICBS and FCH for terminating MMTel AS

1. MMTel AS receives INVITE with forward ICBS data and stores this data for later to be included in ACR(start) when terminating offline charging is applicable.
2. MMTel AS sends the INVITE.



3. MMTel AS receives 18x response without a network provided PANI header.
4. MMTel AS sends Sh-Pull request to HSS to fetch NPLI for the terminating subscriber, when NPLI fetching is configured on node level.
5. MMTel AS receives Sh-Pull response with ECGI and TAC from the HSS.
6. Based on the NPLI, MMTel AS determines the backward ICBS data, CA and CC. When JC service behavior is set as 'parameters', MMTel AS also determines FCH data and adds to the 18x response.
7. MMTel AS adds the CA and CC, the configured AUC value and the fixed ICBS data to the 18x response. MMTel AS stores the backward ICBS data to be included later in ACR(start). The P-Access-Network-Info header is added to the 18x response with the NPLI data fetched from HSS.
8. MMTel AS sends the 18x with the added backward ICBS data and call setup continues.

2.34.2.3 TDS and FCH originating MMTel AS

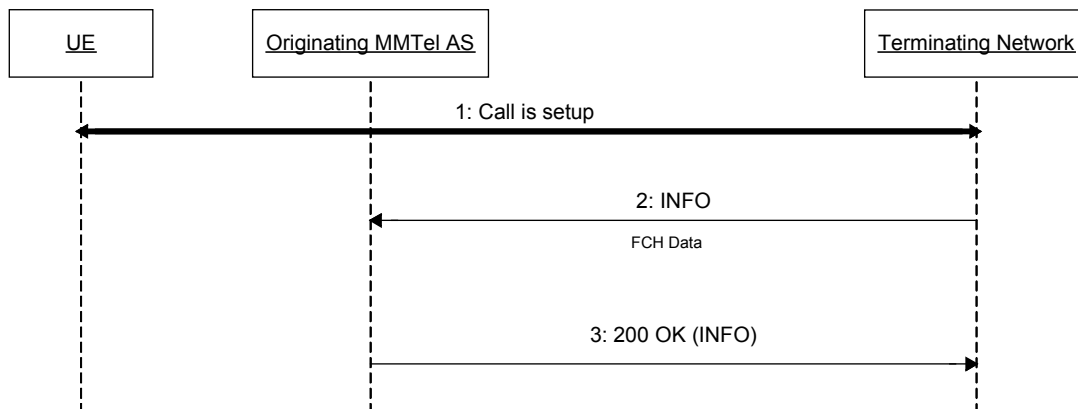


Figure 67 TDS and FCH for originating MMTel AS

1. Call is setup as described in 2.34.2.1.
2. MMTel AS receives a SIP INFO message with FCH data present and the SIP INFO message is sent within an existing session.
3. MMTel AS terminates the SIP INFO by responding with 200 OK (INFO) and triggers the sending of ACR (interim) message where the FCH data received is reported.



2.34.3 Charging

The ICBS and FCH data is reported in offline charging messages, ACR (start), generated during the setup of an MMTel session. FCH data is reported in the offline charging messages ACR(interim), generated when a SIP INFO with FCH data is received in the MMTel AS and Japanese Charging is enabled.

When JC service behavior is set as 'headers', the AVPs used for reporting ICBS and FCH data are listed below:

- Forward-TTC-Charging-Headers AVP contains the AVPs listed below populated with the SIP headers carrying forward ICBS data:
 - Charging-Area
 - Carrier-Information
 - Additional-User-Category
- Backward-TTC-Charging-Headers AVP contains the AVPs listed below populated with the SIP header carrying FCH data and the SIP headers carrying the backward ICBS data:
 - Charging-Area
 - Carrier-Information
 - Additional-User-Category
 - Flexible-Charging-Info

When JC service behavior is set as 'parameters', the AVPs used for reporting ICBS and FCH data are listed below:

- Forward-TTC-Charging-Parameters AVP contains forward ICBS data.
- Backward-TTC-Charging-Parameters AVP contains backward ICBS data and FCH data.

2.34.3.1 Originating charging

MMTel AS reports the forward ICBS data added to the initial INVITE or received in incoming INVITE as described in **Error! Reference source not found.2.34.1.1** and the FCH and the backward ICBS data received in the 18x/200 OK (INVITE) messages in ACR (start).

MMTel AS reports the FCH data received in SIP INFO in ACR (interim).



2.34.3.2 Terminating charging

MMTel AS reports the forward ICBS data received in initial INVITE and the backward ICBS data added to 18x/200 OK (INVITE) messages or received in incoming 18x/200 OK (INVITE) as described in 2.34.1.2 in ACR (start) for call leg A->B.

For CDIV scenarios, MMTel AS reports the forward ICBS data set in the initial INVITE sent towards the redirection target and the backward ICBS and FCH data received in 18x/200 OK (INVITE) messages from the redirection target in ACR (start) for call leg B->C.

2.34.4 Service Interactions

2.34.4.1 Communication Barring

2.34.4.1.1 Incoming communication barring

When Incoming Communication Barring (ICB) for Dynamic Black List (DBL) and Anonymous Communication Rejection (ACR) are configured with established session announcement, MMTel AS will add backward ICBS data to the generated 183 Session Progress message. And MTAS reports both forward and backward ICBS data in ACR (event).

When barring service is not configured with established session announcement, MMTel AS will not add ICBS data to the SIP error response but report forward ICBS data in the ACR(event) message.

2.34.4.1.2 Outgoing communication barring

When a call is barred by Outgoing Communication Barring (OCB) forward ICBS will be reported in the ACR(event) message.

2.34.4.2 Communication Diversion (CDIV)

MMTel AS executing CDIV, executes the terminating JC functionality for the B-user on the leg (A->B). For the new diverted call leg (B->C), the same MMTel AS executes originating JC functionality for B-user.

This means that for B->C leg, the forward ICBS parameters are derived based on the redirecting user (B) instead of the calling user (A). For the A->B leg, the backward ICBS parameters are derived based on the redirecting (served) user (B).

For CDIV use cases without SIP signaling to/from the B-target, for example Communication Forwarding Unconditional (CFU), the NPLI will be fetched from the HSS or the provided home location will be used in order to get forward ICBS data for B to include in INVITE sent towards C. No ICBS or FCH data will be exchanged between A and C.



Service interaction between Japanese Charging and CDIV with originating AS chaining is supported. In this case Japanese Charging will not determine the ICBS data again when the INVITE towards C is received by B's originating MMTel AS since the ICBS data is already present in the SIP INVITE.

2.34.4.3 Ad-hoc conference

In the MMTel AS and Conference AS collocated deployment MMTel AS will determine CA and CC based on the conference creator's location and use that for backward ICBS data for the conference creation call leg.

For the focus (conference creator) to Conference Participant (CP) call leg, i.e. dial-out, the focus adds the conference creator's forward ICBS data to the INVITE towards the CP. The terminating MMTel AS for each CP will add its backward ICBS data based on that CP.

Service interaction between Japanese Charging and Ad-hoc conference collocated deployment with originating AS chaining is supported. In this case Japanese Charging will not determine the ICBS data again when the INVITE to the CP is received by the conference creator's originating MMTel AS since the ICBS data is already present in the SIP INVITE.

Japanese Charging and Ad-hoc conference in stand-alone deployment is not supported. Japanese Charging is not executed in the separate Conference AS.

2.34.4.4 Customized Alerting Tones

For service interaction between Japanese Charging and CAT, the backward ICBS data will be added to the generated 183 Session Progress response.

2.34.4.5 Communication Waiting

For the Communication Waiting (CW), MMTel AS looks up the backward ICBS data for the call A to B and for the call C to B based on the location for B. If NPLI is enabled the location information for B will be fetched from the HSS for both calls (A to B and for C to B). Both backward and forward ICBS data is reported in the ACR(start) for both calls in the same way as done for a basic call between two subscribers.

ICBS logic is only performed during the setup of a call. No ICBS logic is performed when B puts A on hold in order to answer the waiting call from C or when B resumes the call with A.

2.34.4.6 Three Party

For the Three Party (3PTY) interaction with JC, MMTel AS looks up forward ICBS data for the call A to B and for the call A to C based on A's location information. If NPLI is enabled the location information for A will be fetched from the HSS for both calls (A to B and for A to C). Both backward and forward ICBS data is reported in the ACR(start) for both calls in the same way as done for a basic call between two subscribers.



When the 3PTY conference creator creates the 3PTY conference by sending the INVITE to originating MMTel AS, the creators (A's) backward ICBS data will be added by MMTel AS to the 200 OK (INVITE) towards the creator. The backward ICBS data for the 3PTY creator will be sent in the ACR(start).

2.34.4.7 Flexible Communication Distribution (FCD) and Session Transfer to Own Device (STOD)

When outgoing INVITE is sent out, if this is the primary user B, the leg is not a transit leg and ICBS from incoming INVITE is sent in the INVITE; otherwise the leg is a transit leg and B's forward ICBS data are sent in the INVITE.

When provisional response is received, if this is the primary user B, MTAS will generate backward ICBS data if they are not received in the message; if this is not primary user, MTAS will always receive backward ICBS data.

When 200 OK (INVITE) is received from one device, the corresponding forward and backward ICBS data are reported in ACR(start); If the answered device is not the primary user B, there are two ACR(start) messages with ICBS data for both non-transit and transit legs; and for the other unsuccessful call legs ACR(event) is reported with ICBS data. All ICBS data reported are determined in previous two paragraphs accordingly.

2.34.4.8 Communication Completion

Service interaction between Japanese Charging and Communication Completion is not supported. Japanese Charging is not triggered for the recall part of the CC service.

2.34.4.9 Parlay X

Service interaction between Japanese Charging and Parlay X Third Party Call and Call Notification is not supported. Japanese Charging is not triggered for these use cases.

2.34.4.10 gsmSSF

Service interaction between gsmSSF and Japanese Charging is not supported.

2.34.4.11 Gateway Model

When Gateway Model (GM) is used, there is only one dialog towards the caller, hiding multiple outgoing early dialogs. Japanese Charging will send the ICBS and FCH data multiple times even when GM maps messages to one dialog.



2.34.5 Self administration

- N/A

2.34.6 Provisioning

Japanese Charging service utilizes the location information to determine the ICBS data, CA and CC. When no network provided P-Access-Network-Information (PANI) header is available, MMTel AS will fetch NPLI from HSS when configured to do so. If the answer from HSS does not contain a NPLI or the node configuration does not require a NPLI, MMTel AS will use the operator provided home location information of the subscriber.

2.34.7 Configuration

The node-level configuration related to the Japanese Charging service are:

- Enable and disable the Japanese Charging service (administrative state)
- Whether the call should be 'terminated' or 'continued' without ICBS data when the CA and CC data cannot be determined.
- The Additional-User-Category parameter.
- The location to Charge Area (CA) and Carrier Code (CC) mapping.
- Configure the policy for originating/terminating NPLI retrieval.
- Configure the service behavior by the parameter `mtasJcBehaviorType`.

2.34.8 Performance Management

- N/A

2.34.9 Fault Management

For information on the alarm, refer alarm OPI [64] .

2.35 Call Admission Control (CAC)

2.35.1 Description

The User Call Admission Control (CAC) supplementary service may be used in two modes: Fixed (UCAC) and Multi-Device (MDUCAC)

Fixed mode (UCAC) enables the operator to restrict:

- the number of all sessions a served user is involved in
- the number of all originating sessions a served user is involved in
- the number of all terminating sessions a served user is involved in
- the number of active sessions a served user is involved in
- the number of active originating sessions a served user is involved in



- the number of active terminating sessions a served user is involved in
- the number of waiting sessions a served user has
- the number of fixed active sessions a served user is involved in.

Multi-Device (MDUCAC) mode enables the operator to restrict:

- the number of all sessions a served user is involved in
- the number of mobile sessions a served user is involved in
- the number of fixed sessions a served user is involved in
- the number of a served user's fixed devices engaged in calls

The User CAC service checks the user counts against the appropriate limits, as configured for the user.

In case of rejection of terminating communication when a limit is exceeded, the CAC services respond 486 Busy Here, which may be intercepted by other services, such as Communication Waiting and Communication Diversion.

In case of rejection of originating communication when a limit is exceeded, the CAC services optionally play an announcement, then respond 606 Not Acceptable.

2.35.2 Example Call Flow

2.35.2.1 Example Reject Terminating Communication

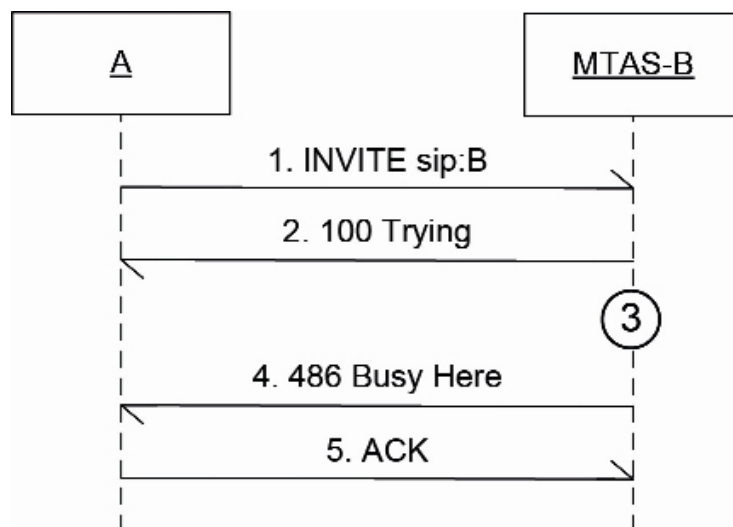


Figure 68 Reject terminating communication.

1. Caller (A) sends an INVITE request to B.
2. MTAS-B sends 100 Trying response.
3. User CAC determines that the new session would cause the served user to exceed the maximum number of active terminating sessions.
4. MTAS-B responds with 486 Busy Here.
5. User A acknowledges receipt of the final response.



2.35.2.2 Example Reject Originating Communication

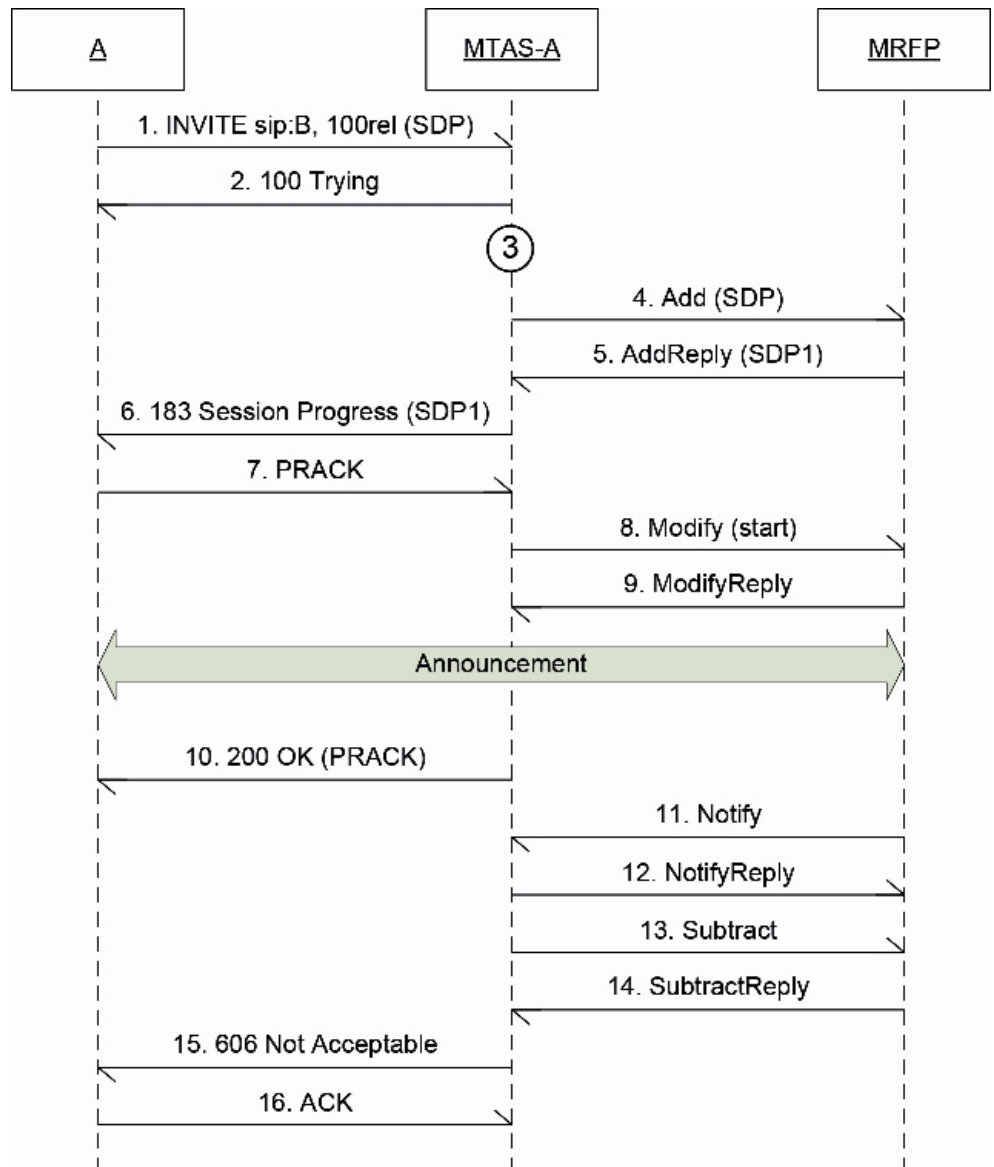


Figure 69 Reject originating communication.

1. Caller (A) sends an INVITE request to B.
2. MTAS-A sends 100 Trying response.
3. User CAC determines that the new session would cause the served user to exceed the maximum number of active originating sessions and determines that an announcement can be played.
4. User CAC issues an H.248 Add for a new termination.
5. User CAC receives the early media session SDP answer.
6. User CAC sends a 183 (Session Progress) that includes:
 - The Require header field with the option tag 100rel.
 - An answer to the SDP in the INVITE request.
 - A P-Early-Media header field set to "sendonly" (backward early media).



7. User A sends PRACK to MTAS-A
8. User CAC issues an H.248 Modify to start playing the announcement.
9. The MRFC replies.
At this point the announcement is being played to the end user
10. MTAS-A sends 200 OK (PRACK) to user A.
11. MRFP notifies CAC that it has finished playing the announcement.
12. MTAS-A replies to MRFP.
13. MTAS-A releases the MRFP resources.
14. MRFP replies to MTAS-A
15. MTAS-A responds to the initial INVITE with 606 Not Acceptable.
16. User A acknowledges receipt of the final response.

2.35.3 Service Interaction

2.35.3.1 Communication Waiting

The Communication Waiting service is dependent on the CAC service.

2.35.4 Configuration

- Activation (active/ disabled)
- Timer
- OCS Address and relation data
- Behavior for session when AoC cannot be provided

2.35.5 Performance Management

Counters for successful and unsuccessful –internal and external



2.36 Multiple Subscriber Number

2.36.1 Description

The Multiple Subscriber Number (MSN) feature allows additional numbers to be assigned to the served user. The MSN feature enables the served user to select the MSN number when making outgoing calls using a Supplementary Service Code and also enables setting different communication diversion and barring rules per MSN number. The selected MSN number is used for identity presentation and charging purposes. For example a user may use different numbers for private and for business calls or a family may use different numbers per family member.

On the originating MMTel AS the MSN service is replacing the identity of the caller with an identity, selected by the served user, also called MSN identity, when an outgoing communication is made from the served user. The served user can select the MSN identity from an operator defined list on a per call basis, using a Supplementary Service Code (SSC).

2.36.2 Example call flow

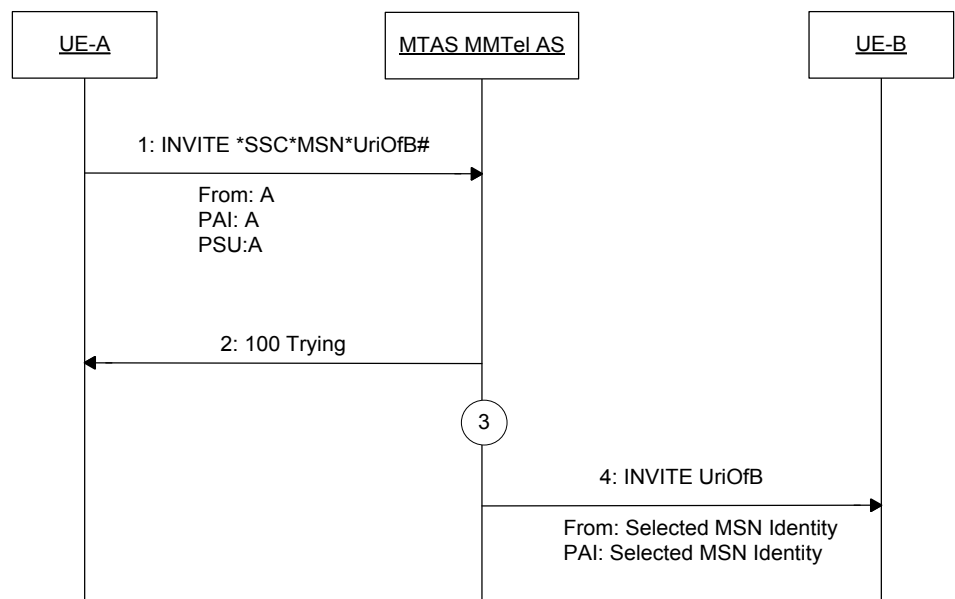


Figure 70 Example call flow for MSN service

- 1 UE-A sends an INVITE. The request URI contains the SSC. The SSC has two parameters: selected MSN identity and the dialed number. The INVITE message contains From and P-Asserted-Identity headers.
- 2 MTAS sends 100 Trying response to UE-A
- 3 MTAS executes the SSC service. User subscription to MSN service and selected MSN identity validity is checked, check is successful.



- 4 Call setup continues. Request-URI is set to the dialed Number, From and P-Asserted-Identity are replaced by the selected MSN identity

2.36.3 Charging

When Multi Subscriber Number is used, then on the originating side the selected MSN identity will be included in the charging messages as the calling party address. Indication of the usage of the MSN service will be also indicated.

Terminating side will treat the call as it were initiated by the selected MSN identity.

2.36.4 Service interaction

2.36.4.1 Identity presentation

Multi Subscriber Number is implemented as a sub-service of Flexible Identity Presentation. When MSN is used then MTAS replaces the user identity the same way as FIP service does. Hence all service interactions are the same as for FIP service.

2.36.4.2 Communication Diversion (CDIV)

The MSN feature enables the use of individual CDIV rules per MSN identity.

2.36.4.3 Communication Barring (CB)

The MSN feature enables the use of individual CB rules per MSN identity.

2.36.5 Provisioning

MTAS enables the operator to configure the service on user-level through the CAI3G interface. The user level configuration includes the pre-defined list of MSN identities and ID numbers (used in SSC to select the desired identity) assigned to them.

2.36.6 Configuration

A node level configuration parameter can be used to enabled or disable Multi Subscriber Number service.

2.36.7 Performance management

- Examples of performance counters related to the Multi Subscriber Number are: Number of successful invocations of MSN service
- Number of successful invocations of MSN service



2.37 Distinctive Ring

2.37.1 Description

The Distinctive Ring (DR) service enables a user to specify different ringtones for incoming calls per each of his/her IP Multimedia Public Identities (IMPU) which are part of the same Implicit Registration Set (IRS).

When the served user has an incoming call to one of its IMPUs, the terminating MMTel AS will add the appropriate Alert-Info header field to the INVITE request sent to the served user. In case the INVITE coming in to MMTel AS already has an Alert-Info header the DR service will overwrite this value.

The Alert-Info value may be used by the served user's terminal also when indicating a call is waiting.

The DR service enables the operator to define whether only operator pre-defined ringtones may be used or if the end user may freely specify their own ringtones.

2.38 Gateway Model

2.38.1 Description

The Gateway Model (GM) service allows MTAS to map events received from terminating User Agent (UA) or network – on one or more dialogs - towards the originating UA or the network on one single dialog. The service can be used when the originating UA or the network does not support multiple early dialogs.

The GM service can be used in static mode, where GM is applied to every call or in dynamic mode where the originating UA or the network indicates limited capabilities by including the “no-fork” directive in the Request-Disposition header of the initial INVITE.

As there is only one dialog between Originating UA or network and MTAS, events received on the terminating side have to be mapped to the single dialog on the incoming side. This means that when exchanging SIP events on the incoming side, the same From tag, To tag and Call ID is used; also when exchanging SDP offers/answers, the same session id is used (one in the Originating User → MTAS direction, and one in the MTAS → Originating User direction), and only the session version is incremented.

2.38.2 Example Call Flow

GM on both Originating and Terminating AS

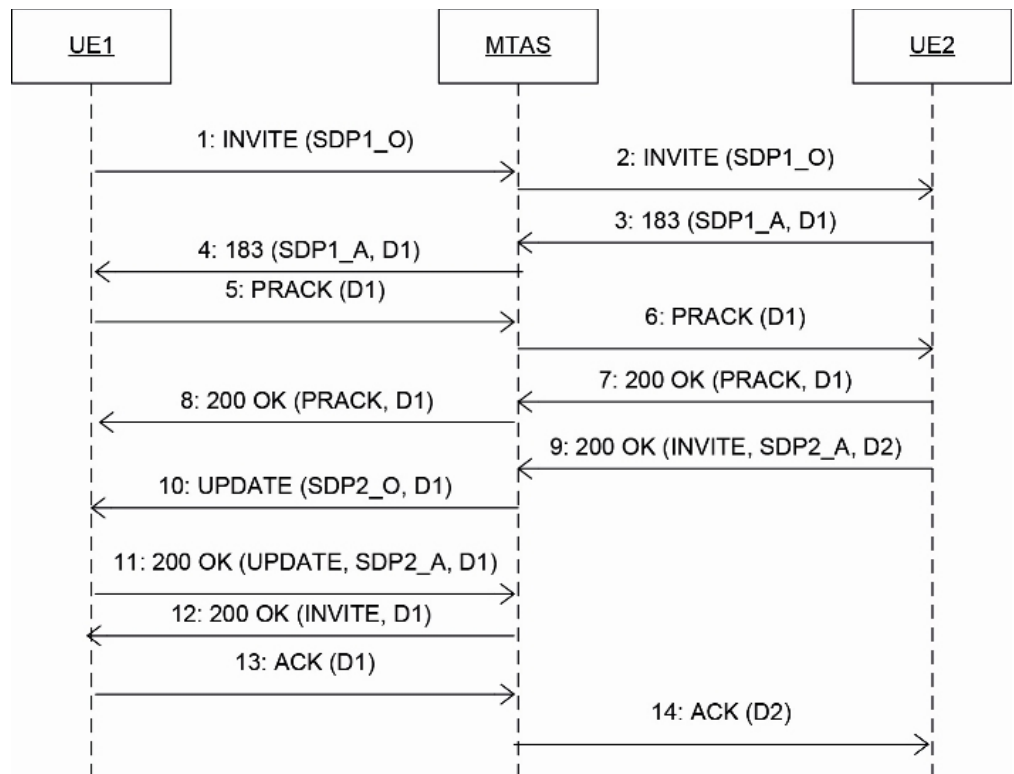


Figure 71 Main Scenario

- 1 Initial INVITE is received with an SDP offer from Originating User. The INVITE contains Require and/or Supported 100rel, Allow UPDATE, and does not contain Supported and/or Require precondition and precondition attributes. When GM is configured to dynamic GM the initial INVITE contains also the Request-Disposition header with the “no-fork” directive to trigger the GM service.
- 2 INVITE is sent to Terminating User.
- 3 Terminating User sends 183 Session Progress reliable provisional response with SDP answer.
- 4 183 Session Progress provisional response with SDP answer forwarded to Originating User.
- 5 PRACK is received.
- 6 PRACK forwarded.
- 7 200 OK (PRACK) received.
- 8 200 OK (PRACK) forwarded. First SDP negotiation completed.
- 9 Terminating User sends 200 OK for the INVITE with SDP answer.
- 10 UPDATE is sent to the Originating User with SDP offer.



