

All Features in MTAS 16

FEATURE DESCRIPTION



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1 Key Benefits

1.1 Shorter call set up due to separate T-ADS timers for Wi-Fi and VoLTE



2 Content Summary

Available Features in current and the previous three Releases.

1/O = Version 1 / Optional Feature

2/B = Version 2 / Basic Feature

Number	Name	15A	15B (FD)	16A	16B (FD)
FAJ 131 0501	Abbreviated Dialing	1/B	1/B	1/B	1/B
FAJ 131 0185	Ad-hoc Conference	4/O	5/O	6/O	6/O
FAJ 131 0169	Add/Drop Media	1/O	1/O	1/O	1/O
FAJ 131 0548	Address Policing	1/O	1/O	1/O	1/O
FAJ 131 0496	Advice of Charge	1/O	1/O	1/O	1/O
FAJ 131 0182	Anonymous Communication Rejection	2/O	2/O	2/O	2/O
FAJ 131 0544	AS Controlled Forking	1/O	1/O	1/O	1/O
FAJ 131 0161	Basic Barring	3/B	3/B	3/B	3/B
FAJ 131 0149	Basic Voice Communication	1/B	1/B	1/B	2/B
FAJ 131 0685	Business Mobility Base	1/O	1/O	1/O	1/O
FAJ 131 0684	Business Mobility Dynamic Service Domain Selection	1/O	1/O	1/O	1/O
FAJ 131 0495	Call Admission Control	1/O	1/O	1/O	1/O
FAJ 131 0633	Call Return	2/O	2/O	2/O	2/O
FAJ 131 0563	Calling Party Category	1/O	1/O	1/O	1/O
FAJ 131 0688	CAPv2 SSF	1/O	1/O	1/O	1/O
FAJ 131 0426	Carrier Pre-Select	5/O	5/O	5/O	5/O
FAJ 131	Carrier Select	3/O	3/O	3/O	3/O



0425					
FAJ 131 0668	Closed User Group	1/O	1/O	1/O	1/O
FAJ 131 0497	Communication Completion Services	4/O	4/O	4/O	5/O
FAJ 131 0175	Communication Deflection	2/O	2/O	2/O	2/O
FAJ 131 0430	Communication Diversion - Operator Blacklist	2/B	2/B	2/B	2/B
FAJ 131 0429	Communication Diversion - to Voice Mail	4/B	4/B	4/B	4/B
FAJ 131 0160	Communication Diversion Notification - Caller	2/B	2/B	2/B	2/B
FAJ 131 0159	Communication Diversion Notification - Reminder	2/B	2/B	2/B	2/B
FAJ 131 0158	Communication Diversion Notification - Served-user	2/B	2/B	2/B	2/B
FAJ 131 0586	Communication Diversion Rule Based	1/O	1/O	1/O	1/O
FAJ 131 0156	Communication Forwarding - Busy	2/B	2/B	2/B	2/B
FAJ 131 0157	Communication Forwarding - No Reply	4/B	4/B	4/B	4/B
FAJ 131 0155	Communication Forwarding - Unconditional	2/B	2/B	2/B	2/B
FAJ 131 0427	Communication Name Identity Presentation	3/O	3/O	3/O	3/O
FAJ 131 0423	Communication Waiting	2/B	2/B	2/B	2/B
FAJ 131 0576	Customized Alerting Tone	1/O	1/O	1/O	1/O



FAJ 131 0422	Dial Tone Management	3/B	3/B	3/B	3/B
FAJ 131 0494	Dynamic Black List	3/O	3/O	3/O	3/O
FAJ 131 0562	Explicit Communication Transfer	2/O	2/O	2/O	2/O
FAJ 131 0168	File Sharing	1/O	1/O	1/O	1/O
FAJ 131 0636	Flexible AVP	1/O	1/O	1/O	1/O
FAJ 131 0492	Flexible Communication Distribution	5/O	6/O	7/O	7/O
FAJ 131 0571	Flexible Identity Presentation	1/B	1/B	1/B	1/B
FAJ 131 0579	Flexible Service Format Selection	2/O	2/O	2/O	2/O
FAJ 131 0491	Gateway Model	2/O	2/O	2/O	2/O
FAJ 