- 11 200 OK for the UPDATE is received from the Originating User containing the SDP answer for the SDP offer sent in step 10. The SDP answer contains same IP address, port and codec that was received in the SDP offer in the initial INVITE in step 1.
- 12 200 OK for the INVITE is sent to Originating User.
- 13 ACK received from Originating User.
- 14 ACK sent to Terminating User.

2.38.3 Service Interaction

2.38.4 Configuration

On/off using License.

Administrative state

Mode (Orig, Term, OrigTerm and Dynamic)

2.39 Flexible Service Format Selection

2.39.1 Description

The Flexible Service Format Selection (FSFS) service makes it possible for S-CSCF or other application servers before MTAS in the ISC chain to influence the set of services to be executed in MTAS for a specific session.

2.39.2 Example call flow

In case of FSFS there are six different user roles:

- O&M Operator
- Other AS
- S-CSCF
- Caller
- Called Party
- Charging Server

O&M - The FSFS service of the MTAS can be configured and modified by Solution Integration.

The other AS before MTAS in the ICS chain can select different FSFS service format by choosing different parameter settings in the INVITE.

The S-CSCF before MTAS can trigger the Remote UE is the called party of the session.



The Caller's configured services are suppressed by the FSFS service in the outgoing communication session.

The Called Party's configured services are suppressed by the FSFS service in the incoming communication session.

This scenario covers the case, where the new session uses a specified FSFS service format. The INVITE message is not modified by the FSFS service in the incoming communication session. The Called Party may be registered or unregistered subscriber.

The Charging Server receives the FSFS service related information conveyed by MTAS. In the offline charging case the charging server is the Charging Data Function (CDF), where in the online charging case the charging server is Online Charging System (OCS).

2.39.2.1 Suppressing Terminating MTAS Services

In this scenario, B-party is the served user. In this case, the FSFS service suppresses all types of Communication Barring, Communication Forwarding Unconditional, and the Flexible Communication Distribution services, following the receipt of SIP INVITE containing *Route* header with *EmergencyCallback* parameter in the terminating MTAS. As the result, the incoming communication attempt is not barred, not diverted and not distributed to other targets, but forwarded to the B-party. MTAS sends the FSFS related charging report towards the charging server.

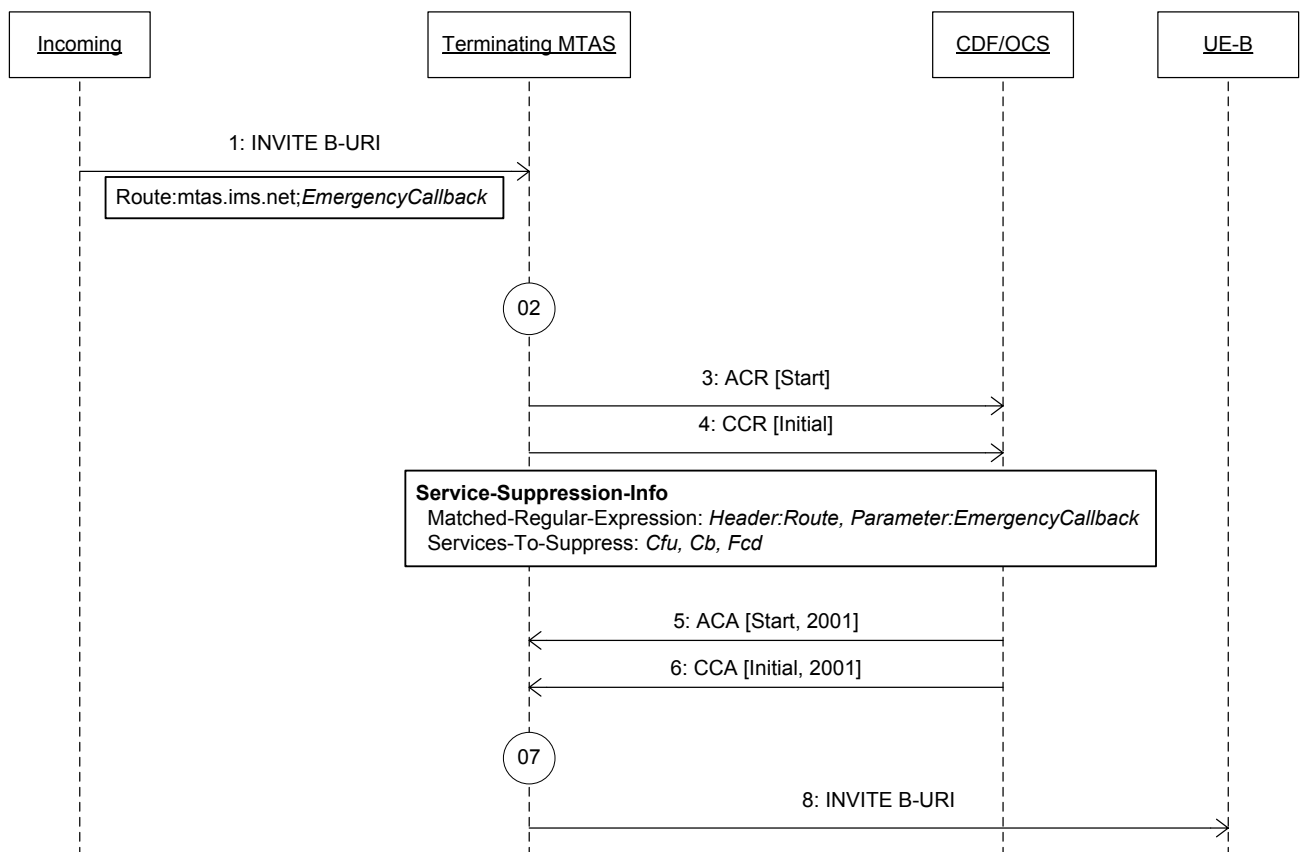


Figure 72 – The FSFS service suppresses the CFU, CB, and FCD services in the Terminating MTAS

- 1 SIP INVITE containing *Route* header with *EmergencyCallback* parameter is received by Terminating MTAS.
- 2 After the header-parameter evaluation, the FSFS service determines that the SIP INVITE contains header and parameter in pair that are matched with the values of configured attributes. Based on the configuration, the FSFS service attempts to suppress CFU, CB, and FCD services.
- 3 If offline charging is active, then MTAS sends ACR Start request towards the CDF. The ACR request contains Service-Suppression-Info AVP (group), Matched-Regular-Expression AVP set to *Header:Route*, *Parameter:EmergencyCallback*, and Services-To-Suppress AVP set to *Cfu, Cb, Fcd*.
- 4 If online charging is active, then MTAS sends CCR Initial request towards the OCS. The CCR request contains Service-Suppression-Info AVP (group), Matched-Regular-Expression AVP set to *Header:Route*, *Parameter:EmergencyCallback*, and Services-To-Suppress AVP set to *Cfu, Cb, Fcd*.
- 5 MTAS receives a response to the ACR request with successful indicator (code 2001).



- 6 MTAS receives a response to the CCR request with successful indicator (code 2001).
- 7 MTAS suppresses communication barring attempt, communication diversion attempt by CFU, and communication distribution attempt by FCD, as if those services were deactivated.
- 8 The SIP INVITE is forwarded towards B-party (served user). Note: The further steps (e.g. 200 OK responses, ACK responses) are not explained to simplify the process.

2.39.2.2 Suppressing Originating MTAS Services

In this scenario, A-party is the served user. In this case, the FSFS service suppresses Outgoing Communication Barring (OCB) service following the receipt of SIP INVITE containing *P-Asserted-Identity* header with *authority* parameter in the originating MTAS. As the result, the outgoing communication attempt is not barred. MTAS sends the FSFS related charging report towards the charging server.

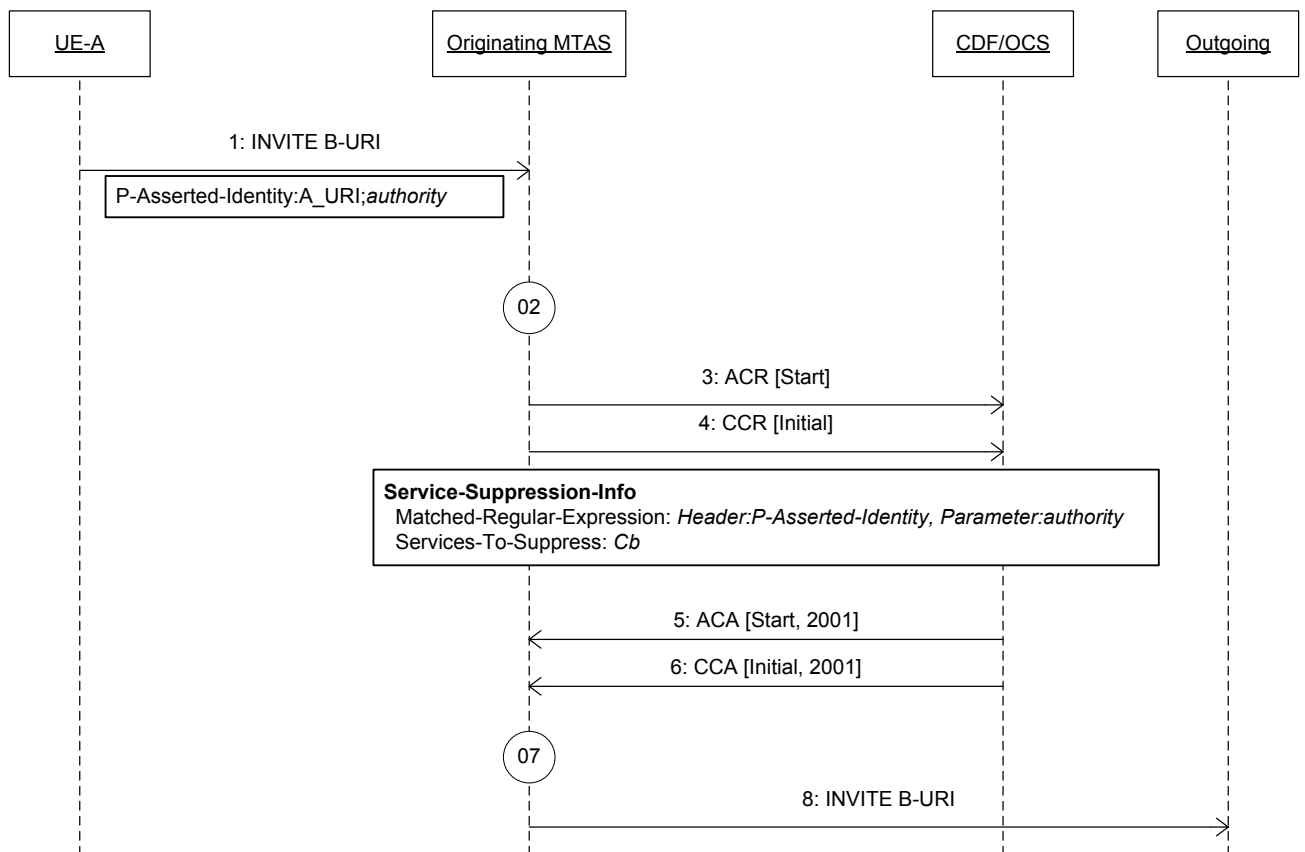


Figure 73 - The FSFS service suppresses the OCB service in the Originating MTAS

1. SIP INVITE containing *P-Asserted-Identity* header with *authority* parameter is received by Originating MTAS.



2. After the header-parameter evaluation, the FSFS service determines that the SIP INVITE contains header and parameter in pair that are matched with the values of configured attributes. Based on the configuration, the FSFS service attempts to suppress the OCB service.
3. If offline charging is active, then MTAS sends ACR Start request towards the CDF. The ACR request contains Service-Suppression-Info AVP (group), Matched-Regular-Expression AVP set to `Header:P-Asserted-Identity, Parameter:authority`, and Services-To-Suppress AVP set to `Cb`.
4. If online charging is active, then MTAS sends CCR Initial request towards the OCS. The CCR request contains Service-Suppression-Info AVP (group), Matched-Regular-Expression AVP set to `Header:P-Asserted-Identity, Parameter:authority`, and Services-To-Suppress AVP set to `Cb`.
5. MTAS receives a response to the ACR request with successful indicator (code 2001).
6. MTAS receives a response to the CCR request with successful indicator (code 2001).
7. MTAS suppresses outgoing communication barring attempt as if that service was deactivated.
8. The SIP INVITE is forwarded towards B-party (served user). Note: The further steps (e.g. 200 OK responses, ACK responses) are not explained to simplify the process.

2.39.3 Service interactions

The FSFS service interacts with the following services.

2.39.3.1 Call Admission Control (CAC)

The FSFS service if configured can suppress any type of CAC services in particular session. When suppressed, the communication restriction is bypassed as if the service was not activated.

2.39.3.2 Communication Barring (CB)

The FSFS service if configured can suppress any type of CB services in particular session. When suppressed, the communication barring attempt is bypassed as if the service was not activated.

2.39.3.3 Communication Diversion (CDIV)

The FSFS service if configured can suppress any type of CDIV services in particular session. When suppressed, the communication diversion attempt is bypassed as if the service was not activated.



2.39.3.4 Customized Alerting Tones (CAT)

The FSFS service if configured can suppress the CAT service in particular session. When suppressed, the caller will be provided with the regular ringing as if the CAT service was not activated.

2.39.3.5 Explicit Communication Transfer (ECT)

The FSFS service if configured can suppress the ECT service in particular session. If at least one of the established legs is suppressed, the triggering of ECT service is rejected with SIP 403 Forbidden.

2.39.3.6 Flexible Communication Distribution (FCD)

The FSFS service if configured can suppress the FCD service in particular session. When suppressed, the communication distribution attempt is bypassed as if the service was not activated.

2.39.3.7 Session Transfer to Own Device (STOD)

The FSFS service if configured can suppress the STOD service in particular session. When suppressed, the session establishment attempt towards target devices and/or related users is handled as if the service is not enabled.

2.39.3.8 3PTY

The FSFS service if configured can suppress the 3PTY service in particular session. If at least one of the established legs is suppressed, the triggering of 3PTY service is rejected with SIP 403 Forbidden.

2.39.4 Configuration

Some configurable options as example are:

- Enable/disable the FSFS service
- Generate list of URI addresses dedicated for Voice Mail servers
- Specify the session case the pattern is applied
- Specify the headers of the incoming INVITE the pattern is applied
- Specify the regular expression of the headers specified by other attribute the pattern is applied
- Set indicator whether or not to remove the matched parameter from the INVITE message
- Specify the algorithm used by the FSFS to evaluate the header that appears more than once in the SIP INVITE, or to evaluate the header that has multiple values



- Configure groups of suppressed services
- Configure services to suppress in a group specified by other attribute

2.39.5 Performance management

The following performance counters are provided by MTAS to evaluate the usage and quality of service:

- Number of received INVITE matched with configured attributes that triggers the service suppression

2.39.6 Fault management

For information on the alarm, refer alarm OPI [65] .

2.40 Video Fallback to Audio

2.40.1 Description

The Video Fallback to Audio service improves the user experience if the originating multimedia User Equipment (UE) calls a destination network that does not support media capability negotiation based on SDP offer/answer protocol. These networks may drop the call with a SIP error response if the media capabilities requested in the SDP offer cannot be satisfied. One example of such behavior is when video call is placed to a terminal connected to a circuit-switched network. In this case if the destination terminal does not support video calling capability or the circuit-switched network has no capacity for video call, the circuit-switched network drops the call instead of falling back to an audio-only call. Video Fallback to Audio service detects such a failed call and repeats the call as an audio-only call. The caller is not aware of the fallback mechanism; he will see only a successful audio-only call.



2.40.2

Example call flow

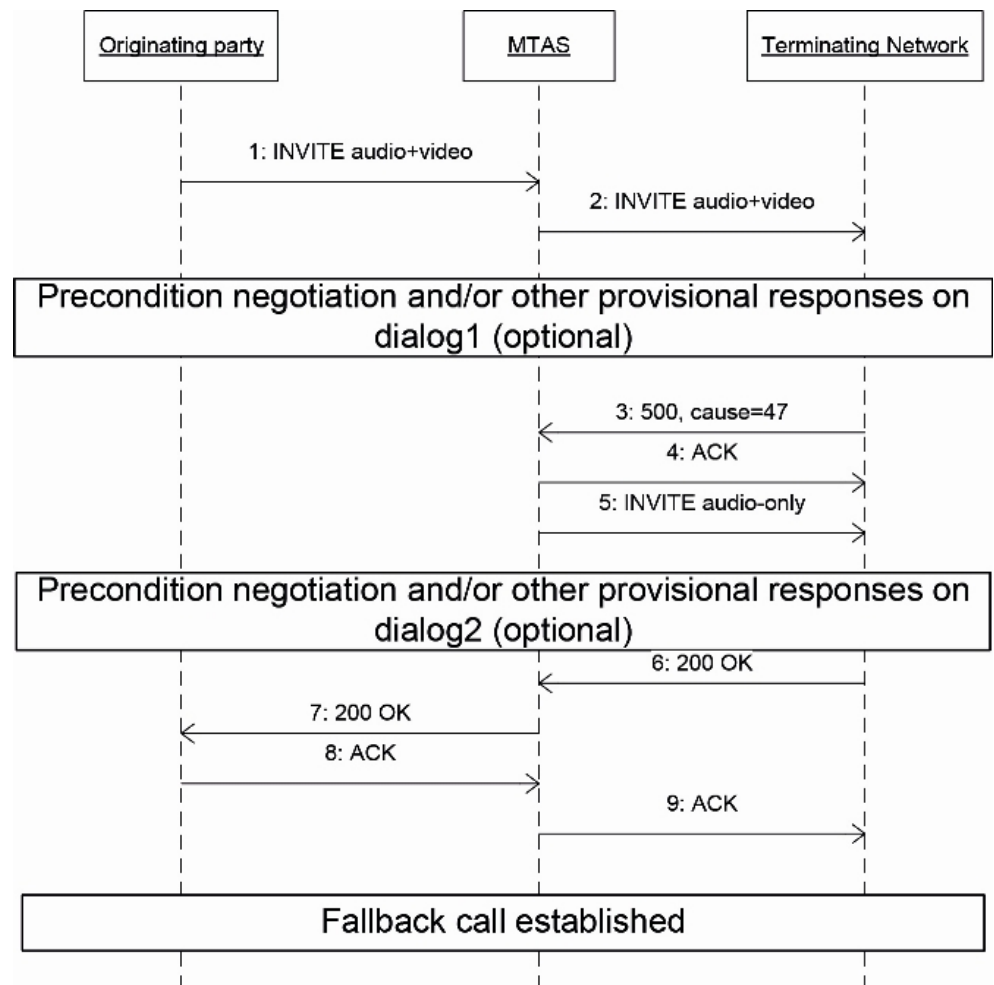


Figure 74 - Successful fallback call

1. Fallback call function starts when the Originating Party sends an INVITE request with audio and non-audio (e.g. video) media descriptions to MTAS.
2. This INVITE is sent to the Terminating Party. Optional precondition negotiation is accomplished between the Originating and Terminating Parties.
3. Terminating Network on behalf of Terminating Party responds with a 500 SIP error response, containing a Q.850 cause code that matches one of the configured cause codes.
4. The SIP error response is acknowledged.
5. The INVITE is repeated where non audio media are not active (ports are set to 0 in SDP).
6. The fallback call is established.
7. MTAS forwards the 200 OK response to the Originating Party.
8. Originating Party acknowledges the call establishment.



9. MTAS forwards the ACK request to the Terminating Network.

2.40.3 Configuration

- The Q.850 cause code list that causes the fallback function to activate.

2.40.4 Performance management

The following performance counters are provided by MTAS.

- A counter that is incremented when Video Fallback to Audio function issues a fallback call.
- A counter that is incremented when a fallback call issued by Video Fallback to Audio function is successfully established.

2.40.5 Fault Management

For information on the alarm, refer alarm OPI [66] .

2.41 Bandwidth Optimization

2.41.1 Description

Bandwidth Optimization is a service intended to eliminate unnecessary bandwidth reservation when the subscriber has bandwidth limitation on his/her access.

In certain use cases bandwidth is reserved for call legs where no communication is likely to occur. It causes problems when a subscriber has limited bandwidth and due to the bandwidth reserved for the above mentioned legs communication is rejected.

Note that Bandwidth Optimization is only applicable in networks where the functionality of the MTAS and the P-CSCF are aligned.

- When the RTP bandwidth (AS) for both Session Level and Media Level is returned by an UE, the MTAS only updates the Media Level bandwidth for optimization, the P-CSCF must handle both Session Level and Media Level updates.
- The P-CSCF must support the RTCP RR and RS Bandwidth AVP in the Rx interface to the PCRF.

2.41.2 Example call flow

In case of BW Optimization there are two user roles:

- Caller
- Called party



Caller - The subscriber that initiates the call, also referred to as subscriber A

Called-party - The subscriber that receives the call, also referred to as subscriber B

Hold in originating MTAS, RS and RR present

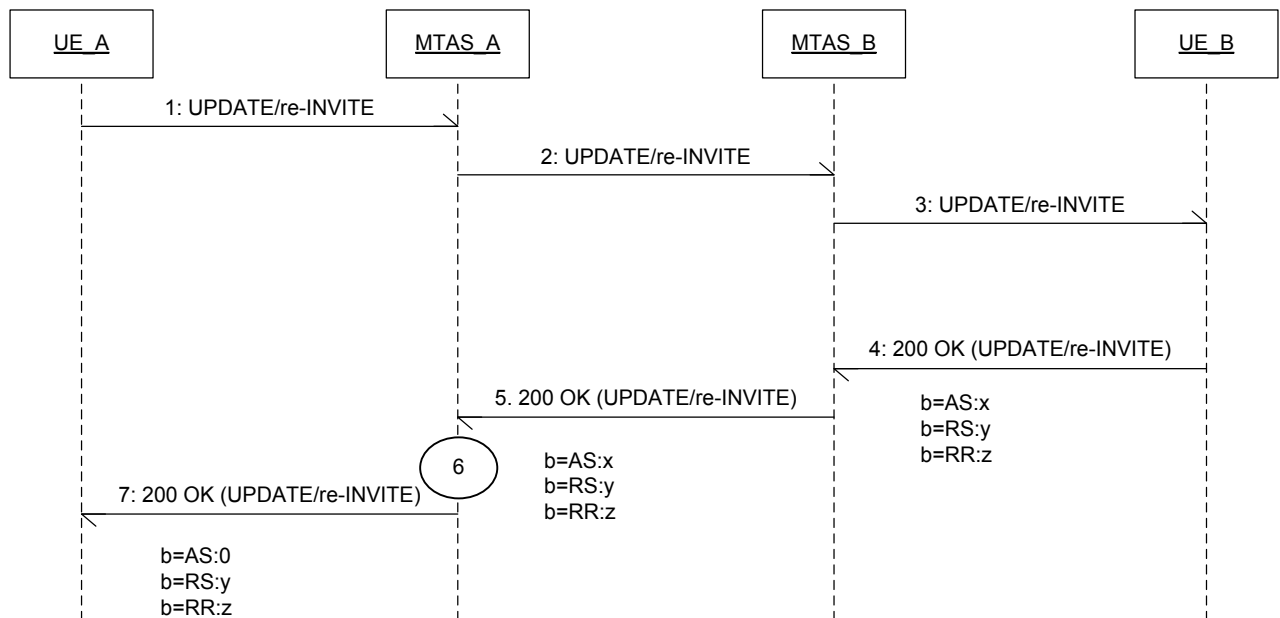


Figure 75 Hold in originating MTAS, RS and RR present

- 1 UE_A sends an UPDATE/re-INVITE with an SDP offer in order to put UE_B on hold, media direction change can either be sendrecv -> sendonly or recvonly -> inactive
- 2 MTAS_A forwards the UPDATE/re-INVITE to MTAS_B.
- 3 MTAS_B forwards the UPDATE/re-INVITE to UE_B.
- 4 UE_B responds with 200 OK (UPDATE/re-INVITE) including an SDP answer. The SDP answer contains parameter "b=AS..." with information about the actual RTP bandwidth. It also contains parameter "b=RS..." and "b=RR..." with information about the RTCP bandwidth.
- 5 MTAS_B forwards the 200 OK (UPDATE/re-INVITE) to MTAS_A.
- 6 Hold service in MTAS_A checks CM parameter `mtasHoldBandwidthOptimizationMode`. If it is set to 1 (bandwidth optimization is enabled) then as "b=AS...", "b=RS..." and "b=RR..." are present, it does the following:
 - leaves "b=RS" and "b=RR" as they are
 - sets "b=AS" to 0
- 7 200 OK (UPDATE/re-INVITE) is sent to UE_A



If Hold was initiated with a re-INVITE, an ACK is sent back from UE_A towards UE_B

Communication Waiting triggered by UDUB, status code 486 with warning code 370 received

Preconditions:

- UE_B is engaged in a call with another subscriber through a Basic call, 3PTY or Conference (not shown on the picture)
- Communication waiting mode can be either Normal Mode, Alternate Mode 1 or Alternate Mode 2

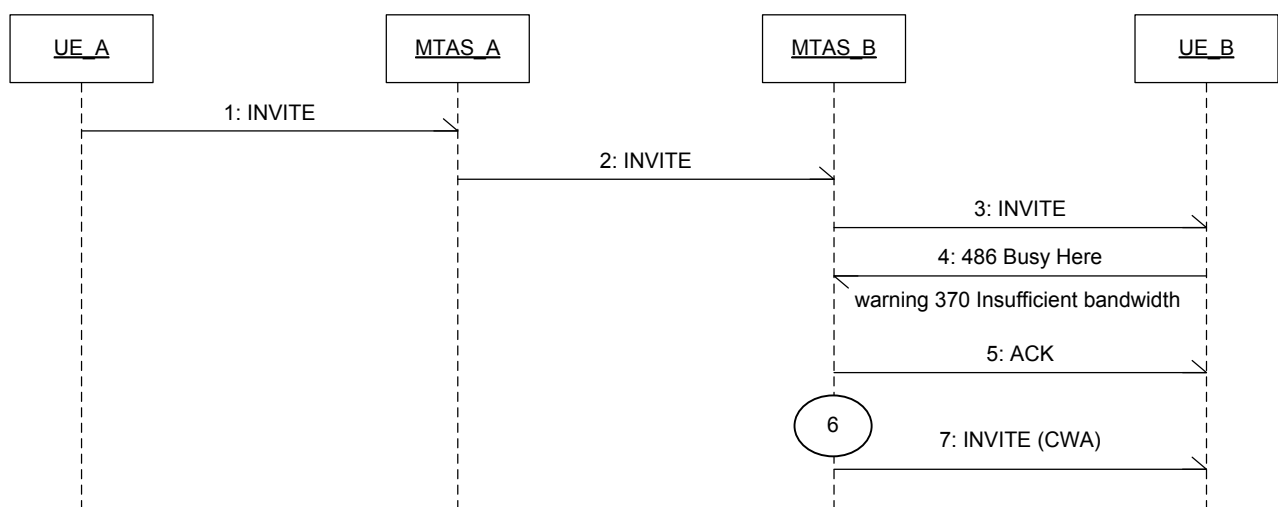


Figure 76 Communication Waiting triggered by UDUB, status code 486 with warning code 370 received

- 1 UE_A sends an INVITE.
- 2 MTAS_A forwards the INVITE to MTAS_B.
- 3 The CW function in MTAS_B detects that UE_B has an active CW subscription. The INVITE is forwarded to UE_B. Store information that INVITE was sent without Communication Waiting Active indication.
- 4 UE_B responds with status code 486 Busy Here including warning code 370 Insufficient bandwidth.
- 5 MTAS_B sends ACK.
- 6 The CW function in MTAS_B finds that status code 486 Busy Here including warning code 370 Insufficient bandwidth was received from UE_B and it also checks that INVITE in step 3 was sent without Communication Waiting Active indication.
- 7 MTAS_B sends a new INVITE to UE_B with Communication Waiting Active indication



2.41.3 Service interactions

2.41.3.1 Hold and CW

In certain Hold and Communication Waiting use cases bandwidth is reserved for call legs where no communication is likely to occur. Such cases are the leg between A and MTAS_HOLD after A put B on hold or between MTAS_CW and B when A tries to call B who is engaged in a call with C.

It causes problems when a subscriber (like A in the hold case or B in the communication waiting case) has limited bandwidth and due to the bandwidth reserved for the above mentioned legs communication is rejected.

2.41.3.2 3PTY and Conference

Other services affected are 3PTY and Conference. When a 3PTY or Conference dialog is put on hold by any of the participants.

2.41.4 Configuration

Node level configuration parameters:

- Bandwidth optimization for Hold (enabled or not)

2.42 Priority Call

The Priority Call feature enables prioritization of communication sessions. MTAS supports two different mechanisms:

- *Priority Services* is a multi-node feature that ensures priority handling of communication throughout nodes. Eligibility for priority is granted by other node (CSCF/SBG). MTAS handles prioritization according to indication in incoming SIP messages. MTAS gives priority treatment and exemption from certain restriction policies for such prioritized communication sessions. Priority Services is a standard mechanism that uses five different priority levels, indicated in the Resource-Priority header, a SIP extension.
- *GETS Priority Service* is an extension to above priority services where the priority can be granted either by the other node (P-CSCF) or by a GETS server. In addition to the presence of Resource-Priority header, the type of the called number is also factored in towards call prioritization.
- The *Priority Call indication* mechanism indicates in outgoing originating requests towards PSTN that the caller user had been provisioned with priority. It is a proprietary mechanism that uses the standard SIP Priority header.



2.42.1 Priority Services

Priority Services concerns prioritization of resources when systems are overloaded, e.g. in disaster situations when damaged network elements and increased number of calls led to network congestion. It ensures that network provides resources for reliable communications of eligible government officials (e.g. rescue forces), even when normal calls cannot be served due to high load.

Priority Services is a multi-node feature, in which MTAS' priority handling is triggered by receipt of "Resource-Priority" SIP header (RPH), a SIP extension defined in [51], of the initial INVITE.

MTAS handles resource priority as following:

- RPH identification and validation: MTAS understands two namespaces, namely ets and wps, within RPH. The ets namespace is not used beyond validation, where either of ets.0 or ets.1 should be present for MTAS to process the RPH. The wps namespace determines the level of resource priority, with wps.0 indicating the highest and wps.4 the lowest priority.
- Priority treatment: The wps namespace value is mapped to a parameter for TSP load regulation permission request. If TSP load regulation response does not allow processing a request with given priority, then it will be rejected by MTAS. In an overload situation, higher priority calls should still pass when lower priority calls are rejected.
- Priority propagation:
 - MTAS forwards RPH when proxying SIP messages.
 - For priority handling purposes, MTAS maps session priority to MRFC via Mp or Mr' interface.
 - As information about resource priority of the IMS multimedia session, MTAS maps the priority value towards charging systems via Ro and Rf interface.
- MTAS protects resource priority calls from communication failures with online or offline charging systems.

2.42.1.1 Priority treatment

The Priority Services feature influences the selection of calls to be rejected by TSP load regulation, when load increases above the engineered capacity.

It ensures that no priority calls will be rejected as long as any normal calls could be accepted. Similarly, all higher priority calls are accepted until any lower priority calls should have been rejected.



2.42.1.2 MRFC communication

2.42.1.2.1 Priority mapping via Mr' (SIP) interface

In SIP messages on Mr' interface, related to a resource priority call session, the copy of original RPH (by which the priority call had been identified) is included.

2.42.1.2.2 Priority mapping via Mp (H.248) interface

In H.248 messages on Mp interface, related to a resource priority call session, the Priority context attribute contains a value mapped from wps of original RPH as follows:

RPH wps attribute	H.248 Priority attribute
wps.0	15
wps.1	14
wps.2	13
wps.3	12
wps.4	11

2.42.1.3 Charging

2.42.1.3.1 Priority information mapping

In charging messages towards charging systems (both on- and offline), related to a resource priority call session, the IMS-Information group AVP contains a Session-Priority AVP with value mapped from wps of original RPH as follows:

RPH wps attribute	Session-Priority AVP
wps.0	Priority-0
wps.1	Priority-1
wps.2	Priority-2
wps.3	Priority-3
wps.4	Priority-4

2.42.1.3.2 Charging system communication failure handling

For resource priority calls, Online Charging function does not apply Credit Control Failure Handling by the charging profile, if communication failures occur to Online Charging System. Instead, it applies the 'Continue' (without further credit control) action.

Similarly, the Offline Charging function shall allow resource priority calls to continue "free of charge" when the charging function has failed to send or backup an ACR, regardless service configuration.



2.42.1.4 Service Interaction

2.42.1.4.1 Ad-hoc conference

Resource priority setting of the conference creator will be in effect on all conference legs.

2.42.1.4.2 Communication Diversion

All forwarded calls inherit resource priority setting of the original call.

2.42.1.5 Configuration

- On/off on node level

2.42.2 Priority Call indication

The Priority Call indication feature assigns a priority indication in the outgoing originating requests. The intention is to set a priority user indication to be used towards the PSTN (protocol mapped in Media Gateway Control Function, external to MTAS).

2.42.2.1 Example Call flow

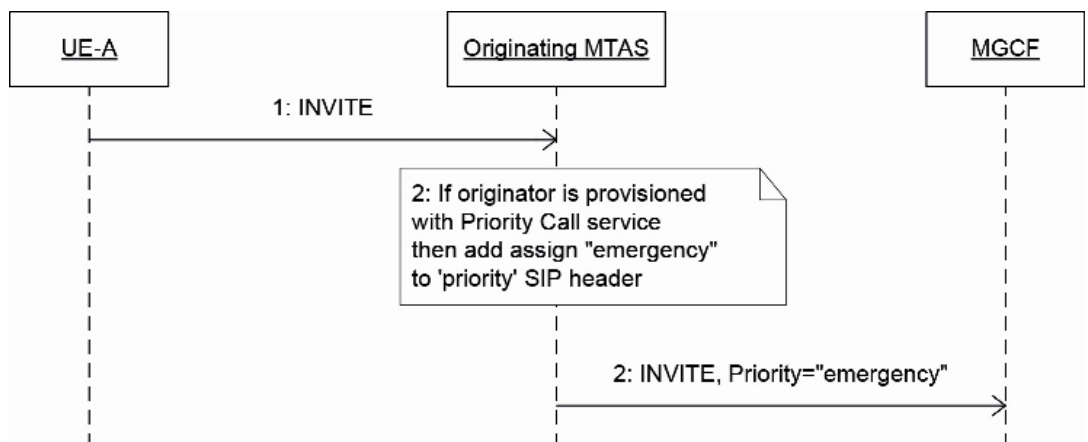


Figure 77 - Priority Call originates an IMS-to-PSTN session.

- 1 Originating MTAS receives an INVITE
- 2 Originating MTAS checks if user is provisioned with the priority-call Service
- 3 If the SIP Priority Header is not already populated, MTAS populates the SIP Priority Header with a value of "emergency" and sends the INVITE onwards



2.42.2.2 Service Interaction

2.42.2.2.1 Conference

The priority call indication is propagated on the conference outgoing legs to all the conference participants

2.42.2.2.2 Communication Diversion

The priority call indication is propagated on outgoing leg to the new target

2.42.2.2.3 Communication Completion

The priority call indication is included in the recall.

2.42.2.3 Configuration

- On/off on node level
- Provisioned to the user as operator domain subscription data

2.42.3 GETS Priority Service

GETS Priority service enables MMTel AS to support the following invocation methods which are as outlined in the GIR for NGN Priority Services. It is an enhancement over the existing Priority Service feature.

This enhancement allows either P-CSCF or a GETS-AS to act as authorization point for the GETS priority service call. MMTel AS inspects the dialed number in order to identify the type of GETS priority call.

The various identified GETS priority call types are as below.

Call type	Dialed number type	Authorization entity	Applicable AS
GETS-FC	Operator configured feature code prefix	Subscription based authorization by P-CSCF	Originating AS
GETS-NT	Operator configured GETS-NT number	PIN based authorization by GETS-AS	Originating AS. Also, terminating AS in case of call diversion when originating AS chaining is disabled
GETS-AN	Operator configured GETS-AN number	PIN based authorization by GETS-AS	Originating AS. Also, terminating AS in case of call



			diversion case when originating AS chaining is disabled
GETS-FC + GETS-AN	Operator configured feature code prefix followed by operator configured GETS-AN number	PIN based authorization by GETS-AS	Originating
GETS-FC + GETS-NT	Operator configured feature code prefix followed by operator configured GETS-NT number	PIN based authorization by GETS-AS	Originating

2.42.3.1 Identifying GETS-FC priority service call

Originating MMTel AS identifies the GETS-FC priority service call type on reception of incoming INVITE.

A feature code (eg: *272) configured as supplementary service code is used to identify GETS-FC priority service call. The aforementioned feature code is matched with the prefix of Req-URI and in case of a match the call is identified as GETS-FC type.

Next, MMTel AS will proceed to remove the GETS-FC feature code from Request URI and proceed with number normalization.

2.42.3.2 Identifying GETS-AN or GETS-NT priority service call

On the reception of incoming INVITE, the originating MMTel AS identifies the GETS-AN or GETS-NT priority service call. This is done by looking up the called party number from Req-URI amongst the list of configured AN/NT numbers for a match. If binary model is enabled, the forward to DN number present in the binary data model will get matched.

List of AN and NT numbers are configured as NSN/OSN numbers in MMTel AS so that the Req-URI is not normalized. This is to avoid a failure when comparing the number in R-URI with list of configured numbers.



Terminating MMTel AS also identifies and process GETS-AN/GETS-NT call as mentioned above when the call is diverted by terminating MTAS when originating AS chaining is disabled.

In case of GETS-AN and GETS-NT call, the wps value in the RPH header as inserted by the GETS-AS in the 200 OK (INVITE) will be considered as the priority of the call.

2.42.3.3 Identifying GETS-FC+AN and GETS-FC+NT priority service call

2.42.3.4 First, the originating MMTel AS will identify the GETS-FC priority service call type on reception of incoming INVITE. See earlier section 'Identifying GETS-FC priority service call' mentioned above.

If call is identified as GETS-FC, further, MMTel AS will proceed to try further to identify the call type as GETS-FC+AN or GETS-FC+NT. For this, as the next step, the same call is attempted to be matched as GETS-AN or GETS-NT type.

This subsequent procedure is same as identification of call as GETS-AN/GETS-NT type. Please see earlier section 'Identifying GETS-AN or GETS-NT priority service call'.

2.42.3.5 Service interactions

2.42.3.5.1 Charging

Interaction is related to charging suppression for priority calls and the impact on AVPs.

a) Charging suppression for the session:

MMTel AS will have functionality to bypass originating and terminating charging based on the identified GETS Priority service call type. The suppression functionality is configurable for the different GETS priority service call types and can be used to suppress online charging, offline charging or both.

For charging suppression, flavors of GETS-FC call like GETS-FC + GETS-AN and GETS-FC + GETS-NT are considered as GETS-FC itself.

In terminating MMTel AS, it is possible to suppress specific charging for priority service call. Charging Suppression will be based on configuration. If RPH header is present then the call type is identified as GETS-TERM and online charging, offline charging or both can be suppressed as per the configuration.

b) AVP usage:

Usage of AVPs in Diameter messages are as below:



ACR Start	Updated wps information reported in in Session-Priority AVP	
	Ericsson SSI AVPs:	
	Supplementary-Service-Identity	To report type of GETS priority service call
	Service-Action	To report usage of GETS Priority service call
CCR Initial	<ul style="list-style-type: none"> • Reports the initial wps value in the Session-Priority AVP • Also, have the two Ericsson SSI AVPs mentioned above in ACR Start 	
CCR Update	Reports any updated wps value in Session-Priority AVP	
CCR Terminate	Reports any updated wps value in Session-Priority AVP	
ACR Stop	Reports any updated wps value in Session-Priority AVP	

c) AVP suppression:

ANI-AVP and Called-Party AVPs are suppressed for GETS-NT priority service call.

2.42.3.6 OCT

A GET-FC priority call from A to Operator transferor (OT) can be transferred via OCT. Charging suppression and Charging AVPs apply for the usual GETS-FC type call.

OCT service will make the refer-to user call using the number in 'Refer-To' header of REFER message from the A to OT call. The call to refer-to user will inherit the priority service call type of A to OT call. Charging suppression and Charging AVPs apply for the GETS-FC call.

2.42.3.7 Call diversion

For a SIP INVITE with RPH header reaching terminating MMTel AS, call diversion can be executed in terminating MMTel AS which will allow diversion to a provisioned Forward-To DN GETS-AN or GETS-NT number.



Attempt to divert call to GETS-FC numbers is blocked by terminating MMTel AS.

The wps value for the outgoing INVITE is configurable and MMTel AS reads the configured value to include it in outgoing INVITE to the diverted party.

For a call diverted to GETS-AS or GETS-NT number, the GETS-AS can authorize the call and set a new priority value for the call by updating the wps value in 200 OK of INVITE. In this case, MMTel AS updates the stored wps value and uses it for further communication on diverted call leg.

2.43 Calling Party Category

2.43.1 Description

Calling Party Category (CPC) service allows the operator to assign CPC values to users. When the user initiates a call, the originating MTAS adds the user's CPC value to the P-Asserted-Identity header. When the served user causes the call to become transit, the CPC value is added to the diverting user's URI in the History-Info header.

CPC value for the user is stored in the operator part of the user's transparent service data and can be provisioned by the operator using CAI3G.

The charging messages generated by MTAS include the CPC value.



2.43.2 Example Call Flow

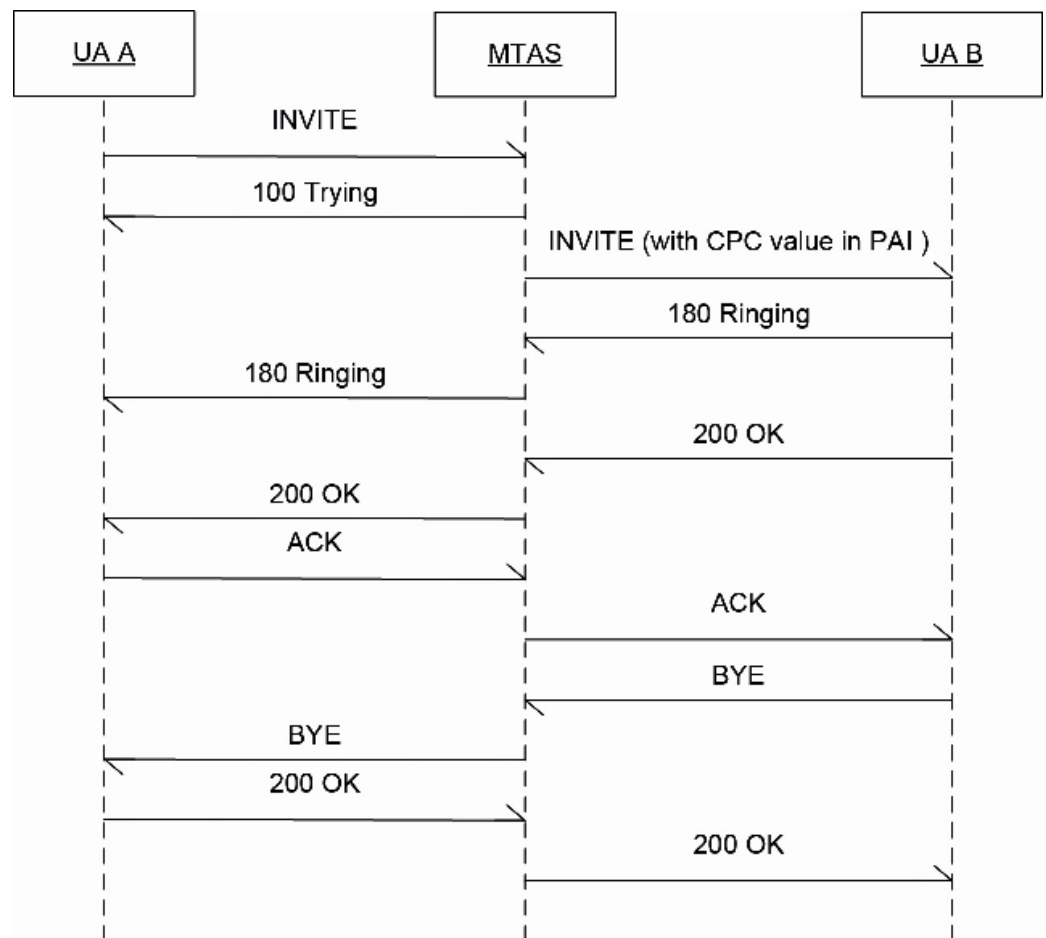


Figure 78 – Calling Party Category

2.43.3 Service Interactions

2.43.3.1 Charging

If the Calling Party Category (CPC) service adds CPC parameter to the P-Asserted-Identity in the initial INVITE, the charging message generated by MTAS will also contain the CPC parameter in the Calling-Party-Address AVP. If the CPC service adds CPC parameter to the served user's History-Info entry, the charging message generated by MTAS will also contain the CPC parameter in the Redirecting-Party-Address AVP. In both cases, Supplementary-Service-Identity AVP with a service code allocated for CPC service will be included into the charging message.

2.43.3.2 ICB

CPC parameter is taken into account when evaluating Global ICB Black List rules. These rules are strings that are matched as substrings against the entire P-Asserted-Identity header value. If any part of the CPC parameter in the P-Asserted-Identity header value matches a Global ICB Black List entry, the call is barred.



CPC parameter is not copied into the Dynamic Black List when the user activates this function and is not taken into account when the incoming call is evaluated against the Dynamic Black List.

2.43.3.3 CCxx

If the calling user with CPC service configured activates CCxx, then the CPC parameter will be added to the P-Asserted-Identity header in the INVITE placed by CCxx after the called user becomes available. The CPC parameter shall be present only in the INVITE request sent to the Terminating Party.

2.43.3.4 Conference

If the Conference Creator is configured with CPC value, the CPC parameter will be present in the INVITE placed by the Conference Focus.

2.43.3.5 Abbreviated dialing

If the calling user with CPC configured also has Abbreviated dialing configured, then the CPC parameter will be present in the INVITE placed by the Abbreviated dialing service.

2.43.3.6 Application Server interworking

If the called user has both CPC and AS Interworking activated and the AS Interworking service creates History-Info entries, CPC parameter of the called user is added to the History-Info entry that belongs to the called user.

2.43.4 Configurations

The Calling Party Category service requires the following Operator Subscription Level information:

- CPC Active/Inactive
- CPC value

2.43.5 Performance Management

The following performance counters are provided by MTAS:

- MtasCpcAdded – successful CPC value addition
- MtasCpcOverridden – successful CPC value overwrite
- MtasCpcRemoved - successful CPC value removal

2.43.6 Fault Management

For information on the alarm, refer alarm OPI [67] .



2.44 Network Announcement

2.44.1 Description

The Network Announcement service offers a possibility to an operator to play an announcement to a calling user when fault situation occur during an MMTel session establishment for example congestion or wrong number. It is possible to configure if an announcement shall be played depending on the received SIP error status code and which announcement shall be played.

The announcement can be a fixed audio, video or audio-video announcement.

The originating MTAS offers also a possibility to a UA to explicitly request an audio announcement to be played to a user by sending an INVITE with a Request URI that is a preconfigured string identifying the required announcement.

The originating MTAS also offers the Network Message function. When Network Message function is enabled and MMTel AS generates or receives a non-200 OK final response on initial INVITE, the content of the error response can be updated before is it sent to the calling user. When a MtasNaNm instance with the key of the warning text exist then a Reason header is added/updated and the status line is updated according to the configuration. It is also possible to suppress the Network Announcement for the received response code when a MtasNaNm with the key of the warning text exist and the configuration is set to suppress the announcement.

Location related additional information can be played depending on the configuration of announcement.

2.44.2 Example call flow: Requested Announcement

The Network Announcement service offered for fault situation is invoked internal by MTAS service. Why it is described in the context of the corresponding MTAS service, e.g. MMTel Basic Voice and Multi-media Communication (See Chapter 2.1).

In this example, only call flow for Requested Announcement that is invoked explicitly by a UA shows.

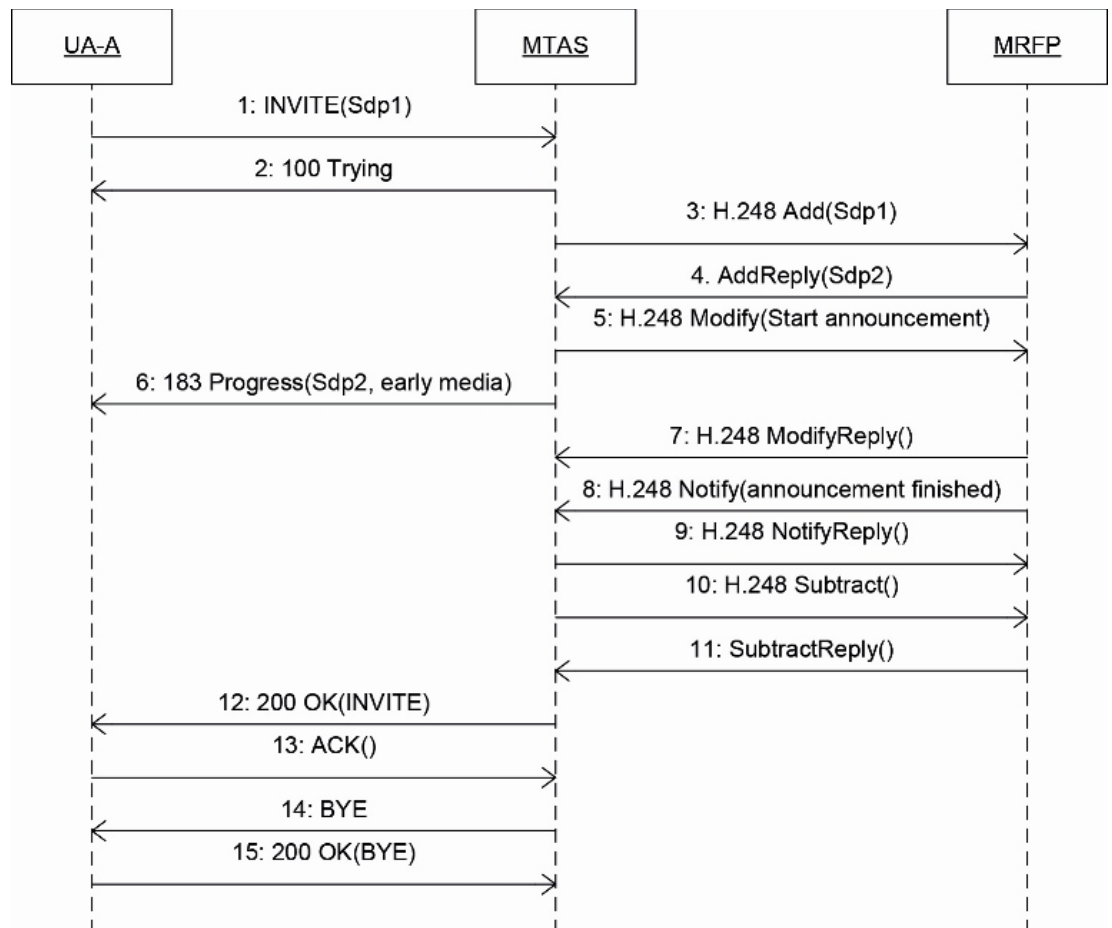


Figure 79 – Requested Announcement

1. Caller (A) performs an operation that results in their UA requesting the connection of an announcement. UA-A sends an originating INVITE where the Request-URI is a predefined string for the required announcement. The INVITE also includes the necessary SDP for UA-A to allow an announcement to be heard.
2. 100 Trying is sent immediately
3. MTAS sends a H.248 Add to the MRFP
4. MRFP responds with AddReply containing an updated SDP
5. MTAS sends 183 Progress requiring SDP answer with the P-Early-Media header set.
6. MTAS sends H.248 Modify to start playing the announcement
7. MRFP confirms started announcement by sending H.248 ModifyReply
8. MRFP sends H.248 Notify when the announcement has finished.
9. MTAS sends a NotifyReply



10. MTAS sends H.248 Subtract to remove the termination
11. MRFP answers with H.248 SubtractReply
12. MTAS sends 200 OK for INVITE
13. UA-A sends ACK
14. MTAS sends BYE to terminate the session
15. UA-A sends 200 OK for BYE

2.44.3 Configuration

- The Requested Announcement service is configured for the received URI

2.44.4 Example call flow: Network Message

This is one example when the 3PTY service rejects the initial INVITE with

403 Forbidden

Warning: (399,"At least one 3PTY participant in the URI list is not put on HOLD by the 3PTY originator")

With the configuration describe in 2.44.5, the following contents can be sent to the caller

403 Forbidden

Warning: (399,"At least one 3PTY participant in the URI list is not put on HOLD by the 3PTY originator")

Reason: SIP; cause=403; "Both 3pty participants must be put on hold before creating a 3pty call"

Note: If a network announcement is configured for the response code, the response line, warning header and reason header will have the same content.

2.44.5 Configuration

The Requested Announcement service is configured for the received URI

The Network message service is configured by

- Enabled the function

For specific content of the Warning header it is possible

- a. Configure the cause text in the in the Reason header in the preferred language
- b. Suppress network announcement or not



2.45 Customized Alerting Tones

2.45.1 Description

The Customized Alerting Tones (CAT) service is an IMS terminating service that triggers generation of customized signal (e.g. selected music or welcome announcement) towards the caller when the served user is alerted.

The CAT signal is generated from an external CAT Server (CAT-S). MMTel AS triggers only the CAT service invocation in the CAT Server.

The provisioning and evaluation of service rules for the CAT signal generation, the provisioning, storing and managing of the media content used as CAT signal is implemented in the CAT Server, so all these use case are out of the scope of the MTAS product.

The interface to the CAT-S is based on SIP.

Depending on MMTel AS node configuration MMTel AS triggers CAT signal generation from the CAT-S when 180 Ringing or 183 Session Progress is received from the served user or when based on a timer as long as the SDP in the received INVITE is not sendonly or inactive.

In order to prevent any announcements from subsequent nodes, MMTel AS sets all media streams to 'a=sendonly' in the INVITE sent to the served user before CAT signal triggering. When the served user answers the call, the media streams will be activated.

When the originating network does not support multiple early dialogs, it may indicate its capability limitation by inserting the no-fork directive in the Request-Disposition header. In this case MMTel AS will maintain a single SIP dialog towards the caller, in relation with the CAT service.

When CAT service is enabled, the end to end negotiation of SIP Preconditions between user A and user B is supported in MMTel AS if there is forking on user B side when MMTel AS node configuration is set to enable the transparent mode.



2.45.2 Example Call Flow

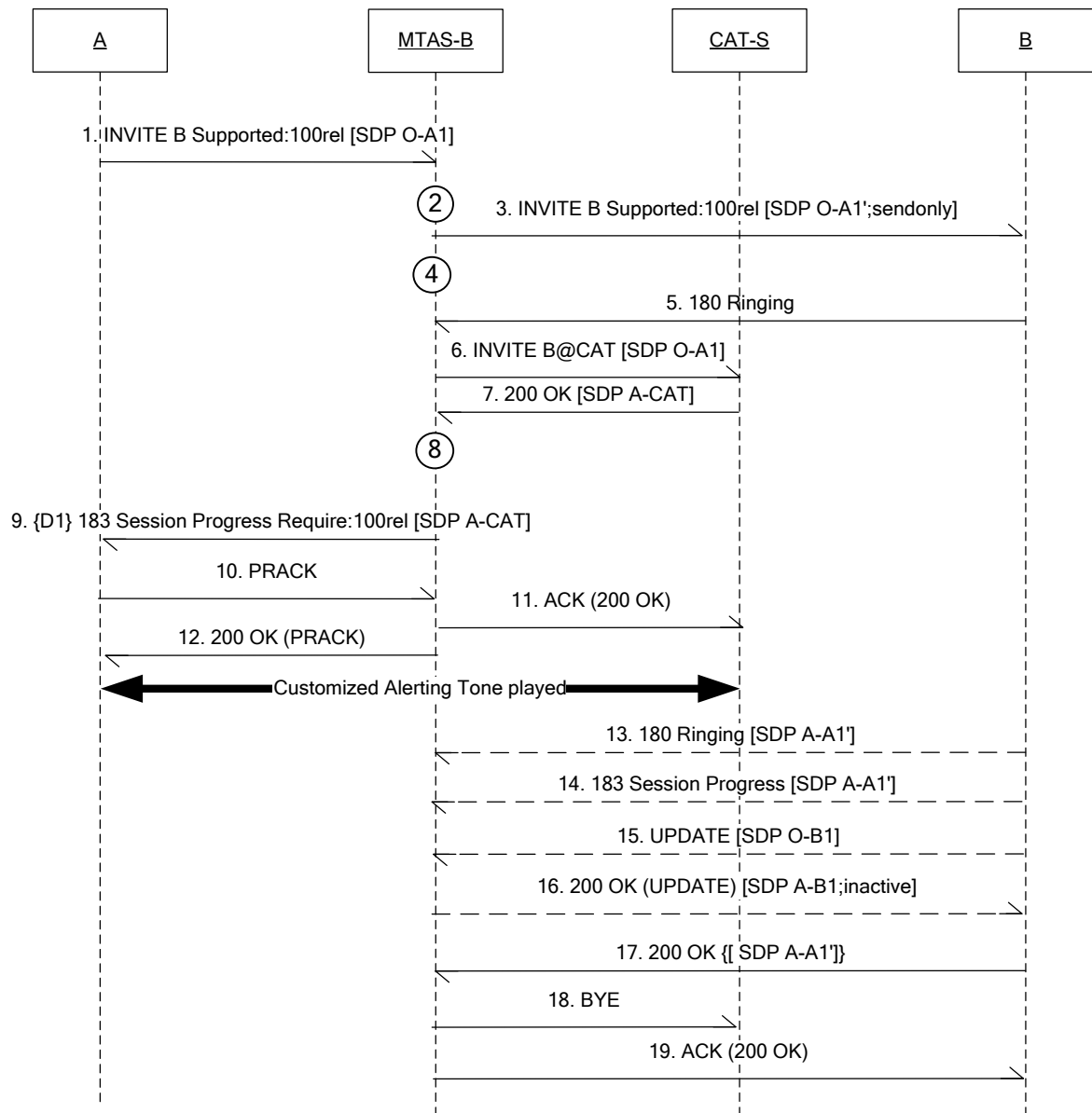


Figure 80 CAT signal successfully triggered, 180 Ringing is received, part 1

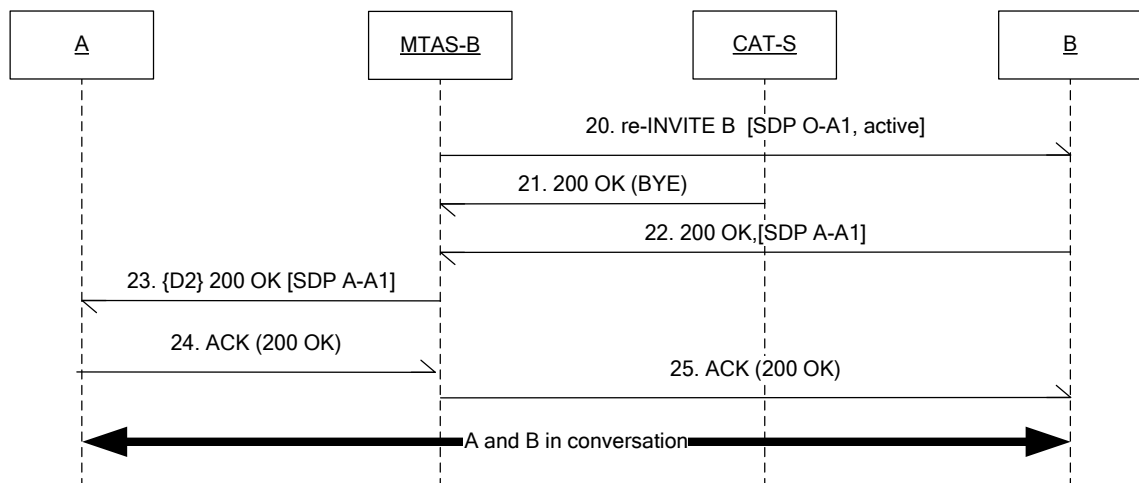


Figure 81 CAT signal successfully triggered, 180 Ringing is received, part 2

1. The MTAS receives an initial INVITE indicating “terminating registered” session case with the identity of the served user, (B) in the Request URI. The INVITE contains an SDP offer (O-A1) from user A.
2. The CAT service is triggered. CAT Service checks the license and the received SDP. If no valid license or the SDP is inactive/sendonly the CAT service is not applicable.
3. MTAS sends the INVITE to the served user, (B). The SDP offer in this request is the offer received in the INVITE from UA-A, step 1, except that all media streams are set to ‘a=sendonly’. This will prevent any announcements from subsequent nodes. If the CM attribute `mtasCatUseBlackHoleIPv4Address` is set to 1 (Yes) and IPv4 address is in the received offer, the INVITE is sent with address 0.0.0.0.
4. If the value of the CM attribute `mtasCatTimer` is not 0, the CAT Timer is started.
5. 180 Ringing is received from the served user, (B). In this example flow the 180 Ringing is received without SDP. If it is received with SDP, the subsequent actions are the same.
In this example flow the 180 Ringing is received without Required: 100rel. If it is received with Required: 100rel, then it is answered with PRACK. The CAT Timer is stopped.
6. MTAS sends INVITE to the CAT-S. The Request URI is composed of the user part of the INVITE received in step and the domain part of the CAT-S name (or IP address) defined in the node configuration..
The To, From, Privacy and P-Asserted-Identity headers are copied from the INVITE received in step 1.
The SDP offer is also copied from the INVITE received in step 1.
The CAT Request Timer, defined by CM attribute `mtasCatRequestTime`, is started.



7. CAT-S returns 200 OK with SDP answer. The CAT Request Timer is stopped.
8. The use of the CAT service is marked in the charging data.
9. MTAS sends a 183 Session Progress to the caller with an SDP answer based upon the local descriptor received from the CAT-S.
10. PRACK is received from A.
11. ACK is sent to CAT-S.
12. 200 OK (PRACK) is returned to A.
13. After step 5 any subsequent 180 Ringing from B (e.g. caused by forking at B) is stopped by MTAS, and answered if required by Required: 100rel.
14. After step 5 any subsequent 183 Session Progress from B is stopped by MTAS, and answered if required by Required: 100rel.
15. After step 5 any subsequent UPDATE from B is stopped.
16. MTAS sends a 200 OK with inactive SDP answer to the UPDATE from B.
17. 200 OK to the INVITE is received from B (with or without SDP depending on the previous responses).
18. MTAS sends BYE to CAT-S.
19. ACK to B.
20. MTAS sends re-INVITE to B with the original SDP offer received from A in step 1.
21. 200 OK (BYE) from CAT-S.
22. 200 OK with the SDP answer from B.
23. MTAS passes on the 200 OK with the SDP answer to A in a new dialog.
24. ACK.
25. ACK.

2.45.3 Charging

The use of Customized Alerting Tones service is reported in charging messages generated during the setup of an MMTel session when the CAT signal is successfully triggered, i.e. the CAT-S returns 200 OK.

The following service specific AVP is applicable to the CAT service:

1. Supplementary Service Information – Indicating use of CAT.



2.45.4 Service Interaction

2.45.4.1 Communication Diversion (CDIV)

CAT signal is not triggered in the CAT-S after CDIV service invocation, that is, after when service specific condition for the diversion is fulfilled (e.g. after No-Reply timer expiry for CFNR or after busy condition detection for CFB).

If the served user has active CAT service and there is a valid license for CAT present on the MTAS, no CDIV announcement is played on CDIV invocation.

Transparent mode is disabled:

If the CAT signal is triggered in the CAT-S before CDIV invocation, then the CAT signal is stopped when provisional response with SDP or final response is received from the diverted-to network.

Transparent mode is enabled:

If the CAT signal is triggered in CAT-S before CDIV invocation, for example, the CDIV invoked after time-out or with a non-200 OK final response after ringing, the CAT signal is stopped and a diversion announcement may be played to the caller and the INVITE is sent to the diverted to user.

2.45.4.2 Flexible Communication Distribution (FCD)

If the served user has active CAT service and there is a valid license for CAT present on the MTAS, no FCD announcement is played on FCD invocation. Instead the CAT signal generation will be triggered when 180 Ringing is received from any of the targets or the CAT Timer, by the CM attribute `mtasCatTimer`, expires.

2.45.4.3 Identity Presentation

MTAS passes on the received Privacy header unchanged to the CAT Server independent of the served user provisioned or not provisioned Identity Presentation services, e.g. OIP, or OIR with Override.

2.45.4.4 Communication Waiting (CW)

Transparent mode is disabled:

The caller will receive no indication that the served user is busy but does have CW, as the CAT service stops the CWU indication included in the received 180 Ringing from the served user.

Transparent mode is enabled:

A CWU indication in a 180 Ringing that also include SDP information is forwarded to the caller. When 180 Ringing includes CWU indication but no SDP information the 180 Ringing is mapped to 183 Progress with the CWU indication and forwarded to the caller.



2.45.4.5 gsmSSF

MTAS passes on the received From header unchanged to the CAT Server, that is the not considering the header update ordered by the CAMEL service. If the CAMEL service orders update of the From header, MTAS passes on the updated From header to the CAT Server.

2.45.4.6 Network provided Ring Back Tone (RBT)

The CAT service is invoked before the Network provide Ring Back Tone service. However, if the CAT Server is not available or cannot handle the customized alerting tone request the RBT service, if active, is invoked.

2.45.4.7 MMTel Basic call

The expiration of MMTel No Reply timer stops the alerting tone and cancels the call.

2.45.5 Configuration

- Activation (active/disabled)
- Allowing CAT signal generation on 183 Session Progress from the served user
- Allowing and setting the CAT Timer
- Determining whether Black Hole IPv4 address shall be used in the SDP offer sent to the served user with active CAT service
- Determining whether MTAS shall maintain single SIP dialog towards the caller when no-fork directive is set in the Request-Disposition header of the received INVITE, in relation with the CAT service
- Setting the CAT Request Timer
- Filtering of the headers sent to CAT Server
- Activation of support of end to end negotiation of SIP preconditions between terminals of users A and B.

2.45.6 Performance Management

- Number of CAT signal generation attempts
- Number of successful CAT signal triggerings
- Number of unsuccessful CAT signal triggering, external
- Number of unsuccessful CAT signal triggering, internal



2.45.7 Fault Management

For information on the alarm, refer alarm OPI [68].

2.46 Network Provided Ring Back Tone

2.46.1 Description

The Network Provided Ring Back Tone (RBT) service is an IMS terminating service that triggers an MRFP to generate an RBT signal (customized welcome message or ringing tone) towards the caller while the served user is alerted. The tone is common for all the IMS subscribers and is only configurable by an operator.

MMTel AS triggers RBT signal generation when the 180 Ringing is received from the served user.

In order to prevent any announcements from subsequent nodes, MMTel AS suppresses P-Early-Media header value by using "P-Early-Media: inactive" header in SIP signals relayed upstream. When the served user answers the call, the media streams will be activated.

When the originating network does not support multiple early dialogs, it may indicate its capability limitation by inserting the no-fork directive in the Request-Disposition header. In this case MMTel AS will maintain a single SIP dialog towards the caller, in relation with the RBT service.

2.46.2 Example Call Flow

The sequence shown use a Supported: 100rel, 199, precondition header in the received initial INVITE request.

MTAS has the transparent mode switched on.

MTAS has the 199 generation switched on.

Calling party A and called party B supports preconditions: header Supported: precondition

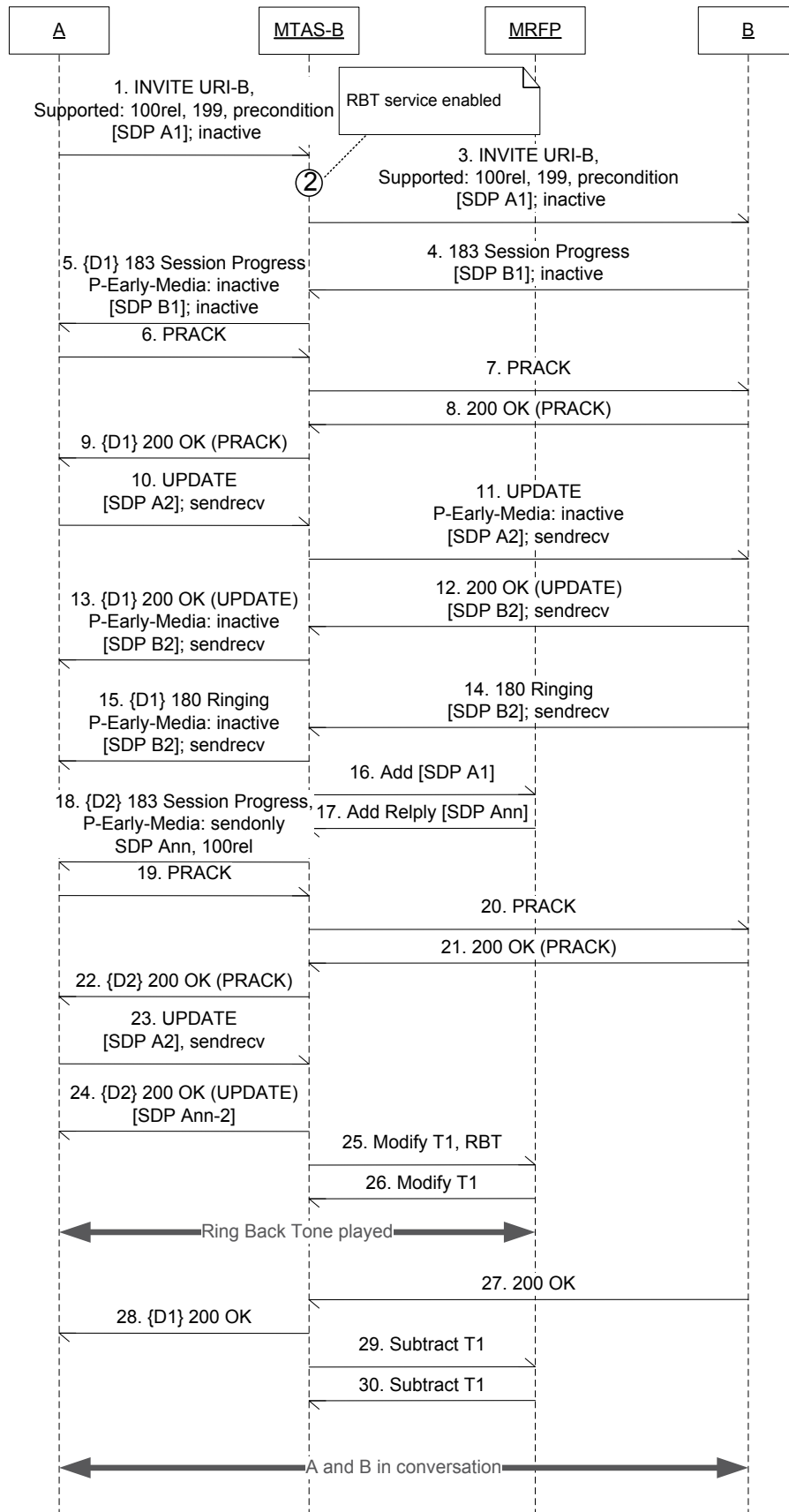


Figure 82 – RBT signal successfully triggered



1. User A starts a communication with User B by sending an INVITE message. The INVITE contains an SDP and Supported header with 100rel, 199 and precondition. The INVITE contains an SDP offer (A1) from user A. The SDP has:
 - all media set to inactive
 - local current media status set to 'none'
 - remote current media status set to 'none'
 - local desired media status set to 'sendrecv'
 - remote desired media status set to 'sendrecv'
2. RBT checks the license and administrative state.
3. INVITE is forwarded to User B unchanged.
4. MTAS receives 183 Session Progress from B with SDP contains:
 - all media set to inactive
 - local current media status set to 'none'
 - remote current media status set to 'none'
 - local desired mandatory media status set to 'sendrecv'
 - remote desired mandatory media status set to 'sendrecv'
 - the confirmation of remote media status set to 'sendrecv' so UA-A will send a confirmation when the status of network resources reaches these condition
5. MTAS relays 183 Session Progress to A unchanged except adding P-Early-Media:inactive header field. P-Early-Media suppresses unwanted backward early media
6. MTAS receives PRACK from A
7. MTAS relays PRACK to B
8. MTAS receives 200 OK(PRACK) from B
9. MTAS relays 200 OK(PRACK) to A
10. MTAS receives from A an UPDATE with SDP(A2) contains:
 - all media set to sendrecv
 - local current media status set to 'sendrecv'
 - remote current media status set to 'none'
 - local desired mandatory media status set to 'sendrecv'



- remote desired mandatory media status set to 'sendrecv'
11. MTAS relays the UPDATE to B unchanged except adding P-Early-Media:inactive header field

12. MTAS receives 200 OK (UPDATE) from B with SDP contains:

- all media set to sendrecv
- local current media status set to 'sendrecv'
- remote current media status set to 'sendrecv'
- local desired mandatory media status set to 'sendrecv'
- remote desired mandatory media status set to 'sendrecv'

It means that all preconditions needed for establishment of connection are met on B and A side

13. MTAS relays 200 OK (UPDATE) to A unchanged except adding P-Early-Media:inactive header field. P-Early-Media suppresses unwanted backward early media.

14. User B sends 180 Ringing which is a trigger for the RBT service to start the sound towards the User A.

15. MTAS relays 180 Ringing to A.

16. MTAS creates a termination in a new context for User A.

17. MRFP responds with an allocated SDP.

18. RBT generates a reliable 183 Session Progress. It contains the MRFP SDP and P-Early-Media header with the value sendonly.

19. User A confirms the reception of 183 with PRACK.

20. MTAS relays PRACK to B

21. MTAS receives 200 OK(PRACK) from B

22. MTAS relays 200 OK(PRACK) to A

23. MTAS receives from A an UPDATE with SDP(A2) contains:

- all media set to sendrecv
- local current media status set to 'sendrecv'
- remote current media status set to 'none'
- local desired mandatory media status set to 'sendrecv'
- remote desired mandatory media status set to 'sendrecv'

24. MTAS sends to A 200 OK (UPDATE).



25. MTAS orders MRFP to start playing the infinite sound with a Modify command.
26. MRFP replies with an empty Modify. The tone is being played to the User A while waiting for the User B to answer the call.
27. 200 OK (INVITE) is received from UA-B.
28. MTAS sends 200 OK (INVITE) to UA-A.
29. RBT orders the MRFP to stop the sound with a Subtract.
30. MRFP replies the Subtract.

2.46.3 Charging

The use of Network Provided Ring Back Tone service is not reported in charging messages.

2.46.4 Service Interaction

2.46.4.1 Customized Alerting Tones

If MTAS is configured to play RBT signal and the served user is provisioned with the CAT service, then the CAT signal is attempted to be played first. If the play of CAT signal fails, MTAS falls back to the RBT signal.

2.46.4.2 Communication Diversion (CDIV)

RBT signal is stopped after CDIV service invocation, that is, after when service specific condition for the diversion is fulfilled (e.g. after No-Reply timer expiry for CFNR or after busy condition detection for CFB).

2.46.4.3 Flexible Communication Distribution (FCD)

FCD announcement is played on FCD invocation even if RBT is enabled.

2.46.4.4 Communication Waiting (CW)

When CWU indication is present in a 180 Ringing with SDP, information is forwarded to the caller. When 180 Ringing includes CWU indication but does not contain SDP, then the 180 Ringing is mapped to 183 Session Progress with the CWU indication and forwarded to the caller.

2.46.4.5 Network Announcement

Network announcements take precedence over RBT and will replace RBT sound with network announcements.



2.46.5 Configuration

- Activation (active/ disabled)
- Setting the RBT signals list

2.46.6 Performance Management

- Number of successful RBT triggerings
- Number of unsuccessful RBT triggering

2.47 Parlay X

MTAS can act as a Service Capability Server, as defined in [39], exposing features for the Parlay X enabled application to use.

The Parlay X function supports the ThirdPartyCall interface and the CallDirection interface (see [40] and [41]).

2.47.1 Third Party Call Interface

2.47.1.1 Description

Parlay X Third Party Call is a service that enables an external Parlay X application to initiate and manage a call between two participants.

The service implementation is based on ref [40] and supports the following operations of the ThirdPartyCall interface:

- MakeCallSession
- GetCallSessionInformation
- EndCallSession

The call can be initiated in two modes depending on the usage of the MakeCallSession operation parameters.

Basic mode, CallingParticipantName is not used

- The call is setup between parties as given by CallParticipants parameter.
- Identities of call participants are not altered.
- If Charging/Code parameter is available – it is used for charging. Otherwise first participant is charged for the call.

Enhanced mode, CallingParticipantName is used

- The call is setup between parties as given by CallParticipants parameter on behalf of caller specified in CallingParticipantName.



- Identities presented to call participants are altered so they see caller as specified in CallingParticipantName.
- Caller specified in CallingParticipantName is charged for the call.

2.47.1.2 Example Call Flow, basic mode, charging/code parameter not used

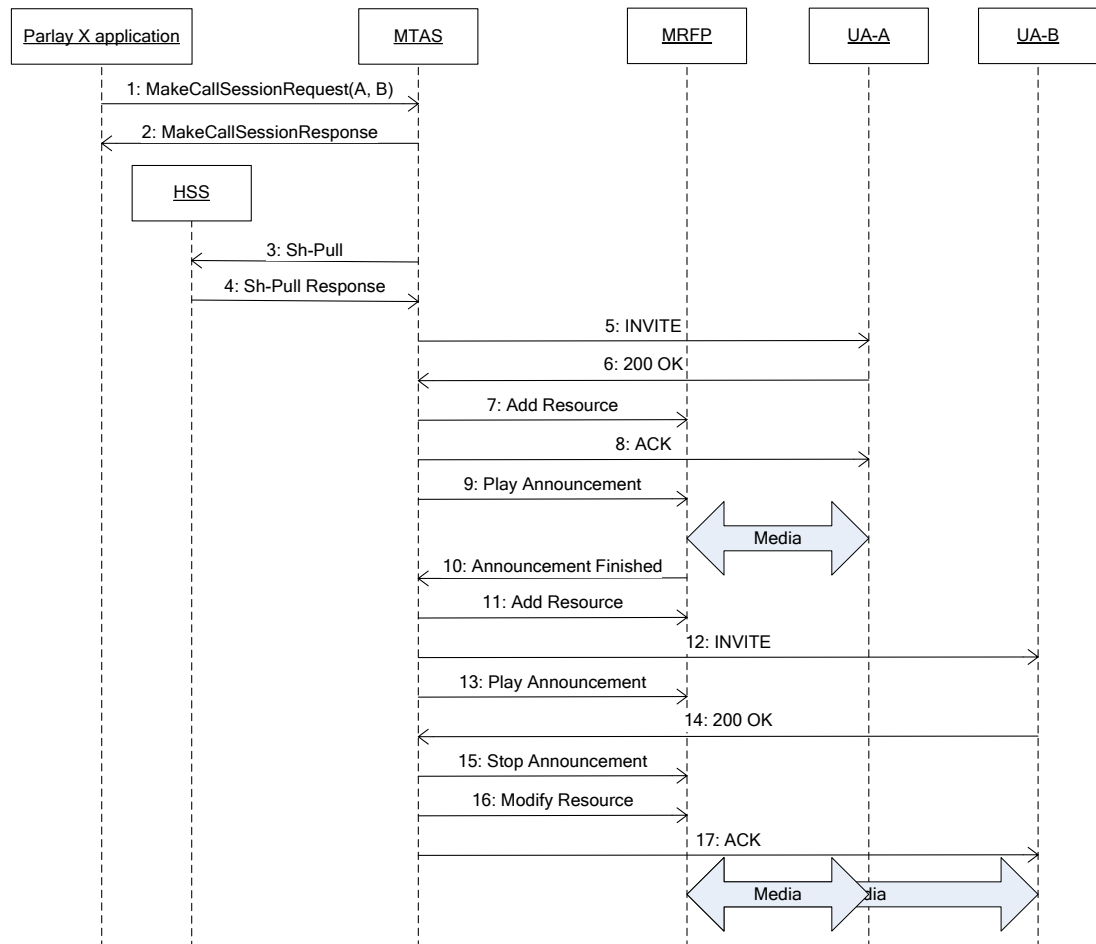


Figure 83 Parlay X Third Party Call

1. Parlay X Application makes MakeCallSession request, CallingParticipantName is not set, Charging/Code is not set.
2. MTAS sends response to Parlay X Application immediately.
3. MTAS sends a request for transparent user data (MSISDN and Charging Information) with the first participant (of CallParticipants) as PUI.
4. The requested data is returned.
5. MTAS sends an INVITE to the UA-A that is first element of CallParticipants parameter.



6. UA-A sends 200 OK.
7. MTAS sends a H.248 Add to the MRFP to reserve media resources.
8. MTAS sends an ACK to the UA-A.
9. MTAS orders the MRFP to play an announcement indicating that the call leg to UA-B will be setup.
10. When the announcement has finished playing, the MRFP sends a Notify to MTAS.
11. MTAS reserves media resources for UA-B in the same context as for UA-A using a H.248 Modify.
12. MTAS sends an INVITE to UA-B that is second element of CallParticipants parameter.
13. MTAS orders the MRFP to play an announcement to UA-A, indicating that UA-B's phone is ringing.
14. UA-B sends a 200 OK to MTAS.
15. MTAS orders the MRFP to stop the announcement towards UA-A.
16. MTAS sets the remote media endpoint of UA-B in the MRFP.
17. MTAS sends an ACK to UA-B. Call between participants is established.

2.47.2 Call Direction Interface

2.47.2.1 Description

The Parlay X Call Direction service supports the HandleCalledNumber operation.



2.47.2.2 Example Call Flow

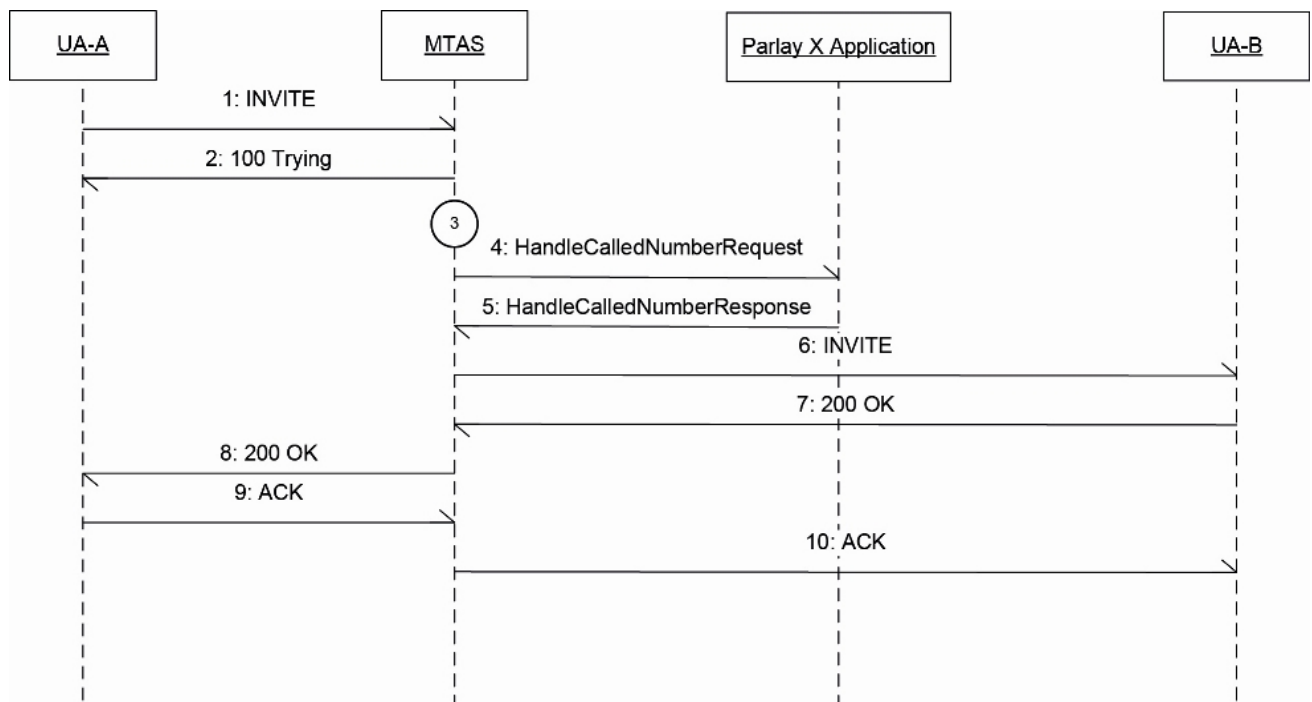


Figure 84 - Route the call to a new target

1. UA-A sends a SIP INVITE to MTAS.
2. MTAS returns a SIP 100 Trying to the UA-A.
3. MTAS creates a subscriber process and decides that this is a call to a temporary number. The decision is based on the fact that the INVITE arrived on the port assigned for ParlayX Call Notifications.
4. MTAS sends a HandleCalledNumberRequest to the external Parlay X application.
5. The external Parlay X application sends a HandleCalledNumberResponse to MTAS indicating that the call shall be routed to a new target.
6. MTAS sends an INVITE to the target specified by the external Parlay X application.
7. MTAS receives a 200 OK from the target.
8. MTAS sends 200 OK to UA-A.
9. UA-A sends ACK.
10. MTAS sends ACK to the target.



2.47.3 Service Interaction

The MTAS Service Capability Server offering Parlay X services is invoked as a separate Application Server in IMS, so there is no interaction with MMTel services of the MMTel AS.

2.47.4 Provisioning

The Parlay X function does not require any service specific subscription data. However the user being charged for the Third Party Call needs to be provisioned in HSS.

2.47.5 Configuration

MTAS supports the following configuration:

- Administrative state.
- Third Party Call audio and video announcement codes.
- Privacy of participant B in Third Party Call
- Third Party Call media bridge removal flag
- Third Party Call setup timeout
- URL to the external Parlay X application

2.47.6 Performance Management

The following performance counters are provided by MTAS:

- Counter for number of successfully established Parlay X Third Party Call sessions from originating MTAS handling Parlay X requests.
- Counter for total Third Party Call communication duration time for originating calls.
- A counter that is incremented when sending a Parlay X Handle Called Number request to the external Parlay X application. The counter is keyed by the service that the request was sent from.
- A counter that is incremented when receiving a Parlay X Handle Called Number error response from the external Parlay X application. The counter is keyed by the service that the request was sent from.



2.48 Parlay X in MMTel

2.48.1 Call Notification Interface

2.48.1.1 Description

Parlay X in MMTel is a service that can execute in an originating and/or terminating MMTel AS. Which session case the service executes in, is based on provisioning.

The service implementation is based on ref [39] and supports the CallNotification interface. This interface provides an external Parlay X Application with notifications about the progress of the served user's calls.

2.48.1.1.1 General

Since the Parlay X application cannot control the call, there are no service interactions between the IN services in the Parlay X application and MMTel services.

In general, when MMTel services creates a new outgoing call leg, a new Parlay X trigger will fire and the Parlay X application will be notified about this new call event. Below is a list of MMTel services that has this behavior.

2.48.1.1.2 Call Diversion

When a new outgoing call leg is created as a result of Call Diversion, the Parlay X application will be notified based on the served users originating Parlay X trigger.

2.48.1.1.3 Ad-hoc Conferencing

When this MMTel service invites a new participant a new outgoing call leg is created. The Parlay X application will then be notified based on the served users originating Parlay X trigger.

2.48.1.1.4 Communication Completion

When a Communication Completion recall is made, a new outgoing call leg is created. The Parlay X application will then be notified based on the served users originating Parlay X trigger.

2.48.1.1.5 Flexible Communication Distribution

When a new outgoing call leg is created as a result of Flexible Communication Distribution, the Parlay X application will be notified based on the served users originating Parlay X trigger. Note that this might lead to that the Parlay X application is notified many times for the served user.



2.48.1.1.6 Session Transfer to Own Device

When a new outgoing call leg is created as a result of Session Transfer to Own Device, the Parlay X application will be notified based on the served users originating Parlay X trigger. Note that this might lead to that the Parlay X application is notified many times for the served user.

2.48.1.2 Example Call Flow

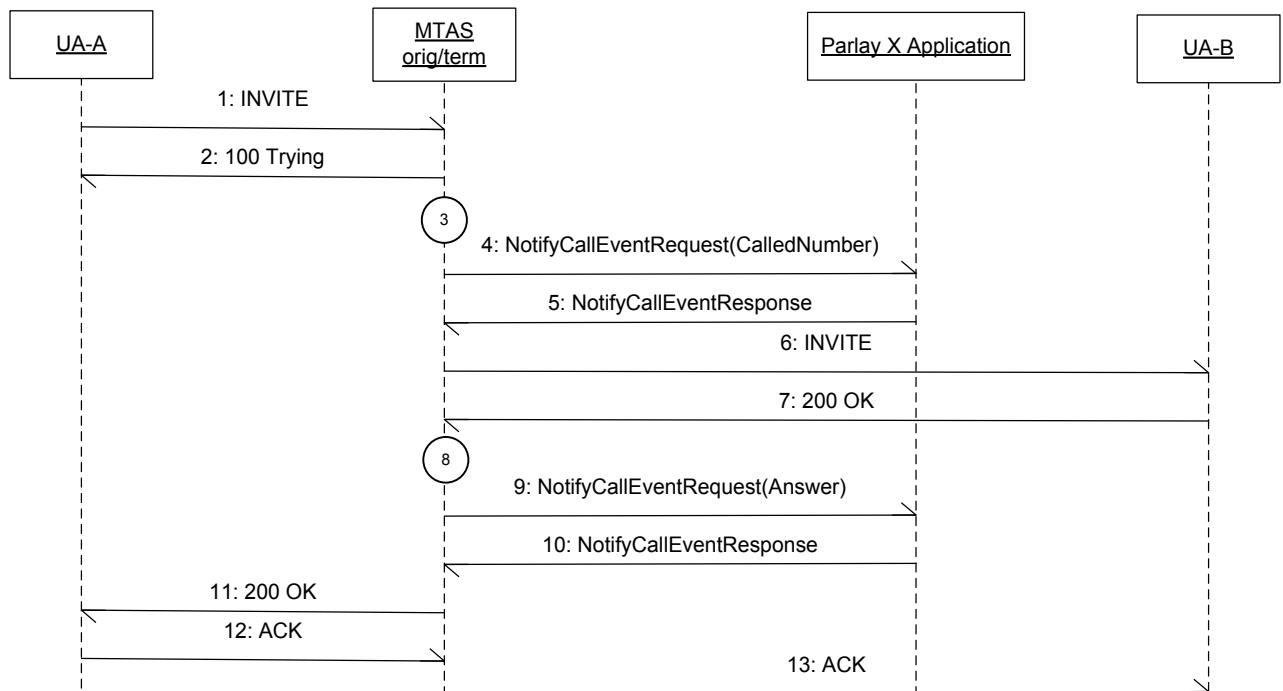


Figure 85 Parlay X call notifications

1. UA-A sends a SIP INVITE to MTAS.
2. MTAS returns a SIP 100 Trying to the UA-A.
3. MTAS evaluates the pre-condition above to be true and decides that the Parlay X application shall be notified of this call event.
4. MTAS sends a NotifyCallEventRequest with the CallEvent “CalledNumber” to the external Parlay X application and continues processing the call.
5. MTAS receives a NotifyCallEventResponse from the external Parlay X application. No further actions are taken by MTAS based on this response.
6. MTAS sends an INVITE to the original target.
7. MTAS receives a 200 OK from the target.
8. MTAS evaluates the pre-condition above to be true and decides that the Parlay X application shall be notified of this call event.



9. MTAS sends a NotifyCallEventRequest with the CallEvent “Answer” to the external Parlay X application and continues processing the call.
10. MTAS receives a NotifyCallEventResponse from the external Parlay X application. No further actions are taken by MTAS based on this response.
11. MTAS sends 200 OK to UA-A.
12. UA-A sends ACK.
13. MTAS sends ACK to the target.

2.48.2 Configuration

MTAS supports the following configuration:

- Administrative state

2.48.3 Performance Management

- Counter for sent Parlay X requests in the originating MTAS
- Counter for received Parlay X responses in the originating MTAS
- Counter for sent Parlay X requests in the terminating MTAS
- Counter for received Parlay X responses in the terminating MTAS
- Counter for sent Parlay X requests in the originating unregistered MTAS
- Counter for received Parlay X responses in the originating unregistered MTAS
- Counter for sent Parlay X requests in the terminating unregistered MTAS
- Counter for received Parlay X responses in the terminating unregistered MTAS

2.49 GSM compatible SSF

2.49.1 Description

MTAS can interact with CAMEL services deployed on an SCP. The dialog towards the CAMEL service is established by MTAS if the served user is provisioned with CAP support or using B-Number based CAP Triggering.

MTAS acts as an SSP co-located with an SRF, as defined in [47], exposing features for the CAMEL service to use.

2.49.2 Example Call Flow

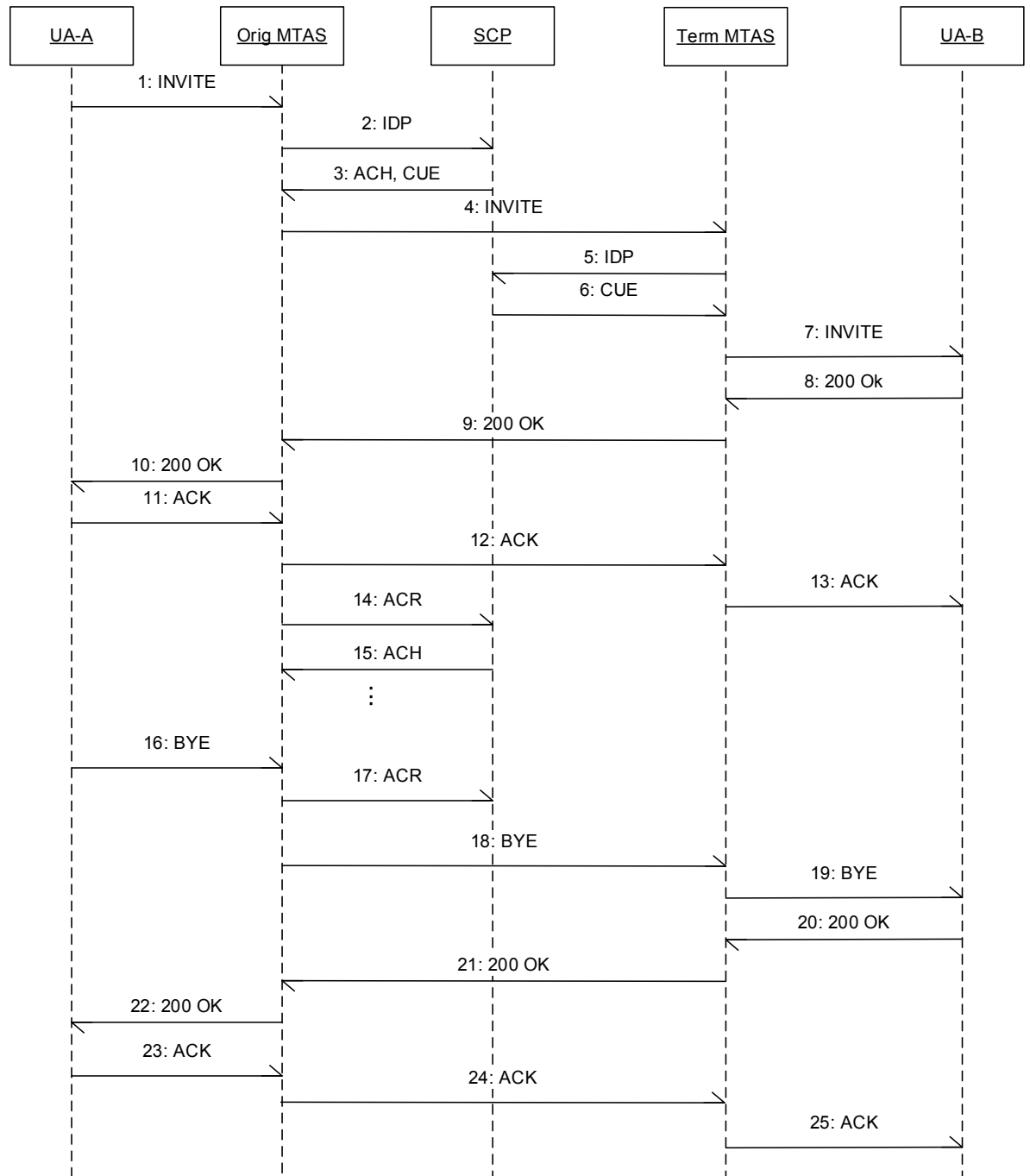


Figure 86 CAMEL pre-paid charging

1 UA-A sends a SIP INVITE to MTAS



- 2 MTAS sends a CAP IDP (Initial DP) to the SCP. This creates the TCAP dialog.
- 3 SCP returns with a CAP CUE and a CAP ACH to MTAS. The CUE indicates that the call shall be setup using available call data. No DPs are armed in this scenario.
- 4 MTAS sends a SIP INVITE to the Terminating MTAS serving UA-B.
- 5 MTAS sends a CAP IDP (Initial DP) to the SCP. This creates the TCAP dialog.
- 6 SCP returns with a CAP CUE to MTAS. The CUE indicates that the call shall be setup using available call data. No DPs are armed in this scenario.
- 7 – 13. The scenario continues with the basic call setup.
14. The MaxCallPeriodDuration timer is elapsed and the charging report that was ordered in step 3 is sent to the SCP.
15. SCP returns a CAP ACH to give another maximum call duration. Step 14 and 15 can occur repeatedly.
16. UA-A sends a SIP BYE to release call.
17. MTAS sends a CAP ACR to the SCP as a final call time report.
18. Steps 18- 2 the normal call clearing procedure applies.

Note: If CM mtasNccCreditAnnouncementName is configured, the configured warning tone is played to the served user for an ongoing call before the allowed duration has been reached, if the earlier received CAPv2 Apply Charging (ACH) message included the "Release If Duration Exceeded" with a Tone parameter.

2.49.3 Charging

The CAMEL service impacts MMTel charging. The impact depends on which CAP operation that is received or sent. Below is a list of the CAP operations that have an impact:

2.49.3.1 Connect (CON)

If a CAMEL diversion takes place, the new diversion target is set in the Charging Function.

2.49.3.2 Furnish Charging Information (FCI)

Any free format data received in a CAP FCI is stored in the Service-Specific-Data AVP.

The value Party To Charge information element is used as the charged party.



2.49.3.3 Initial Detection Point (IDP)

Some Information Elements (IE) are also reported on charging to reflect the call information, for instance IMSI and MSC address.

The usage of CAMEL service (O-CSI/T-CSI/Destination based CAP Trigger) is reported as Supplementary-Service-Identity AVP

2.49.4 Service Interactions

2.49.4.1 Abbreviated Dialing

The CAMEL service will be invoked after the Abbreviated Dialling service. Thus, a short number is translated to a long number before it is handled by the CAMEL service.

2.49.4.2 Address Policing

Address policing will be applied when the CAMEL service has translated the dialed number.

2.49.4.3 Call Return

Originating CAMEL invocation will be done for the INVITE sent when the served user wants to call the last user, who called the served user.

2.49.4.4 Calling Party Category

If a Calling Party Category is received from the CAMEL service this value overrides the Calling Party Category value provisioned for this user.

2.49.4.5 Charging Service

The CAMEL service impacts MMTel online and offline charging. The impact depends on which CAP operation is received or sent.

2.49.4.6 Communication Barring

2.49.4.6.1 Outgoing Communication Barring

Outgoing Communication Barring is performed on the result from the CAMEL service invocation on the originating MTAS. This means that any new destination address received from the SCP will be checked by the Outgoing Barring service.

The same as above applies on a terminating MTAS for a diverted call.

2.49.4.6.2 Incoming Communication Barring

Incoming Communication Barring is performed on the terminating MTAS before CAMEL service invocation.



2.49.4.7 Communication Completion

A CAMEL service invocation is done on the call leg for the re-call towards B.

2.49.4.8 Communication Diversion

If a CAMEL call diversion takes place, that is when the SCP sends a new called party number in the CAPv2 CON message, then MMTel Communication Diversion service is not invoked.

If a MMTel Communication Diversion at Busy or No-Reply service diverts the call and if the SCP node armed event reporting with RRB (tBusy/tNoAnswer) before that, the ERB notification is not sent by CAMEL service."

When MMTel CDIV takes place, as by default SCP O-CSI applicable is set to true, MTAS will trigger CAMEL interaction for the diverted leg. This is the case when CUE is sent as response for the IDP for the MT case.

In case of Communication Deflection, the deflected call will be subject to CAMEL control in the same way as for a forwarded call.

2.49.4.9 Conference

CAMEL interaction is not invoked for the INVITE sent for conference creation. A CAMEL invocation, based on the O-CSI for the conference participant, will be done for every outgoing call leg.

Terminating services for a participant, including CAMEL service invocation, will be executed as for any terminating call. No specific action related to CAMEL is taken related to that the terminating user is a conference participant.

2.49.4.10 Flexible Communication Distribution

One CAMEL invocation is made for the original target and one CAMEL invocation is made for each outgoing call leg.

If a CAMEL diversion takes place, FCD is not executed.

2.49.4.11 Flexible Service Format Selection

It is not possible to suppress the execution of the CAMEL service by using the FSFS functionality.

2.49.4.12 Identity presentation

When the CAMEL service invocation results in the reception of a CONNECT message including a 'Generic Number' parameter, this number is included in the display name part of the From and P-Asserted-Identity headers in the INVITE message.



When Identity Presentation is enabled the OCNIP and CNIP service will not allow presentation of the From and P-Asserted-Identity headers unless valid license for CNIP is available and OCNIP and CNIP is enabled. When Identity Presentation is not enabled the Generic Number will be present in the From and P-Asserted-Identity headers when received in CON.

MTAS uses the values “presentation allowed” and “presentation restricted” for the IDP.CallingPartyNumber.ScreeningIndicator parameter field. If the SIP Privacy header contains any of the values “id”, “user” or “header”, MTAS maps this to “presentation restricted”. In all other cases MTAS uses the value “presentation allowed”.

2.49.4.13 Number Normalization

The numbers sent to the CAMEL service in the information elements of the IDP, may or may not be in a normalized format. This is to enable CAMEL services like VPN groups to translate the dialed number into a public number.

Numbers received from the CAMEL service may or may not be in a normalized format. The Number Normalization common component must be configured in a way that allows all non-global numbers received from the CAMEL service to be normalized. Therefore, every number in the Request URI of the outgoing INVITE is normalized.

2.49.4.14 Number Portability

A number portability analysis will be made on numbers received from the CAMEL service.

2.49.4.15 Number Translation

The number translation analysis will be made on number received from the CAMEL service.

2.49.4.16 Parlay X

CAMEL interaction will not be invoked when Parlay X is enabled on a per subscriber basis.

2.49.4.17 Session Transfer to Own Device

One CAMEL invocation is made for the original target, but no CAMEL invocations are made for the outgoing STOD call legs.

2.49.4.18 Short Number Dialing

The CAMEL service will handle the request URI before it is seen by the SND service. This mean that the CAMEL service might translate a short number into a public number before SND handles the destination address.



2.49.4.19 Supplementary Service Codes

The SSC service will handle the request URI before it is seen by the CAMEL service.

SSC will not be invoked on a Follow-on Call.

2.49.4.20 Three Party Call

The CAMEL service is executed on the call legs for the 3PTY participants as for a normal originating/terminating session independently of the 3PTY execution.

No CAMEL service is executed on the call leg for the 3PTY originator.

2.49.5 Configuration

MTAS provides the operator a number of node-level configuration parameters which affects the behavior of the CAP support. The parameters are managed by LDAP. The parameters are:

- 1 Enable and disable this service
- 2 MTAS subsystem number - This attribute defines the subsystem number that MTAS will use in the SS7 network
- 3 Global Title - This attribute is the Global Title that MTAS is assigned in the SS7 network.
- 4 BNumberBasedTrugger – This attribute defines whether the CAMEL triggering based on B-Number is supported. If it is enabled, an IDP can be sent based on the mtasNccBNumberList configuration
- 5 BNumberList – This attribute defines a list of strings, including <B-Number-pattern>, <gsm-scf-address>, <service-key>, <default-call-handling>. Each field is separated by “|”.
- 6 ReportSsiToCharging – This attribute defines whether MTAS reports the usage of Camel service to charging in Supplementary-Service-Information AVP (both Rf and Ro interfaces).

2.49.6 Performance Management

The following performance counters are provided by MTAS:

- A counter that is incremented when sending a CAP operation to the SCP. The counter is keyed by the operation name.
- A counter that is incremented when receiving a CAP operation from the SCP. The counter is keyed by the operation name.



- A counter that is incremented when ACH message received from SCP which contains IE `ReleaseDurationExceeded`.
- A counter that is incremented when the call is released because of the ACH message received from SCP which contains IE `ReleaseDurationExceeded`.
- A counter that is incremented when CAP is triggered by B-Number.

2.50 Multiple Languages Support

2.50.1 Description

MTAS supports sending of announcements and voice prompts to the served user according to the user's provisioned language preferences.

Languages for announcements are specified by the content of the language prefix inserted in front of the announcement code.

The language prefix used for announcements is selected according to the served user's provisioned language tag.

When no language tag is provisioned, or the announcement is not sent to the served user, the default prefix is used.

2.50.2 Configuration

- The default language prefix
- The language prefix for each provisioned language tag

2.50.3 Performance Management

The following performance counters are provided by MTAS:

- A counter for each language that is incremented when the language is selected for announcement based on provisioned language preference.

2.51 Emergency Call Notification

2.51.1 Description

Start and stop of an Emergency Call is indicated to OCS over the Ro interface based on SIP NOTIFY received from CSCF. The data in the SIP Event header received in the NOTIFY is used by MTAS to populate the Service-Specific-Info AVP having sub-AVPs Service-Specific-Type and Service-Specific-Data.

2.51.2 Example Call Flow

The call flow is summarized below.

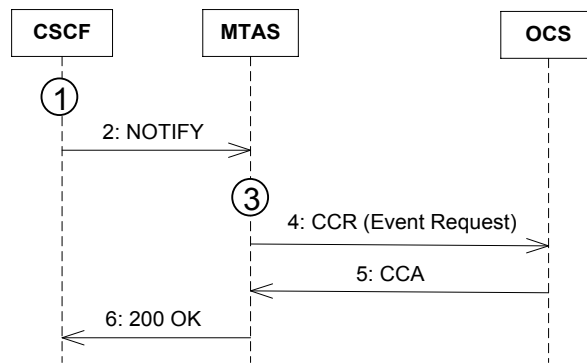


Figure 87 Emergency Call Notification

- 1 Emergency call is initiated or ended.
- 2 CSCF sends an unsolicited SIP NOTIFY on the ISC interface to the originating MMTel AS with session case unregistered and with Event header value “emergencyCall” and parameter “start” or “stop”.
- 3 Emergency Call Notification function of MTAS is invoked. The following conditions hold:
 - The charging profile for Emergency Call is set to “Online charging”.
 - It is found that the Subscriber is an online charging subscriber by matching the P-Charging-Function-Addresses (PCFA) header included in the NOTIFY against the provisioned ECF address of the Subscriber or the default ECF when there is no provisioned value.
- 4 MTAS sends CCR Event on Ro interface with Service-Specific-Info AVP (and sub-AVPs) populated to report Emergency Call start or end according to data received in the Event header.
- 5 OCS answers with CCA.
- 6 MTAS sends 200 OK to CSCF as a response to the NOTIFY.

2.52 Closed User Group

2.52.1 Description

The Closed User Group (CUG) service enables operators to form groups of users, whose communication profile is restricted for incoming and outgoing communications.

Members of a specific CUG can communicate among themselves but not with users outside the group, unless they have additional capabilities that allow them to initiate outgoing communications.

The Closed User Group service partially conforms to [50] with the following limitations:



- Only the terminating AS part is supported.
- No support for end-user configuration of CUG.
- No support for configurable Incoming Access.

2.52.2 Example Call Flow

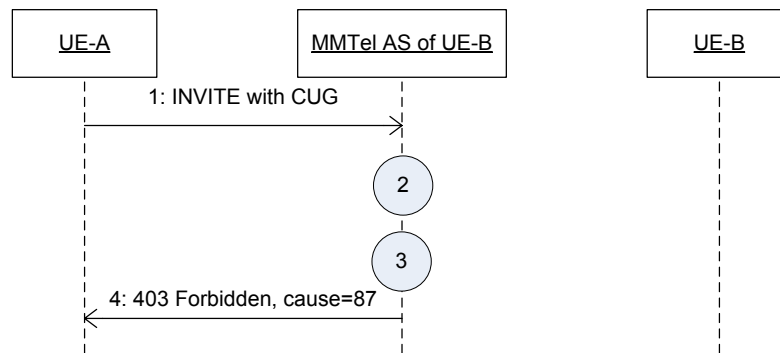


Figure 88 Rejected CUG call when INVITE contains CUG information

1. A Terminating INVITE containing CUG information is received by MMTel AS.
2. The CUG checks are run. The checks fail because one or more of the following conditions are true:
 - a. The calling party is associated with a different CUG
 - b. The calling party has no Outgoing Access and the called party is not associated with a CUG
3. An announcement is played towards UE-A.
4. MMTel AS responds with 403 Forbidden. The SIP response contains the following SIP header:

```
Reason: Q.850 ;cause=87 ;text="User is not a member of
Closed User Group (CUG) "
```



2.52.3 Service Interactions

2.52.3.1 Communication Barring

The Incoming OTP Global White list and the VTP Global White List take precedence over the Closed User Group service.

The Closed User Group Service takes precedence over the operator and user allowed rules, and may reject communications which matched entries in the operator and user allowed rules.

Incoming Communication Barring may reject communications which would be allowed by the Closed User Group Service.

This is shown in the following diagram:

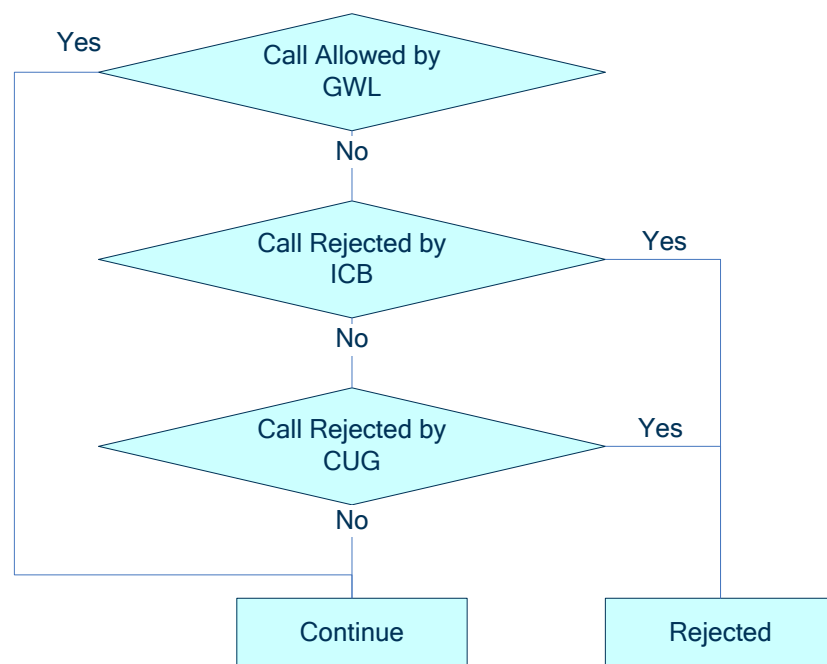


Figure 89: Communication Barring

2.52.3.2 Communication Diversion

The SIP INVITE transmitted as a result of a diversion contains all the CUG information present in the original INVITE message.

The Outgoing Communication Barring (OCB) service is suppressed when an initial INVITE request with CUG indication is diverted.



2.52.3.3 Flexible Communication Distribution

When an INVITE request with Closed User Group (CUG) indication is distributed by the Flexible Communication Distribution (FCD) service in the terminating MTAS the Primary User and the Related Users are handled differently from CUG perspective.

The CUG checks are run against the Primary User. When the CUG checks pass the INVITE request is forwarded towards the Primary User with the CUG indication removed. When the CUG checks fail the INVITE request is not forwarded towards the Primary User.

The CUG checks are not run against the Related Users. The INVITE request is forwarded with CUG indication towards the Related Users.

2.52.4 Provisioning

MTAS enables the operator to configure the service on user-level through the CAI3G interface. Possible settings are:

- Activate/deactivate the Closed User Group subscription
- Interlock code (unique identification) of the Closed User Group

2.52.5 Configuration

MTAS supports the following configuration:

- Administrative State
- Announcement when rejecting due to CUG

2.52.6 Performance Management

The following counters are provided by MTAS:

- The number of successful calls in a Closed User Group
- The number of INVITEs rejected by the Closed User Group (CUG) service.

2.52.7 Fault Management

For information on the alarm, refer alarm OPI [69].

2.53 Wholesale

2.53.1 Description

The Wholesale function makes it possible for an organization that owns and operates a telephony network to sell capacity to other operators that do not own and operate their own networks.



Both the organization that owns and operates the network, the Operating Telecom Provider (OTP), and their wholesale customers, the Virtual Telecom Provider (VTP), sell services to end users.

MTAS supports Wholesale by allowing the OTP to specify a set of VTPs, and determine which VTP a served user should belong to.

Each VTP is associated with a set of domains. A served user can be assigned to a VTP by mapping the host part of his primary PUI or by provisioning a VTP domain to the wholesale user.

Any user not associated with any VTP is treated as a user of the OTP.

The Wholesale function allows each VTP to create a VTP-specific configuration for the MTAS services. MTAS then applies this configuration when the service is provided to a subscriber of that VTP.

Each VTP can also choose to use the OTP values for the service configuration rather than configure the service itself.

The OTP remains in control of which services are provided by MTAS as the OTP administrative state of each service take precedence over the corresponding VTP administrative state.

The OTP-controlled per-VTP Dial Plan (see chapter 2.16) allows the operator to specify all the addresses that can be reached by the end-users belonging to each VTP.

To be able to use the Wholesale function, a valid license for Wholesale must be installed.

2.53.2 Operator Node Level Configuration

Configuration specified in this chapter pertains to data set via the O&M interface.

If there is no valid license for Wholesale, it is not possible to create or change any per-VTP MOC instances.

The overall MO structure is shown in Figure 89.

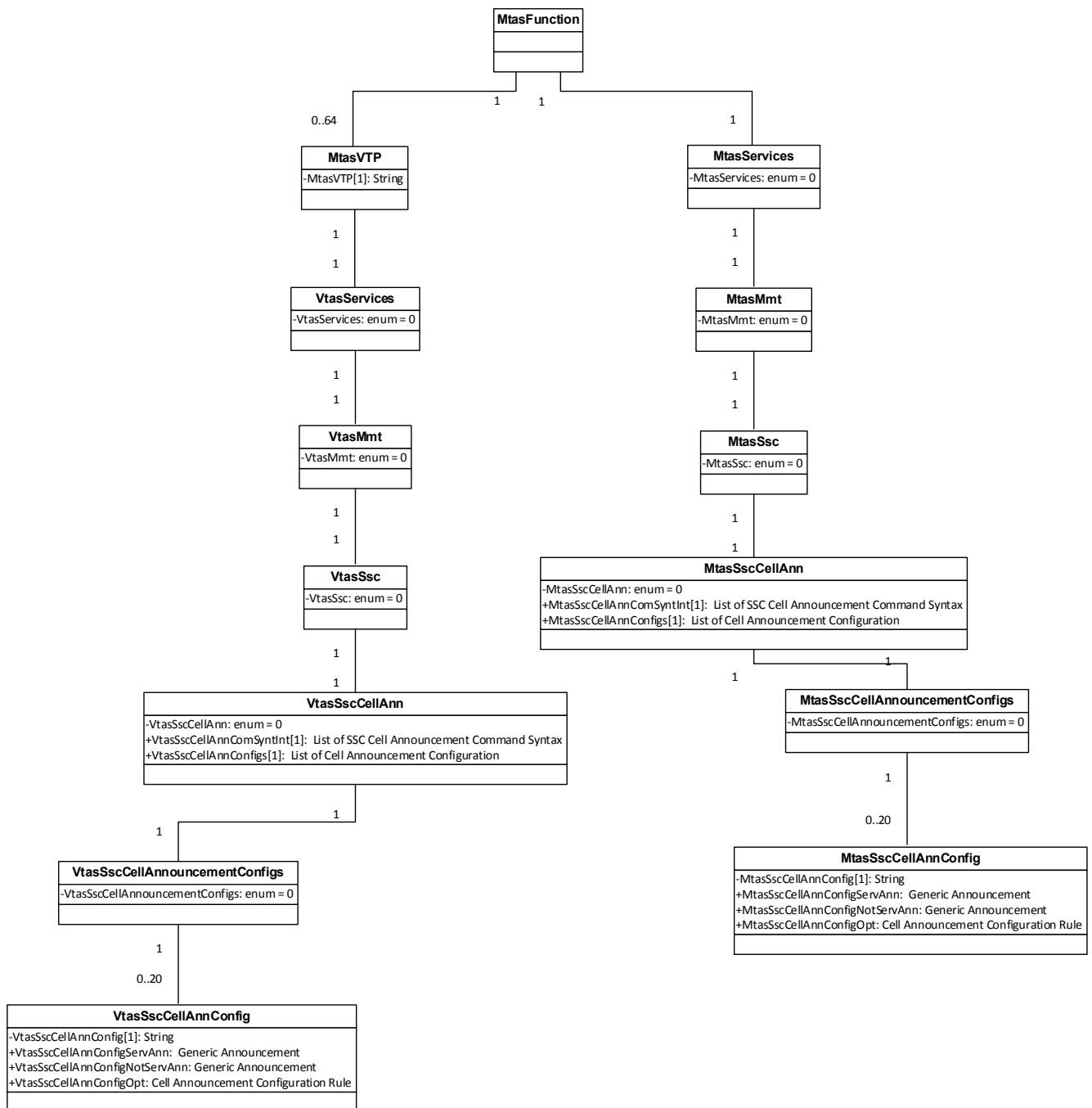


Figure 909 - MO Structure for Wholesale

Each MO instance in the structure is identified by its distinguished name (DN). This makes it possible to separate MOs for different VTPs.

E.g. The MTAS node is named “MTAS-1”, there is a VTP named “A” and this VTP has defined a Generic Announcement named “GA-1”, then the DN of this announcement is:



```
VtasGaAnn=GA-1, VtasGa=0, VtasMmt=0,  
VtasServices=0,MtasVtp=A, applicationName=MtasFunction,  
nodeName=MTAS-1
```

E.g. The MTAS node is named “MTAS-1”, there is a VTP named “A” and the OTP has defined an OTP controlled, per-VTP dial plan for this VTP, the DN of this dial plan is:

```
MtasDpv=A, MtasDialPlan=0, MtasMmt=0, MtasServices=0,  
applicationName=MtasFunction, nodeName=MTAS-1
```

The list of functions and services subjected to the Wholesale:

- Basic Voice and Multi-media communication
- Hold communication
- Communication Diversion
- Communication Waiting
- Three Party Call
- Ad-hoc Conference
- Explicit Communication Transfer
- Session Transfer to Own Device
- Dial Plan
- Communication Barring
- Malicious Communication Identification
- Dynamic Black List
- Abbreviated Dialing
- Identity Presentation
- Number Portability
- Call Admission Control
- Flexible Service Format Selection
- Video Fallback to Audio
- Priority call
- Calling Party Category
- Network Announcement
- Flexible Communication Distribution



- Call return
- Hotline
- Address Policing
- Short Number Dialing
- Number Translation service
- Advice of Charge
- Closed User Group
- Multi Subscriber Number
- Self Administration via SSC
- Operator Controlled Transfer

2.53.3 Charging

When the Wholesale function is enabled, and a served user is identified to belong to a VTP, the charging messages will include the Tenant AVP, which contains the name of the VTP.

2.54 Subscriber Credit Notification

2.54.1 Description

Subscriber Credit Notification (SCN) in MTAS is a feature that enables monitoring of a served end users credit state and playing announcements on credit state degradation, before establishing a session, during a session and when terminating a session.

The credit states supported are

- Credit ok
- Credit low
- Credit very low
- No credit

Subscriber Credit Notification is based on MTAS Online Charging Function and the interaction with an Online Charging System (OCS).

Subscriber Credit Notification is also responsible for playing Online Charging System initiated announcements.



The Notify Subscriber Credit function consists of a credit state machine to keep track of the credit state for the served user and a service that acts on credit state changes, controls pre-session and mid-session announcements, the interaction with other services and terminates the call if needed. The credit state machine is activated for the served user and updated at session setup and during session on reception of CCA from the OCS and when the final granted service units are consumed.

The Notify Subscriber Credit function is indicated when the credit state has degraded and a credit announcement is to be played to the served user, and if the session is to be terminated due to credit limit reached. Credit state changes may be detected at session setup and during established session. As a result announcements are played pre-session, mid-session and at session termination

OCS initiated announcements should be played when Announcement-Instructions AVP is received in CCA message on Ro interface. There are three types of such announcements: pre-quota, post-quota and prompt-and-collect.

2.54.2 Main scenarios

2.54.2.1 Notify subscriber credit

- Play pre-session announcements
 - No credit at session setup
 - Credit warning at session setup
 - OCS initiated announcement at session setup
- MTAS returns the VXML file containing a chained announcement to MRF
 - Credit warning at session setup
 - Credit warning during session
 - Connect and play mid-session announcement
 - OCS initiated pre-quota announcement during session
- Play session termination announcement

2.54.2.2 Terminating OCS initiated final announcements towards a caller

Terminating MTAS will play the announcement specified by OCS to the caller, when OCS denies an incoming call to be terminated to any target, so that the caller is notified that the call denial reason lies on subscriber's side and is not related to any problem with operator's network.



2.54.2.3 Originating OCS initiated retarget

When Retarget function is activated MTAS will retarget the original call to the destination address specified in the initial CCA message. The destination can either be in Final-Unit-Indication AVP or in Retarget-Instruction AVP.

When FUI is present with action to retarget the session and OCS did not granted service units for subscriber, the destination is usually a top-up server where user can refill his or her account.

When RI is present the user's session is usually retargeted to a generic service node.

If both FUI and RI are present, the address specified in FUI will take precedence.

Retarget function will halt the progression of the call setup towards the original called party.

Any call type classification of the original call will be kept for the new destination. Call type classification is not done as part of the retarget function.

If the Announcement-Instruction AVP is present in the CCA an announcement is played to the user before the call is retargeted to the new destination.

2.54.3 Service Interactions

2.54.3.1 Communication Diversion

For the diverting MTAS in Communication Diversion (CDIV) function, no credit announcements are played to the "diverted to" C-party.

2.54.3.2 Flexible Communication Distribution

For the distributing MTAS in Flexible Communication Distribution (FCD), credit announcements for the served user B is only played when call has been established to the IMS primary user (B).

2.54.3.3 Conference

An MTAS node acting as Conference Creator monitors credit state changes for the conference creator (A) and plays the following announcements: no credit", pre-session and mid-session credit warning, credit limit reached.

The conference participants remains connected to the conference and the conference is put on hold (inactive) whilst credit announcement is played to the conference creator (A).

2.54.3.4 Three party

An MTAS node providing the Three Party (3PTY) service monitors the three party session for the served users credit state and play the following announcements to the served user: credit limit, pre-session and mid-session credit warning, credit limit reached.



The other users remain connected and the three party conference is put on hold whilst a credit announcement is played to the served user.

2.54.3.5 Hold

In addition to the direct use of Hold by the Subscriber Credit Notification function for mid-session announcements, credit warning announcements are not played to a user when the users SDP does not allow to, i.e. when the users media streams are inactive or send-only. The announcement is played later in the session when the users SDP has been changed to allow announcement (send-receive).

2.54.3.6 Communication Completion

Credit limit announcement is played to the served user if the Communication Completion (CC) call cannot be established due to credit restrictions.

Credit warning announcement is played to the served user when it is indicated that the user's credit is low or very low when establishing a CC call. The announcement is played after CC call has been successfully established.

2.54.3.7 Explicit Communication Transfer

If the credit limit is reached for the served user A during an Explicit Communication Transfer (ECT), the credit limit reached announcement is played to the transferor (A), the sessions A-C and A-B are restored and the transfer is rejected.

Credit warning indications during a call transfer is ignored without announcement.

Since the served user is not in the call anymore after a successful call transfer, no announcement will be played. Credit warnings are ignored, but when the served user runs out of credits, the call will be ended.

2.54.4 Configuration

MTAS supports the following configuration with CM attributes:

On Node level:

- enabling/disabling the SCN function
- announcements mappings configuration

On Charging Profile level:

- announcements names
- warning delay



When OCS initiates announcement playing then the contents of Announcement-Instructions AVP should be mapped to an MTAS generic announcement.

2.54.5 Performance Management

The following PM counters applies to Subscriber Credit Notification:

- MtasChargingCcaLowBalanceIndication
- MtasChargingCcaFinalUnitIndication
- MtasChargingCcaCreditLimit

2.55 Voice Mail Service

2.55.1 Description

The Voice Mail (VM) service in MTAS provides common control for voice-mail deposit and retrieval targets, for all services where a voice-mail target is relevant as described above, for either deposit or retrieval of voice-mail messages. Provisioning of targets is done on subscriber level, and may point to node level configurations for voice-mail deposit server address and voice-mail retrieval server address, respectively.

The VM service also provides a node-level facility for all subscribers, irrespective of provisioning, to access deposited voice-mail messages by calling their own number, called Self Call to Voice Mail. This is controlled by CM attribute `mtasVoiceMailRetrievalOnSelfCall`.

2.56 SSC Cell Announcement

2.56.1 Description

Cell Announcement feature allows a subscriber to dial a feature code and receive a dynamic announcement providing the identification of the serving femtocell.

In order to simplify services that use originating location information, the P-CSCF will map the femtocell ID received in the `utran-cell-id-3gpp` access-info of the PANI header to the associated macrocell ID using mapping tables. The P-CSCF will then set the `utran-cell-id-3gpp` access-info to the macrocell ID in the PANI header of the INVITE that it sends over the ISC interface to other IMS core network elements including MTAS. The P-CSCF will provide the femtocell information in a new parameter, called `femto-utran-cell-id-3gpp`, added to the PANI header.

P-Access-Network-Info:3GPP-E-UTRAN-FDD;utran-cell-id-3gpp=<macro>;femto-utran-cell-id-3gpp=<femto>



MTAS plays a location dependent announcement to the calling user triggered by a supplementary service code command. This SSC service is performed on Originating MTAS MMTel AS.

The location dependent announcement service will identify the radio access cell that user is being served by in order to tell that during communication with the operator's customer care.

When the User dials SSC command for cell announcement service, MTAS plays cell served announcement if matching PANI header found.

- Cell Served Announcement Example
 - *"You are currently being served by Femtocell ID < xxxx>"* where xxxx is the variable part of the announcement containing the decimal value of the last seven digit of "femto-utran-cell-id-3gpp" parameter.

When the User dials SSC command for cell announcement service, MTAS plays cell not served announcement if matching PANI header not found.

- Cell Not Served Announcement Example
 - *"You are not currently being served by a Femtocell".*

When the User dials SSC command for cell announcement service, MTAS plays SSC negative announcement in error scenarios like no PANI header received in INVITE request. Playing of negative announcement will be as per legacy SSC service behavior.

2.56.2 Main Scenario

Served User dials SSC code i.e. *48# for cell announcement service interrogation. MTAS receives INVITE with PANI header including femto-utran-cell-id-3gpp access info. MTAS SSC service parses the PANI header, play segmented cell served announcement to Served User using External MRFC. After announcement completion MTAS SSC service generates online charging message and increment PM counter for successful cell announcement service interrogation.

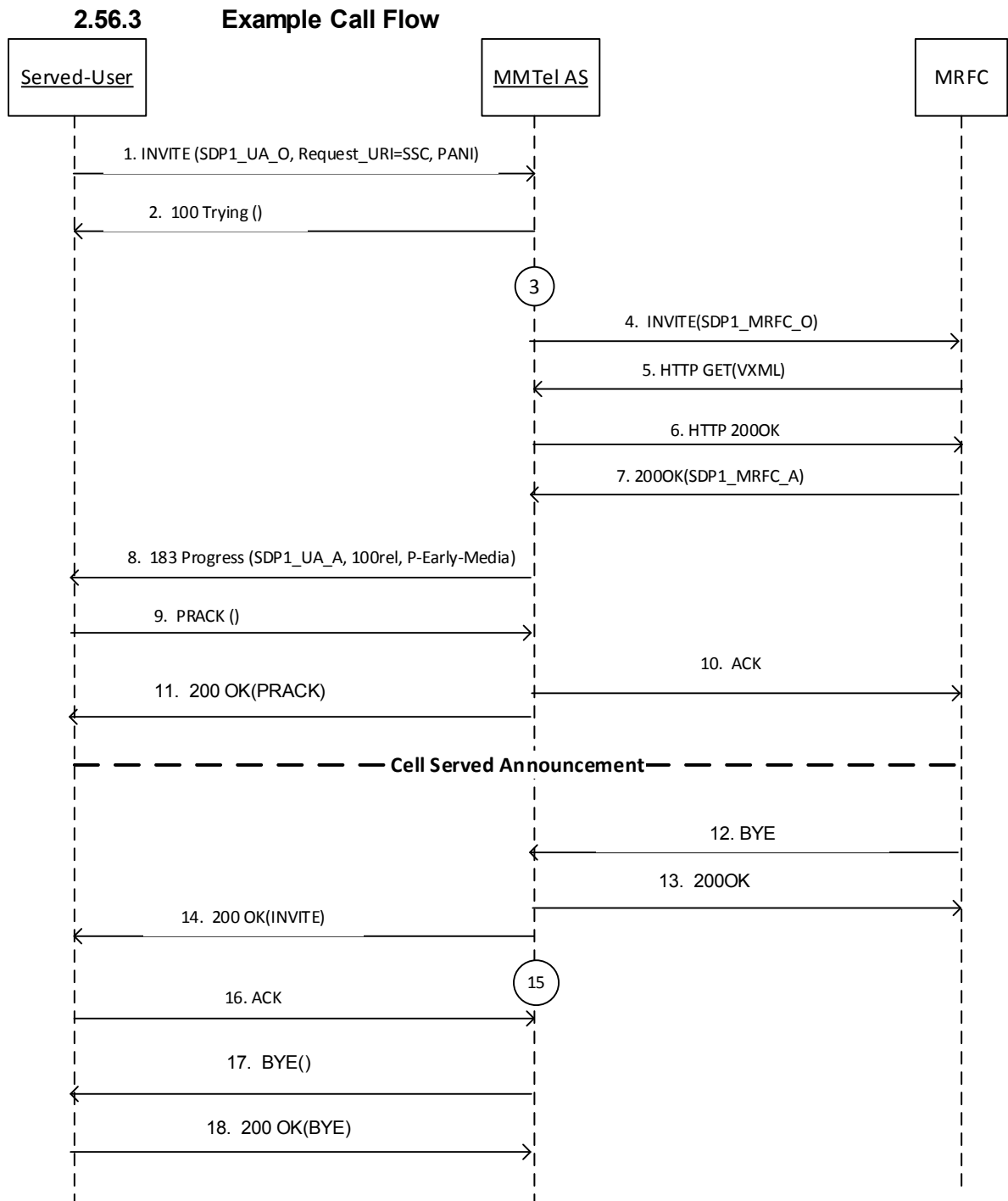


Figure 90: Cell Served Announcement via EXTERNAL MRFC

1. An INVITE request message is received in the MTAS as a consequence of the user initiated a service code command for the interrogation of a cell announcement supplementary service. The message includes:
 - Request-URI that contains service code command for interrogation of the service. E.g. *48#
 - P-Access-Network-Information that contains *femto-utran-cell-id-3gpp*.



- *E.g. P-Access-Network-Info: 3GPP-E-UTRAN-FDD;
utran-cell-id-3gpp=3114809600002583d;
femto-utran-cell-id-3gpp=3114803a0500bbc02*
 - P-Asserted-Identity, that contains public identity of the subscriber (user) issuing the service code command
 - An SDP offer for a voice call.
 - Support for reliable provisional responses, 100rel, indicated
2. MTAS responds with TRYING message towards the UA.
 3. MTAS extract Cell ID from PANI header and prepare announcement string as per configuration.
 4. MTAS sends INVITE to MRFC with SDP received from UA.
 5. MRFC sends a HTTP GET request to MTAS to the URL received in the INVITE
 6. MTAS returns the VXML file to MRFC.
 7. MRFC responds with 200 OK to MTAS with the SDP answer.
 8. MTAS sends the SDP answer to UA in a 183 Session Progress.
 9. UA responds with a PRACK.
 10. MTAS requests MRFC to process the VXML file by sending the ACK.
 11. MTAS sends 200 OK for the PRACK.
- The announcement is now played.
12. MRFC sends a BYE to MTAS including the invocation result.
 13. MTAS responds with a 200 OK for the BYE to the MRFC. MTAS deletes the VXML file
 14. MTAS send 200OK for the INVITE to UA
 15. SSC service will generate the online charging message with SSID specific to SSC Cell Announcement Interrogation service. It is as per legacy SSC design.
- Increment MtasSSCodesIntCellAnnOk PM Counter.
16. UA sends ACK.
 17. MTAS sends BYE to UA.
 18. UA sends 200OK for the BYE to MTAS.



2.56.4 Configuration

- Configure mtasSscCellAnn MOC for,
 - operator defined SSC command syntax for cell announcement service
 - Cell announcement configuration for various operator defined dialed phone number for cell announcement service.
- Configure mtasSscCellAnnConfig MOC for each operator defined dialed phone number for cell announcement service. This contains below configuration,
 - Cell served announcement
 - Cell Not served announcement
 - Cell ID fetching rule from INVITE PANI header.
- Configure mtasGaAnn for below,
 - Segmented Announcement vector for cell served announcement
 - Segmented announcement vector for cell not served announcement

2.56.5 Performance Management

The following 3 PM counters applies:

- MtasSSCodesIntCellAnnOk: The counter is incremented by 1 when the Cell Served Announcement completed successful
- MtasSSCodesIntCellAnnNOK: The counter is incremented by 1 when the Cell Not Served Announcement completed successful
- MtasSSCodesIntCellAnnErr: The counter is incremented by 1 when SSC Negative Announcement for cell announcement service completed successfully or no announcement is played due to error

2.57 Long Duration Call Supervision

2.57.1 Description

Long Duration Call Supervision service in MTAS is a feature that enables MTAS to supervise long duration call. MTAS starts Long Duration Supervision Timer on call establishment. On Long Duration Supervision Timer expiry, MTAS terminates the call and set reason for disconnect as “Long Duration Call” in reason header of BYE Message.



Long Duration Call Supervision service is handled on the originating and terminating MTAS node .

It shall be noted that for Long Duration Call Supervision service will be defined and configured by operator. Long Duration Supervision Timer value can be configured differently for following:

1. Originating call : Operator can configure different Long Call Duration Supervision timer value for following category:
 - a. Default Originating Supervision timer: This timer controls the originating call supervision. No long duration call supervision for any originating call if this timer value is zero. This timer is applicable only when other Originating Supervision timers are not applicable
 - b. Service Number specific Originating Supervision timer: This timer is applicable when destination number is identified as service number. No long duration call supervision, when destination number is identified as service number and this timer value is zero. Following are the examples of service numbers:
 - i. Destination classified as OSN/NSN by number normalization configuration
 - ii. DNM Location based short code
 - iii. National short code classified by Number Translation
 - iv. Directory assistant (OCT) call
 - c. Operator defined destination categories: specific Originating Supervision timer: This timer is applicable when destination number is matched with operator defined destination category. No long duration call supervision, when destination number is matched with operator defined destination category and this timer value is zero.

On receiving an INVITE, MTAS will try to match req-uri in following order with:

- Service number category
- Operator Defined destination category

If match found then corresponding timer value of that category is applicable.

If match did not found then Default originating supervision timer value is applicable



2. Terminating call:

- a. Default Terminating Supervision timer: This timer controls the terminating call supervision. No long duration call supervision for any terminating call if this timer value is zero

On receiving an INVITE, Default Terminating supervision timer value is applicable

Long Duration Call Supervision service for all originating call can be disabled by setting Default Orig Supervision timer value to 0.

- If operator wants disable, Long Duration Call Supervision Service only for Service number, then corresponding service number timer value will be set to 0.
- If operator wants disable, Long Duration Call Supervision Service only for Operator Defined destination categories then corresponding Operator specific destination categories timer value will be set to 0.

Long Duration Call Supervision service for Terminating call can be disabled by setting Default Term Supervision timer value to 0.

2.57.2 Main Scenario

2.57.2.1.1 Originating long duration call supervision

An operator wants to supervise long duration calls for an Originating call. On receiving INVITE, MTAS will identify the originating long duration call supervision timer value by matching the normalized Request URI of an INVITE request with configured values. On reception of ACK from calling party, MTAS should start the supervision timer.

On supervision timer expiry, MTAS should disconnect the call by sending BYE to involved parties. Otherwise if call is terminated by involved parties or due to network issue MTAS should stop the supervision timer.

2.57.2.1.2 Terminating long duration call supervision

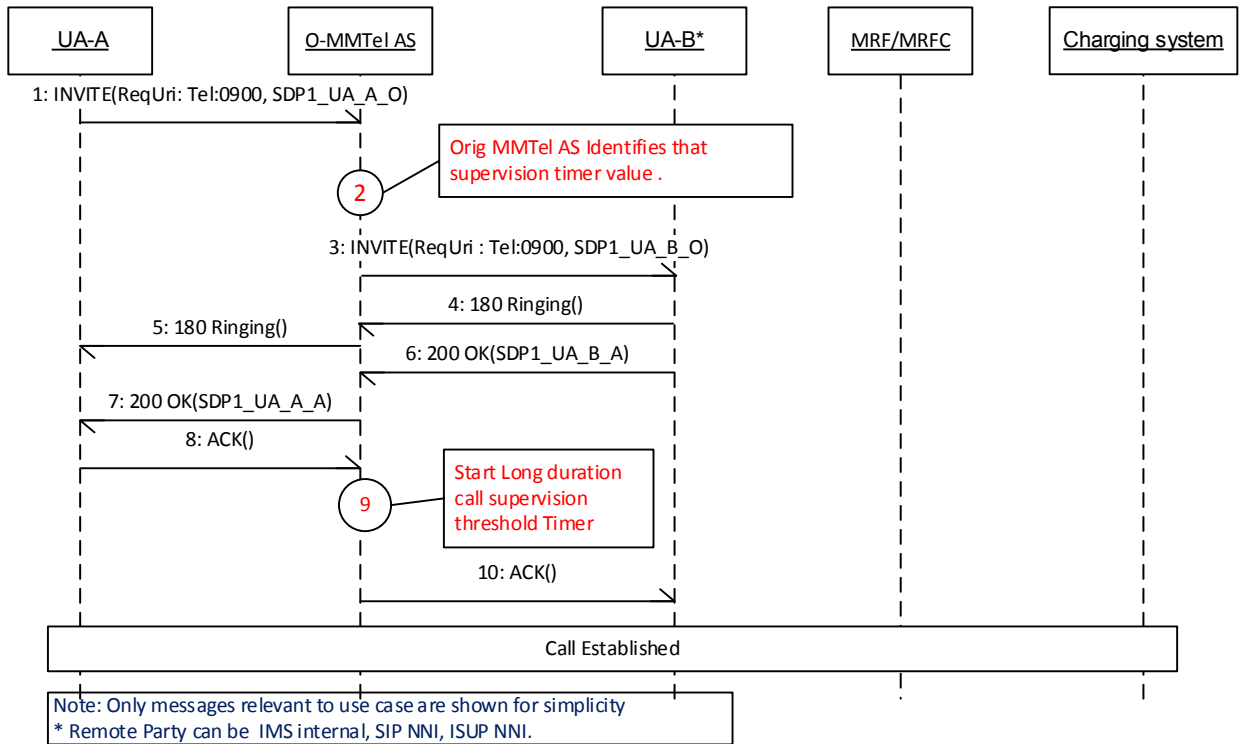
An operator wants to supervise long duration calls for Terminating call. On reception of ACK from calling party, MTAS should start the default Terminating supervision timer. For all Terminating call, Only default terminating supervision timer will be applicable.

On supervision timer expiry, MTAS should disconnect the call by sending BYE to involved parties. Otherwise if call is terminated by involved parties or due to network issue MTAS should stop the supervision timer.



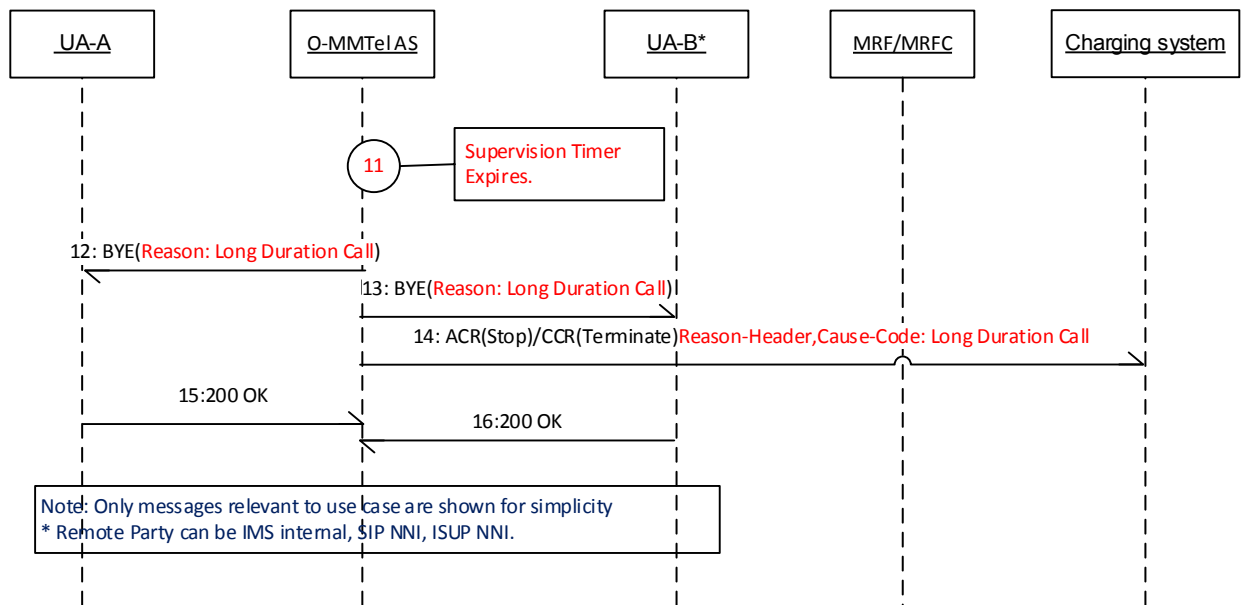
2.57.3 Example Call Flow

2.57.3.1.1 Originating long duration call supervision



Step-2: Originating MTAS receives INVITE message. MTAS checks if normalized request URI of INVITE UA-B (Called Party Number) is service number. If Called party number is not service number then Originating MTAS matches it with configured operator defined destination categories. If normalized request URI did not match configured operator defined destination categories then default originating supervision timer value will be applicable.

Step-9: On reception of ACK from calling party, originating MTAS starts selected originating supervision timer.



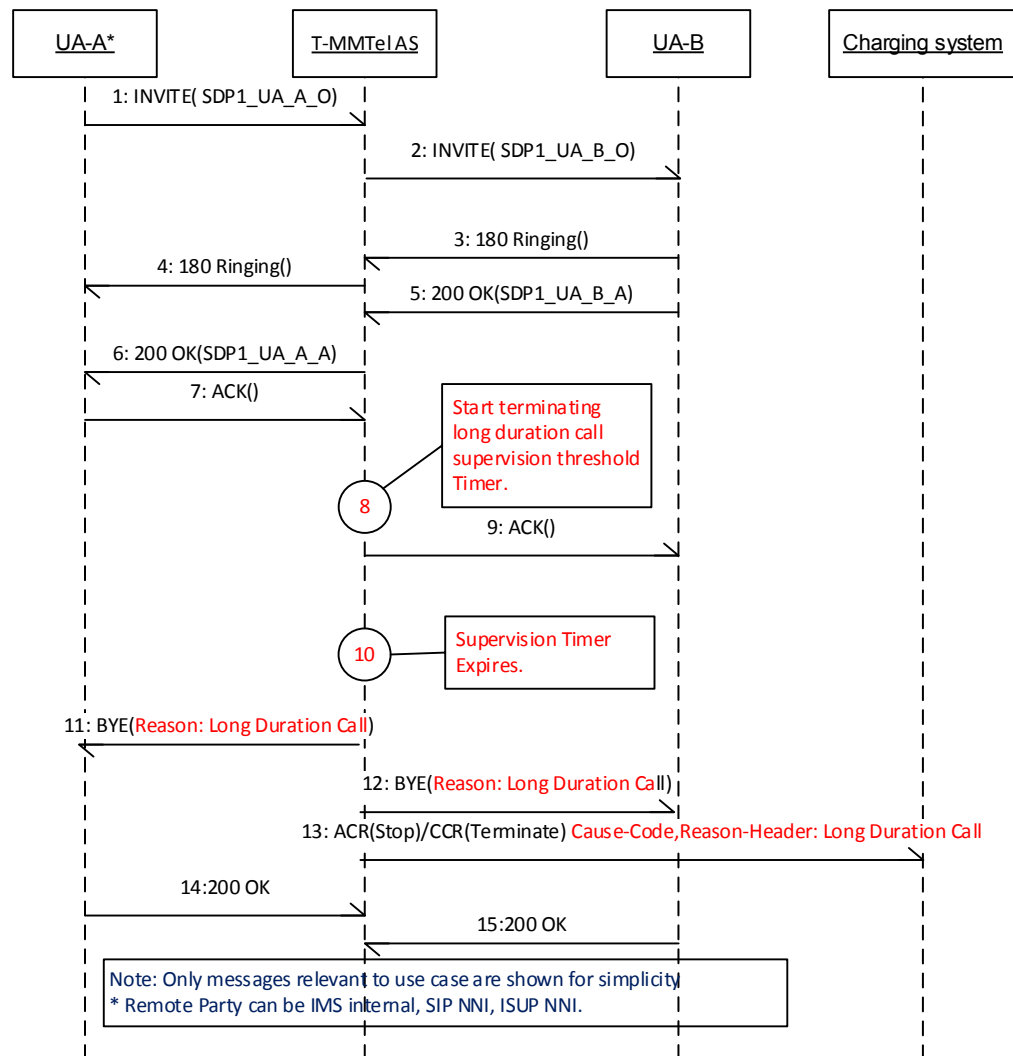
Step 11: On Supervision timer expiry. Increment MtasMmtLongDurationCallOk OK PM Counter with Key set to “Orig”.

Step 12: Originating MTAS terminates call towards UA-A by sending BYE with Reason Header set to “Long Duration Call”.

Step 13: Originating MTAS also terminates the call towards UA-B by sending BYE with Reason Header set to “Long Duration Call”.

Step 14: Originating MTAS sends charging message (ACR Stop/CCR Terminate) with Reason-header and Cause-Code AVP set to “Long Duration Call”.

2.57.3.1.2 Terminating long duration call supervision



Step 8: On reception of ACK from User UA-A, Terminating MTAS starts default terminating long duration supervision timer.

Step 10: On supervision timer expiry. Increment MtasMmtLongDurationCallOk OK PM Counter with Key set to "Term".

Step 11: Terminating MTAS terminates call towards UA-A by sending BYE with Reason Header set to "Long Duration Call".

Step 12: Terminating MTAS also terminates the call towards UA-B by sending BYE with Reason Header set to "Long Duration Call".

Step 13: Terminating MTAS sends charging message (ACR Stop/CCR Terminate) based on charging configuration with Reason-Header and Cause-Code AVP set to "Long Duration Call".



2.57.4 Charging

When call is disconnected by long call supervision Service, Reason-Header AVP and Cause-Code AVP in ACR(STOP) and CCR (Terminate) charging message will be populated as follows :

- Cause-Code AVP = "Long Duration Call" "-1001"
- Reason Header AVP = "Long Duration Call"

2.57.5 Service interactions

2.57.5.1 Call Diversion (CDIV)

On receiving INVITE for UA-B on Terminating MTAS and UA-B is configured for Call Diversion service, Terminating MTAS will execute CDIV service and diverts the communication to UA-C. When call is accepted by UA-C, Long duration Call Supervision will be performed on established session as per configured value.

2.57.5.2 Flexible Communication Distribution (FCD)

On receiving INVITE for UA-B on Terminating MTAS and UA-B is configured for FCD service, FCD service will distribute the communication to Primary User and Related User. When call is accepted by any user, Long duration Call Supervision will be performed on established session as per configured value.

2.57.5.3 Session Transfer to Own Device (STOD)

Before STOD service Execution, supervision timer must be running for an established session. On STOD execution session will be transferred to related user or primary user and it will not impact the active running supervision timer. On STOD execution, If transferred call is accepted by related user only then additional New Originating supervision will be started.

2.57.5.4 Operator Controlled Transfer (OCT)

An originating MTAS receives an INVITE with operator configured phone number in request URI from Caller UA-A. By checking the normalized Request URI (Operator Transferor, OT) of an outgoing INVITE, an originating MTAS identifies the originating supervision timer value.

Later on when the call is accepted by OT, an originating MTAS of UA-A, starts supervision timer. Further during OCT service execution, BYE will be sent towards OT, and stops originating supervision timer. OCT service will send new INVITE towards redirected user UA-C and starts new originating supervision timer by considering the redirected destination.



2.57.5.5 Ad-Hoc:

An MTAS matches the normalized Request URI of outgoing INVITE request for each Focus-CP (Conf participants) Call leg with configured destination categories and identifies the originating long duration call supervision timer value. MTAS will stop the supervision timer for each Focus-CP call leg on timeout or on call termination by CP or CC or due to network issue. On timeout Mtas should disconnect the particular call Focus-CP call leg by sending BYE to respective participant on supervision timer expiry and add reason of disconnect in charging message as well as in BYE.

2.57.5.6 3PTY :

The service allows a user who is involved in two separate 2-party sessions with another two users to bring these two users into a 3PTY session. Before 3PTY Call execution, supervision timer must be running for established sessions between

- Session 1 between UA-A and UA-B
- Session 2 between UA-A and UA-C

3PTY Call execution should not impact the active supervision timers.

MTAS will stop the supervision timer for session on timeout or on call termination in respective session or due to network issue. On timeout Mtas should disconnect the particular session by sending BYE to respective participant on supervision timer expiry and add reason of disconnect in charging message as well as in BYE.

2.57.6 Configuration

MTAS supports the following configuration:

- MtasMmtLongDurationCall:

Configure for the Long Duration Call Supervision service in an MTAS node.

- MtasMmtLongDurationCallOrig:

Configure for the Long Duration Call Supervision service in an MTAS node for originating call. This contains below configuration:

- mtasMmtLongDurationCallOrigTimer: Configure for Default Originating Supervision timer
- mtasMmtLongDurationCallOrigServiceNumberTimer: Configure for Originating Supervision timer for Service number
- mtasMmtLongDurationCallOrigDestCat: Used to define the set of destination category configurations

- MtasMmtLongDurationCallDestCats:



Configure Structure to define entry for all MtasMmtLongDurationCallDestCat.

- MtasMmtLongDurationCallDestCat:

Configure for destination category configuration for long duration originating call supervision. This contains below configuration:

- mtasMmtLongDurationCallDestCatList: Defines the global destination category list applicable for long duration call supervision in Node.
- mtasMmtLongDurationCallDestCatTimer: Define long duration supervision timer specific to destination category.

2.57.7 Performance management

The following PM counters applies

MtasMmtLongDurationCallOk: The counter is incremented unconditionally when long duration supervision timer expired. This is keyed counter with possible values are Orig, Term and Dest_Orig.

2.58 Dialog Event Notifier service

2.58.1 Description

The MMTel AS allows “out-of dialog” subscription to Dialog Events as defined in RFC 4235. Each device sends an explicit SUBSCRIBE request to the subscriber's own number using the “dialog” event package. Devices refresh the subscription periodically as specified in RFC 6665.

MMTel AS maintains separate subscriptions for each device and sends NOTIFY requests whenever:

- an originating or terminating call is answered.
- a call is pulled from one device to another.
- the media state of the call changes (e.g.: the media changes from active to hold state or from audio only to audio and video).
- a call is terminated.

Each NOTIFY includes information about all the calls in progress at all of the other devices but does not include information about calls currently handled by the recipient of the NOTIFY (NOTIFY only reports calls at OTHER devices).



For each call, the following information is included in the NOTIFY that is sent from the MMTel AS to each device:

- Dialog identifiers: Local tag, Remote tag and Call ID of the dialog from the MTAS toward the IMS network.
- Call state: “confirmed” - Call has been answered.
- Direction: Indicates if this is an originating or terminating call.
- +sip.rendering parameter: It set to “no” if the call is on hold.
- local <identity>: The element contains a “gr” parameter that contains the +sip.instance ID.
- <exclusive>: The element indicates (as specified in RFC 7463 - Shared Appearances of a Session Initiation Protocol) if the call is pullable (true) or not (false).
- Media Information: information for each SDP media line is included indicating whether the media is sendrecv, sendonly, recvonly, or inactive, and also indicating whether the media is audio, video, text, image, or other. <port0> indicates a downgraded video call (call downgraded from video to voice, or video call answered as voice)

NOTE: Media information includes media direction after negotiation is completed. If media is in negotiation phase when the pull request is received then the request will fail.

2.58.2 Example call flow

In this scenario the subscriber registers three devices. All three devices subscribe to dialog event. Two of the devices make a call and release it. Various notifications are sent to each device as shown in Figure 91.

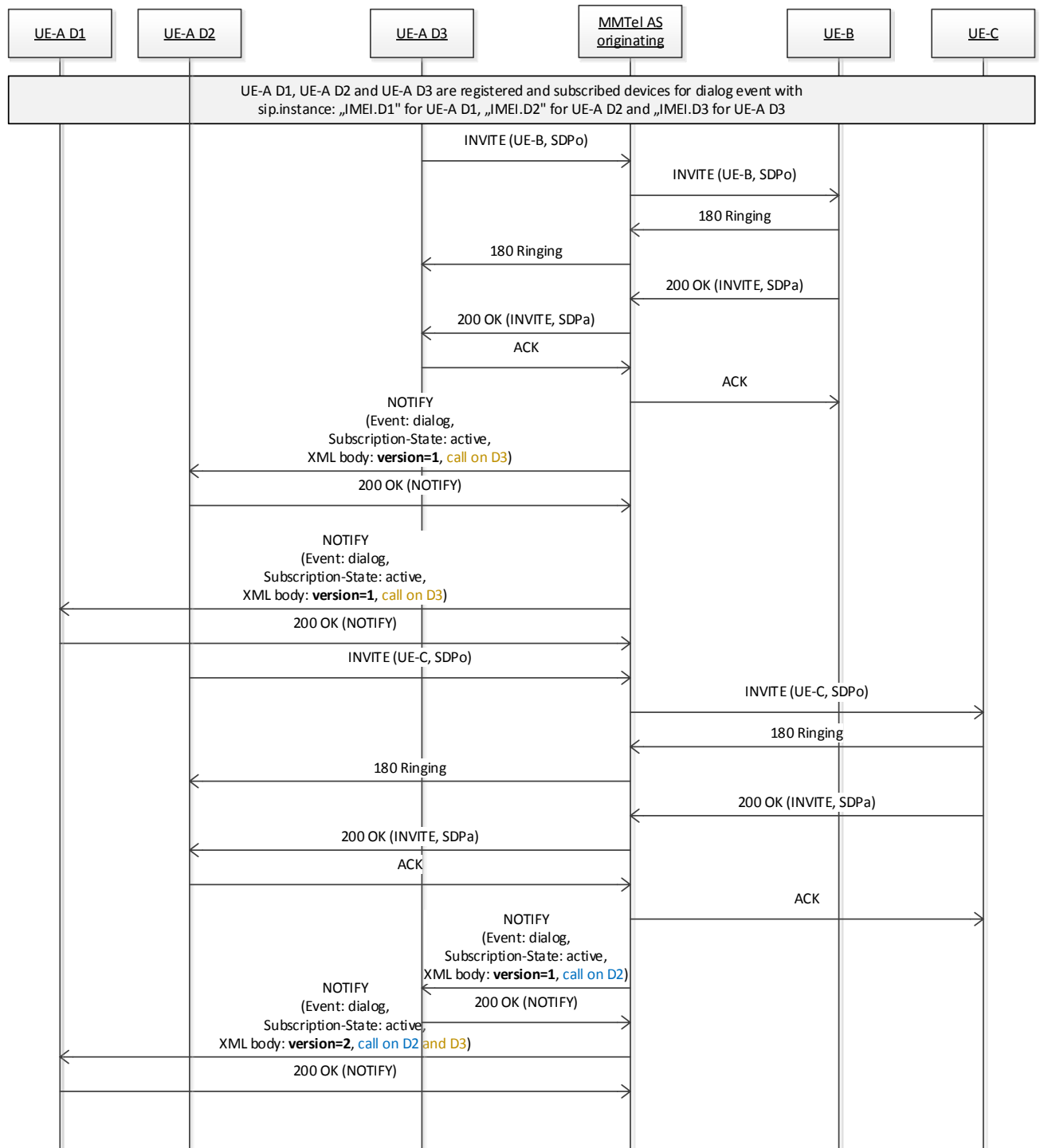


Figure 91 Call setup and notification about multiple calls

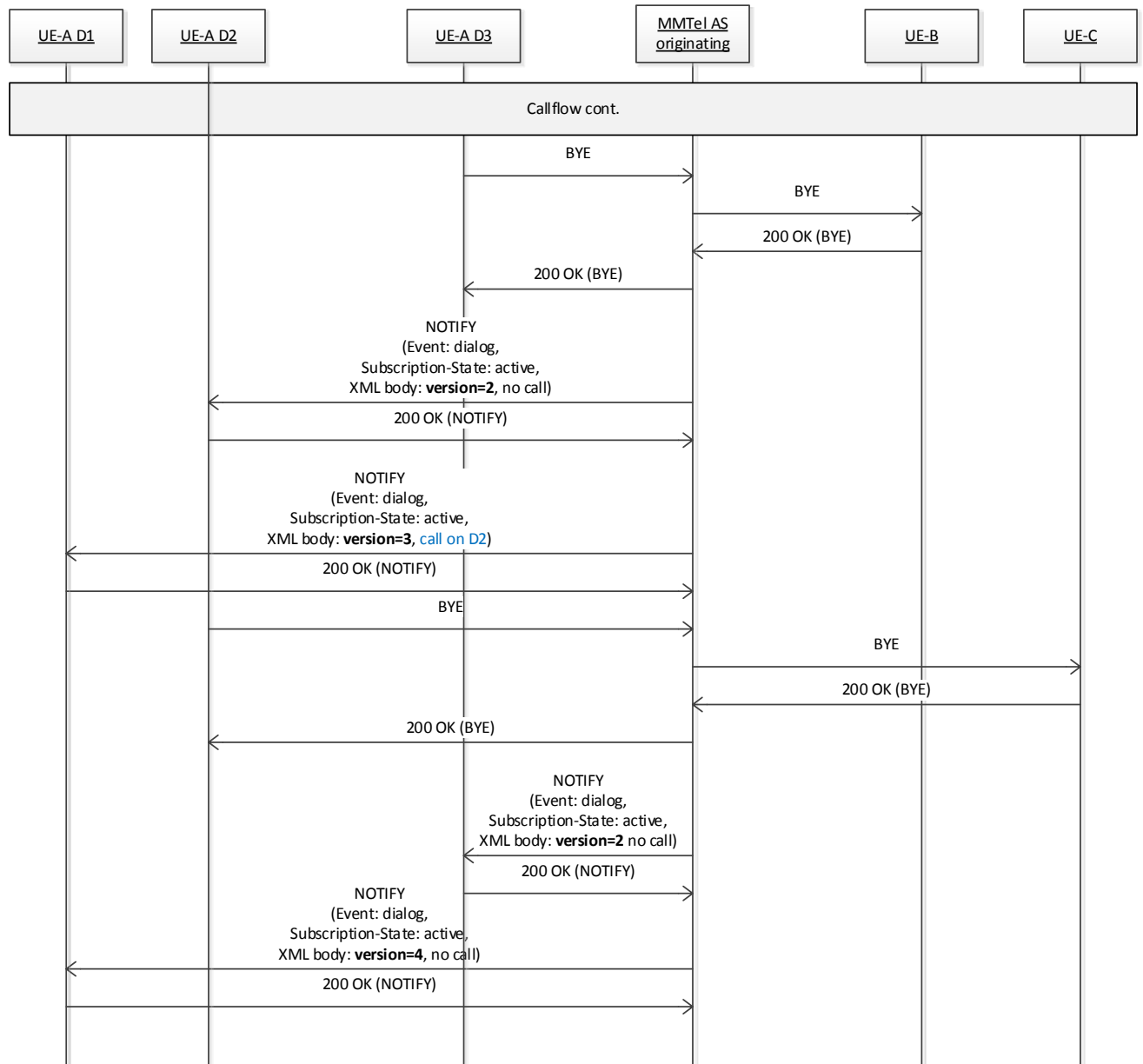


Figure 92 Call release and notification of multiple calls

2.58.3 Charging

The use of Dialog Event Notifier service is not reported in charging messages.



2.58.4 Configuration

OAM operator can configure the following node-level parameters:

- administrative state of DEN service
- minimum, maximum and default values for the subscription expiration timer
- presence of <exclusive> element in XML document of NOTIFY requests
- presence of <mediaAttributes> element in XML document of NOTIFY requests
- display name for Ad-hoc conference calls
- display name for Voicemail call
- maximum number of subscription sessions per subscriber

2.58.5 Performance management

The following performance counters are provided by MTAS.

2.58.5.1 Node level gauge

MtasFuncOngoingSubsSess

Description: The number of currently active subscription sessions on node level.

Condition: The gauge is incremented when a subscription session is created and decremented when a subscription session is released.

2.58.5.2 MMTel AS counters

MtasMmtSubsSessOk

Description: The number of successfully created out-of-dialog subscription sessions within the MMTel AS.

Condition: The counter is incremented by 1 in the originating MMTel AS when responding with 200 OK to a received initial out-of-dialog SUBSCRIBE.

MtasMmtSubsSessNOkE

Description: The number of unsuccessfully created out-of-dialog subscription sessions within the MMTel AS due to node external reason.



Condition: The counter is incremented by 1 in the originating MMTel AS when responding to a received initial out-of-dialog SUBSCRIBE failed due to node external reason.

MtasMmtSubsSessNOkl

Description: The number of unsuccessfully created out-of-dialog subscription sessions within the MMTel AS due to node internal reason.

Condition: The counter is incremented by 1 in the originating MMTel AS when responding to a received initial out-of-dialog SUBSCRIBE failed due to node internal reason.

MtasMmtSubsSessAttempt

Description: The number of out-of-dialog subscription session creation attempts within the MMTel AS.

Condition: The counter is incremented by 1 in the originating MMTel AS when receiving an initial out-of-dialog SUBSCRIBE.

2.58.5.3 Service level counters

MtasDenSubsSessOk

Description: The number of successfully created 'dialog' event package subscription sessions.

Condition: The counter is incremented by 1 by the Dialog Event Notifier service when 200 OK is sent to the SUBSCRIBE request of a 'dialog' event package subscription.

MtasDenSubsSessNOkE

Description: The number of unsuccessfully created 'dialog' event package subscription sessions due to node external reason.

Condition: The counter is incremented by 1 by the Dialog Event Notifier service when responding to the SUBSCRIBE request of a 'dialog' event package subscription failed due to node external reason.

MtasDenSubsSessNOkl

Description: The number of unsuccessfully created 'dialog' event package subscription sessions due to node internal reason.

Condition: The counter is incremented by 1 by the Dialog Event Notifier service when responding to the SUBSCRIBE request of a 'dialog' event package subscription failed due to node internal reason.



2.59 SIP Upstream Overload Control

2.59.1 Description

The SIP Upstream Overload Control in MTAS implements the reporting/server role of the RFC 7339 “SIP Overload Control”, which describe the exchange of the overload information between the SIP servers and the SIP clients, so that the SIP clients can reduce the volume of traffic sent towards the overloaded SIP servers, avoiding congestion collapse and increasing useful throughput.

OC parameters are added by reacting node (CSCF) in the topmost Via header of the SIP INVITE request, which indicate Overload Control is supported.

Example:

```
Via: SIP/2.0/TLS p1.example.net
    ;branch=z9hG4bK2d4790.3;
    ;oc;oc-algo="loss";
```

When feature is enabled and OC parameters are in the request, MTAS adds OC parameters to the topmost Via header of the response.

Example:

```
Via: SIP/2.0/TLS p1.example.net
    ;branch=z9hG4bK2d4790.3;received=192.0.2.111
    ;oc=20;oc-algo="loss";oc-validity=1500
    ;oc-seq=1456387313.187
```

Details of each parameter are described in table below.

Table 2 SIP OC parameter

OC parameter	Content
oc	A value to indicate the level of traffic reduction to apply, i.e. “oc=0” if no traffic reduction is required (because the reporting node is not overloaded and e.g. “oc=20” when overload control active and requesting 20% traffic reduction. The value is calculated by MTAS based on current system overload status.
oc-algo	It is used to indicate which traffic reduction algorithms it supports, where “loss” currently will be the only supported one.
oc-validity	It indicates for how long duration in milliseconds the sent “oc” parameter shall be used by the reacting/client node. The default value can be configured by CM attribute mtasSipOcValidity.
oc-seq	It is used to differentiate two “oc” parameter values generated by an overload control algorithm at two different



	<p>instants in time.</p> <p>MTAS's generated "oc-seq" will be based on "epoch time" and will have the following format: [epoch seconds].[epoch milliseconds].</p>
--	---

If MTAS operates in normal condition, thus, the node is not overloaded then the same OC parameters with "oc=0; oc-algo="loss"; oc-validity=0; oc-seq=xxx" are reported in each response. The oc-seq here is not changed until the system enters the overload status.

If MTAS operates in overload condition, then the oc value changes between 1-100 and reported in a configurable interval. The oc value is calculated from the current system utilization level and from the previous oc status. The oc-validity can be configured by the operator and the oc-seq indicates the time stamp when the oc value has been calculated.

2.59.2 Example call flow

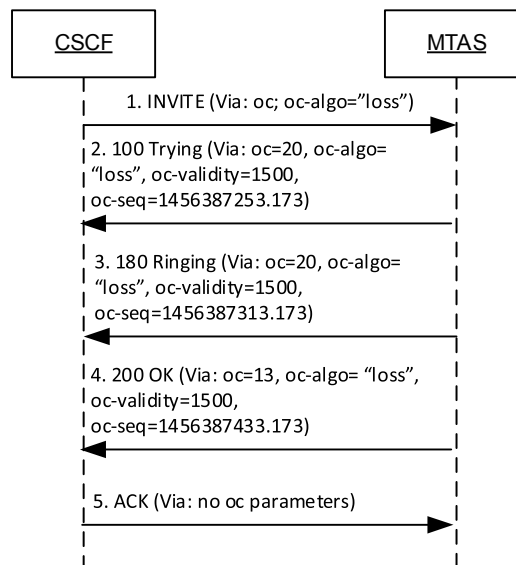


Figure 93 SIP Upstream Overload Control

- An "oc" parameter without any value as indication to the downstream neighbors that the client supports overload control.
- An "oc-algo" parameter to indicate which overload algorithms it supports, where "loss" is default and mandatory.



Note: the “oc-seq” will not update until `mtasSipOverRegulationInterval` expires, so in the time period, all SIP response will add the same OC parameters. Only Topmost Via header will be updated, MTAS will not make any update for the other Via headers.

3. After several seconds, the OC value is not changed, MTAS sends set of OC parameters with overload status (same oc value and new oc-seq) in topmost Via header in 180 Ringing.
4. After several seconds, the OC status changed, MTAS sends the set of OC parameters with overload status (new oc value and new oc-seq) in topmost Via header in 200 OK.
5. CSCF send ACK without oc parameters.

2.59.3 Configuration

OAM operator can configure the following node-level parameters:

The following table describes MOCs and CM parameters to be implemented.

Table 3 SIP Upstream Overload Control CM attributes

MOC	CM Attribute	Type	Range	Default value
MtasSipOc	mtasSipOcAdministrativeState	Integer-Enum	0=LOCKED, 1=UNLOCKED	0=LOCKED
MtasSipOc	mtasSipOcOnset	Integer-range	0-200	85
MtasSipOc	mtasSipOcAbatement	Integer-range	0-200	75
MtasSipOc	mtasSipOcDefIncrStep	Integer-range	0-100	12
MtasSipOc	mtasSipOcDefDecrStep	Integer-range	0-100	8
MtasSipOc	mtasSipOcValidity	Integer-range	500-10000	1500
MtasSipOc	mtasSipOcRegulationInterval	Integer-range	1000-10000	1000
MtasSipOc	mtasSipOcResource	String		

Description of CM attributes:

- `mtasSipOcAdministrativeState`: This attribute defines lock or unlock the support of adding OC parameters in Via header of SIP response for SIP Upstream Overload Control. Possible values are 0 (LOCKED) and 1 (UNLOCKED). Default value is 0 (LOCKED).



- **mtasSipOcOnset:** This attribute defines the threshold which MTAS increments the reported 'oc' value periodically while the resource utilization level is above it.
- **mtasSipOcAbatement:** This attribute defines threshold which MTAS decrements the reported 'oc' value periodically while the resource utilization level is below it.
- **mtasSipOcDefIncrStep:** This attribute defines the default step value by which the 'oc' will be increased.
- **mtasSipOcDefDecrStep:** This attribute defines the default step value by which the 'oc' will be decreased.
- **mtasSipOcValidity:** This attribute defines the OC validity in ms which set as oc-validity in Via header.
- **mtasSipOcRegulationInterval:** This attribute defines the OC regulation period in ms.
- **mtasSipOcResource:** This attribute defines the resources and optionally their limits which are considered in overload situation. Format shall be <resource type>[&limit]. Limit is integer from 0-100. Examples: Cpu, Cpu&80.
The available resource types are described in the MTAS SIP Upstream Overload Control Management Guide.
PLEASE BE CAUTIOUS BY OVERRIDING THE DEFAULT VALUES! - it significantly impacts the behavior of the SIP Overload Control feature.

2.59.4 Performance management

The following 4 performance counters are provided by MTAS.

MtasSipOcOvIPeriods

Description: The number of the cases when the node entered the overload operation mode within the measured period.

Condition: The counter is incremented by 1 when the node entered the overload mode (OC value changes from 0 to some other value).

MtasSipOcOviDuration

Description: the number of seconds which the node spent in overload condition within the measured period.

Condition: The counter is incremented by 1 in each second during the overload period when the OC value is greater than 0.

MtasSipOcOviPeak

Description: the maximum SIP Overload Control value sent towards upstream nodes within the measured period.

Condition: The gauge is updated when the oc value is greater than previous.



MtasSipOcOvlAvg

Description: the average of the reported oc values within the measured period. Only those oc values are counted which belong to an overload condition interval of the node.

Condition: The gauge is updated when OC value is greater than 0.

3 Self administration

3.1 Self Administration via Supplementary Service Codes

3.1.1 System defined SSC commands

3.1.1.1 Description

This feature enables users to gain access to and control supplementary services by using system defined Supplementary Service Code Commands. It means, there is a predefined set of service code commands of which configuration is supported by built-in command syntax attributes – see the list below.

The service is independent of user access type and includes all necessary procedures to activate, deactivate, disable, interrogate and invoke a supplementary service.

MTAS provides a solution for a user to gain access to, and control of, the following supplementary services:

- Communication Forwarding Unconditional (CFU)
- Communication Forwarding Unconditional to Voice Mail (CFUVM)
- Communication Forwarding on Busy (CFB)
- Communication Forwarding on Busy to Voice Mail (CFBVM)
- Communication Forwarding on No Reply (CFNR)
- Communication Forwarding on No Reply to Voice Mail (CFNRVM)
- Communication Forwarding on Not Logged in (CFNL)
- Communication Forwarding on Not Logged in to Voice Mail (CFNLVM)
- Originating Communication Barring (OCB)*
- Anonymous Communication Rejection (ACR)
- Modification of PIN
- OIP Activation (activate/deactivate/interrogate)



- OIR Activation (activate/deactivate/interrogate)
- OIR Default behavior (presentation restricted / presentation not restricted)
- TIP Activation (activate/deactivate/interrogate)
- TIR Activation (activate/deactivate/interrogate)
- TIR Default behavior (presentation restricted/presentation not restricted)
- CNIP Activation (activate/deactivate)
- Communication Waiting (CW)
- Malicious Communication Identification (MCID)
- Dynamic Black List (DBL)
- Malicious Communication Rejection (MCR)
- Abbreviated Dialing
- Communication Completion (CC)
- Explicit Communication Transfer (ECT)
- Multi Subscriber Number (MSN)
- Cell Announcement For Mobile Location (CellAnn)

MTAS also provides the end-user with the ability to enter a single interrogation request and be informed of all the active Communication Forwarding services (CFU, CFUVM, CFB, CFBVM, CFNR, and CFNRVM).

* Includes the control of OCB service with the barring programs only.

The CFU, CDIV and CFNR are also referred as CDIV services further in the document, while the OCB and ACR are referred as CB services.

MTAS supports each of the three major code schemes specified by ITU-T Recommendations, E.131 [35]:

- AT&T code scheme (USA)
- CEPT (ETSI) code scheme (Europe)
- NTT code scheme (Japan)

The main differences between those schemes are in the methods used to encode various command parameters, and the order in which they must be presented in service code command.

The general syntax for service code command looks like:

PX SC (FC) ((SR) SI) SX



Where:

- **PX - Service Prefix (ETSI) or Access Prefix (AT&T and NTT)**
A mandatory prefix. Generally consists of the star (*) and/or square (#) symbols, but it can be configured to use other symbols as well. In ETSI coding schemes it is used to define both access to the supplementary service and the operation (function) requested of the service. For AT&T and NTT coding schemes it is used to indicate access to the supplementary services only.
- **SC - Service Code**
A two or three digit mandatory code that is used to identify the service being accessed. In AT&T coding scheme it also indicates what operation (function) is requested for the service.
- **FC - Function Code**
An optional parameter used in NTT coding scheme to indicate the operation requested of the service.
- **SR – Block Separator**
One or more optional separator(s) for ETSI and NTT only. If required, it is used to separate the service code/function code from any supplementary information, or to separate two items of supplementary information.
- **SI - Supplementary Information**
An optional element that is normally composed of digits, but may also include alphabetic characters or symbols at the discretion of the operator/service provider. The supplementary information shall not include symbols that are usually used for service/access prefix, separator or command suffix (for example, * or # if ETSI coding scheme is used).
- **SX – Service Suffix**
This optional suffix is used to indicate the end of the complete service code command string.

3.1.1.1.1 Update Service Data

The MTAS may need to update the service data in the HSS as a result of service execution. Activation/Deactivation of CF using service codes functionality is an example of such a service that needs to alter the service data for a user.

3.1.1.1.2 Network Announcement

When the procedure identified by the service code command is completed successfully, if the node is configured to play the announcement from the MRFP and the received SDP indicates audio, the MTAS initiates a positive announcement, - which might be both fixed or segmented type in cases of call forward activation with new destination (ND) - from the MRFP towards the user. Following that, MTAS responds to the INVITE request with 200 OK.



To the contrary, if the procedure specified by the service code command can not be completed successfully, the MTAS initiates a negative announcement towards the user. In this case MTAS ends the procedure by sending a 4xx/5xx final response to the INVITE request.

3.1.1.2 Example Call Flow

3.1.1.2.1 Example Supplementary Service Codes

Below some examples are listed which are prepared according to the ETSI coding:

*61*PIN#	Activation of Communication Forwarding on No Reply (CFNR) with PIN
#21#	Deactivation of Communication Forwarding Unconditional (CFU)
*34*PIN*BP#	Activation of a Barring Program (BP) with PIN
#31*DN	Disabling of Originating Identification Restriction (OIR) on a per call basis (DN: Destination Number)

3.1.1.2.2 Successful Service Data Update using SSC

Following diagram shows an example call flow of successful service data update using SSC.

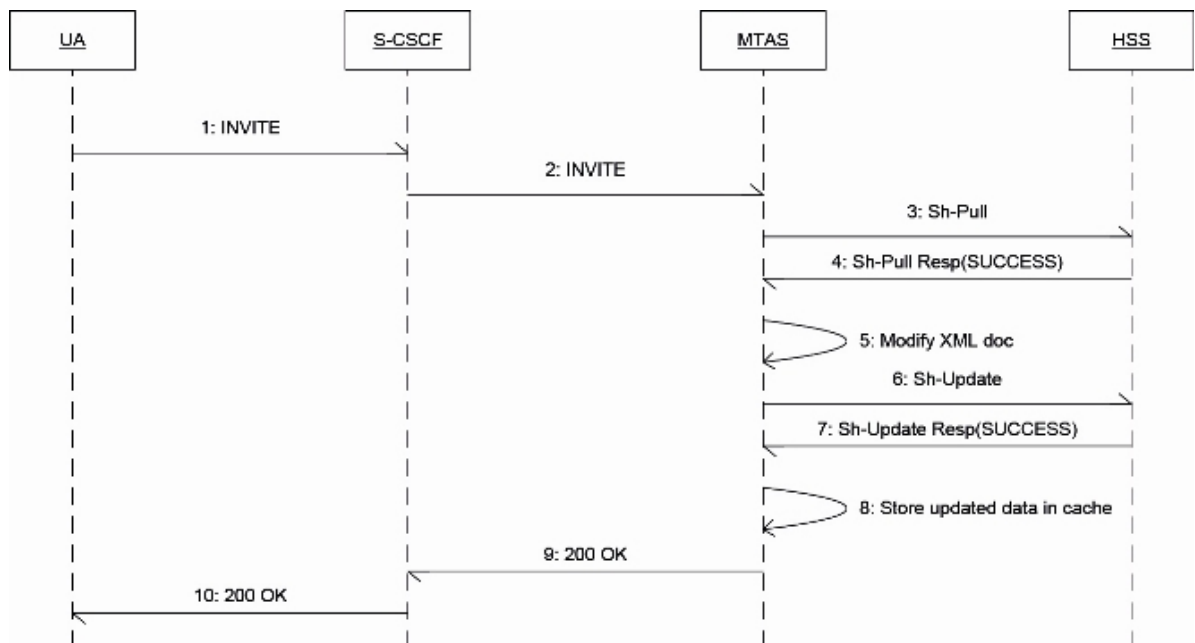


Figure 94 - Successful Service Data Update using SSC

1. The UA send an INVITE for a service requiring update of service data (supplementary service codes)
2. The S-CSCF evaluates the initial filter criteria and sends the INVITE to the MTAS



3. The MTAS, as a result of a service invocation (supplementary service codes), must update the service data. It sends a Sh-Pull request (User Identity, Requested Data, Service Indication) to fetch the service data XML. Note that the PUI is the default (first) in the IRS and might not be the same as in the INVITE.
4. The HSS sends a successful Sh-Pull Response with the requested data.
5. The MTAS, as a result of a service invocation (supplementary service codes), must update the service data. The MTAS merges the modified service data in the XML document received in step 4.
6. The MTAS sends a Sh-Update (PUI, Requested Data, Service Indication, Seq Number, UserData in XML doc)
7. The HSS sends back a successful Sh-Update Response.
8. The MTAS stores the updated service data in its cache.
9. The MTAS sends a successful SIP INVITE response to the S-CSCF. Note that a service may in addition play announcements or continue with the normal call establishment procedure. This is covered in the corresponding service functional specification.
10. The S-CSCF sends the 200 OK responses to the UA.

3.1.1.3 Configuration

The node-level configuration parameters related to Supplementary Service Codes are for example:

- Service code command syntax and parameters
- Administrative state
- Voice announcement codes, IDs (in case of generic, like segmented, announcements)

3.1.1.4 Performance Management

MTAS provides performance counters for each subscriber controlled supplementary service. The following types of performance counters are available:

- Number of successful activations of the service
- Number of unsuccessful activations of the service due to MTAS internal fault
- Number of unsuccessful activations of the service due to MTAS external fault
- Number of incorrect activations of the service



- Number of successful interrogations of the service

3.1.2 Generic SSC commands

3.1.2.1 Description

The generic Supplementary Service Codes are very similar to the “system defined” Supplementary Service Codes described in the chapter 3.1. The main point of this feature is that the operator can configure new Supplementary Service Codes dynamically and can assign operations on the user’s service data to them. Each generic Supplementary Service Codes (GenSSC) has an associated operator configuration (set of CM parameters stored in a CM MOC instance).

The Generic SSC includes an engine which performs the SSC command based on the operator configuration. The engine takes the operator configuration and the SSC command string and generates an XCAP request from them – please refer to chapter 3.2 for more details about the XCAP/Ut interface. The XCAP request is sent to the XDMS subsystem which performs the requested operations on the user’s service data, updates the HSS and sends the XCAP response back to the GenSSC engine. The engine evaluates the XCAP response, generates charging notifications, increases the corresponding PM counters and plays an appropriate announcement to the user.

The structure of the generic service code command is described below.

(PX) SC (SR) (SI) (SX)

Where:

- **PX – Service Prefix**
Prefix character(s) of the command string. Usually, it consists of the star (*) and/or square (#) symbols but it may contain zero digits as well. It may define the operation of the requested service in few limited cases: it may indicate the activation, deactivation, interrogation or the state altering (active $\leftarrow \rightarrow$ inactive) of the referred service.
- **SC - Service Code**
A string of two up to eight digits mandatory code. The service code is the only mandatory part of the command string.
- **SR – Block Separator**
SR is usually a star (*) and it is used to separate the command string pieces from each other.
- **SI - Supplementary Information**
An optional element that is normally composed of digits, but may also include alphabetic characters or symbols at the discretion of the operator/service provider.
- **SX – Service Suffix**
The SX is commonly a square (#) symbol and is used to indicate the end of the complete service code command string.



3.1.2.2 Example Call Flow

3.1.2.2.1 Example Generic Supplementary Service Codes

Below some examples are listed:

21	Activation of Communication Forwarding Unconditional (CFU)
#21PIN#	Deactivation of Communication Forwarding Unconditional (CFU)
*21*DN#	Update the CFU forward-to number with <DN>
231<timeout>#	Update the Communication Forwarding No Reply (CFNR) timer value with <timeout>
*240*2#	Interrogation of Hotline service
*240*3#	Altering (active ← → inactive) of Hotline service

3.1.2.2.2 Successful Service Data Update using SSC

The Generic SSC has the same call flow as the “system defined” one – please refer to chapter 3.1.1.2.2.

3.1.2.3 Configuration

The relevant node-level configuration parameters related to Generic Supplementary Service Codes are listed below.

- Administrative state of the SSC service is reused
- Generic SSC command syntax and parameters
- Generic SSC mapping to Ut command
- Voice announcement codes

3.1.2.4 Performance Management

MTAS provides performance counters for each subscriber controlled generic supplementary service. The following types of performance counters are available:

- Number of successful invocations of the service
- Number of unsuccessful invocations of the service due to MTAS internal fault
- Number of unsuccessful invocations of the service due to MTAS external fault
- Number of unsuccessful invocations of the service due to not provisioned user service data.



- Number of incorrect invocations of the service – user fault.
- Number of successful interrogations of the service
- Number of unsuccessful interrogations of the service due to MTAS internal fault
- Number of unsuccessful interrogations of the service due to MTAS external fault
- Number of unsuccessful interrogations of the service due to not provisioned user service data.
- Number of incorrect interrogations of the service – user fault.

Each performance counter is keyed with name of the corresponding MOC instance.

3.2 Self Administration via Ut interface

3.2.1 Description

The Self Administration via Ut-interface feature enables an MTAS subscriber to read, modify and delete his/her MTAS service specific provisioning data, for example settings belonging to his/her Communication Forwarding service. Ut interface is standardized by OMA [20] and XDM format for MTAS provisioning is specified by TISPAN [19].

The Ut-client generates XCAP (XML Configuration Access Protocol) requests which are sent to the MTAS XDMS (XML Document Management Server).

The XCAP Request messages are authenticated before they are received by the MTAS XDMS. This is done by the Aggregation Proxy (AP).

Possible types of Ut Clients include:

Ut-client implemented in operator's existing WEB portal.

UE (User Equipment) with Ut-client

MMTel WUIGM (Web User Interface for Group and Data Management) which implements an Ut-client for a WEB portal

3.2.2 Configuration

The Self Administration via Ut-interface feature has one operator controlled, node-level parameter which is used to activate/deactivate the Ut-interface.



3.2.3 Performance Management

MTAS provides Ut/XCAP interface counters to evaluate the usage and quality of service, like:

- Number of Ut interface XCAP GET requests that resulted in a successful response
- Number Ut interface XCAP GET requests that failed due to reasons external to the XDMS
- Number of Ut interface XCAP PUT requests that resulted in a successful response



4 Acronyms and Abbreviations

3GPP	3rd Generation Partnership Project
3PCC	Third Party Call Control
ACR	Anonymous Communication Rejection
API	Application Programming Interface
AS	Application Server
ASCII	American Standard Code for Information Interchange
ATCF	Access Transfer Control Function
ATGW	Access Transfer Gateway
ATU-STI	Access Transfer Update Session Transfer Identification
AUC	Additional User Category
bfd	Bidirectional Forwarding Detection
BGCF	Border Gateway Control Function
BP	Barring Program
BW	Bandwidth
C-MSISDN	Correlating MSISDN
CA	Charge Area
CAC	Call Admission Control
CAI3G	Customer Administration Interface 3rd Generation
CAMEL	Customized Application for Mobile Enhanced Logic
CAP	CAMEL Application Protocol
CARI	Carrier Information
CAS	Conference Administration Server
CAT	Customized Alerting Tones
CB	Communication Barring
CC	Country Code
CC	Carrier Code (in the context of Japanese Charging)



CCBS	Communication Completion on Busy Subscriber
CCMP	Centralized Conferencing Manipulation Protocol
CCNL	Communication Completion on Not Logged-in Subscriber
CCNR	Communication Completion on No Reply
CD	Communication Deflection
CDF	Charging Data Function
CdPN	Called Party Number
CDS	Communication Details Server
CDTF	Communication Details Transfer Function
CDIV	Communication Diversion
CDS	Communication Details Server
CFB	Communication Forwarding on Busy
CellAnn	Cell Announcement For Mobile Location
CFNL	Communication Forwarding Not Logged in
CFNR	Communication Forwarding No Reply
CFU	Communication Forwarding Unconditional
CGI	Cell Global Identification
CgPN	Calling Party Number
CM	Configuration Management
CNIP	Calling Name Identity Presentation
CNS	Calling Name Server (NameDb)
CS	Circuit Switched
CSC	Carrier Selection Code
CSCF	Call Session Control Function
CSI	CAMEL Subscription Information
CSRN	CS domain Routing Number
CUG	Closed User Group
DBL	Dynamic Black List



DN	Directory Number of a called party
DND	Do Not Disturb
DNDCB	Do Not Disturb Communication Barring
DNDCF	Do Not Disturb Communication Forwarding
DNM	Dialed Number Mapping
DTMF	Dual Tone Multi-Frequency
ECIM	Ericsson Common Information Model
ECMP	Equal-Cost Multi-Path routing
EDGE	Enhanced Data rates for GSM Evolution
EMA	Ericsson Multi Activation
ENUM	E.164 Number Mapping
ets	Emergency Telecommunication Service
ETSI	European Telecommunication Standards Institute
E-UTRAN	Evolved UTRAN
FCD	Flexible Communication Distribution
FCDDP	Flexible Communication Distribution Divert Primary
FCH	Flexible Charging
FMC	Fixed Mobile Convergence
FSFS	Flexible Service Format Selection
GERAN	GSM EDGE Radio Access Network
GETS	Government Emergency Telecommunication Service
GETS-AS	GETS – Application server
GETS-FC	GETS - Feature Code
GETS-AN	GETS – Access Network
GETS-NT	GETS – Number Translation
GIR	Government Industry Requirements for IP IMS Core Network
GM	Gateway Model
GMSC	Gateway MSC



GPRS	General Packet Radio Service
GSM	Global System for Mobile Communications
gsmSCF	GSM Service Control Function
gsmSSF	GSM Service Switching Function
HLR	Home Location Register
HPLMN	Home PLMN
HSS	Home Subscriber Server
I-CSCF	Interrogating CSCF
ICB	Incoming Communication Barring
ICBS	Interconnection Charge Billing System
ICID	IMS Charging Identifier
ICS	IMS Centralized Services
IEC	Immediate Event Charging
IETF	Internet Engineering Task Force
iFC	initial Filter Criteria
IMPI	IP Multimedia Private Identity
IMPU	IP Multimedia Public Identity
IMRN	IP Multimedia Routing Number
IMS	IP Multimedia Subsystem
IMSI	International Mobile Subscriber Identity
IRS	Implicit Registration Set
ISC	IP multimedia Service Control
ISDN	Integrated Service Digital Network
ISUP	ISDN User Part
JC	Japanese Charging
LAI	Location Area Identification
LDAP	Lightweight Directory Access Protocol
LTE	Long Term Evolution



MDUCAC	Multi Device User Call Admission Control
MAP	Mobile Application Part
MCC	Mobile Country Code
MCID	Malicious Communication Identification
MCR	Malicious Communication Rejection
MGCF	Media Gateway Control Function
MME	Mobility Management Entity
MMTel	Multi-Media Telephony
MNC	Mobile Network Code
MO	Mobile Originating call
MRFC	Media Resource Function Controller
MRFP	Media Resource Function Processor
MSC	Mobile Switching Centre
MSISDN	Mobile Subscriber Integrated Service Digital Network
MSRN	Mobile Station Roaming Number
MSRP	Message Session Relay Protocol
MT	Mobile Terminating call
MTAS	Multimedia Telephony Application Server
ND	New Destination number
NTT	PSTN operator in Japan
NP	Number Portability
NPLI	Network Provided Location Information
NSN	Nationally Significant Number
O-SDS	Originating SDS
OCB	Outgoing Communication Barring
OCNIP	CNIP on originating MMTel AS
OCS	Online Charging System
OCT	Operator Controlled Transfer



OIP	Originating Identity Presentation	
OIR	Originating Identity Restriction	
OMA	Open Mobile Alliance	
OSN	Operator Service Number	
OSPF	Open Shortest Path First	
OTP	Operating Telephony Provider	
P-CSCF	Proxy CSCF	
PAI	P-Asserted-Identity	
PANI	P-Access-Network-Information	
PCRF	Policy and Charging Rules Function	Public
Land Mobile Network	PLMN	
PS	Packet Switched	
PSTN	Public Switched Telephony Network	
QoS	Quality of Service	
RBT	Ring Back Tone	
RFC	Request for Comment	
RPH	Resource Priority Header	
S-CFU	Selective CFU	
S-CSCF	Serving-CSCF	
SCP	Service Control Point	
SCUR	Session based Charging with Unit Reservations	
SDP	Session Description Protocol	
SDS	Service Domain Selection	
SGSN	Serving GPRS Support Node	
SIP	Session Initiation Protocol	
SND	Short Number Dialing	
SPT	Service Point Trigger	
SRF	Specialized Resource Function	
SRVCC	Single Radio Voice Call Continuity	



SCC	Service Centralization and Continuity
SCN	Subscriber Credit Notification
SDS	Service Domain Selection
SR-VCC	Single Radio Voice Call Continuity
SSC	Supplementary Service Code
SSF	Service Switching Function
SSP	Service Switching Point
STN-SR	Session Transfer Number for SRVCC
STOD	Session Transfer to Own Device
T-ADS	Terminating Access Domain Selection
T-SDS	Terminating SDS
TCP	Transport Control Protocol
TDS	Telephone Directory Service
TIP	Terminating Identity Presentation
TIR	Terminating Identity Restriction
TISPAN	Telecoms & Internet converged Services & Protocols for Advanced Network
UA	User Agent
UE	User Equipment
URI	Uniform Resource Identifier
UTRAN	UMTS Terrestrial Radio Access Network
UMTS	Universal Mobile Telecommunications System
VLR	Visited Location Register
VMSC	Visited MSC
VoLTE	Voice over LTE
VoPS	Voice over PSVPLMN Visited PLMN
VTP	Virtual Telephony Provider
wps	Wireless Priority Service
XCAP	XML Configuration Access Protocol



XDMS	XML Data Management Server
XML	eXtensible Markup Language



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** See the Customer or Support library for the Application System in question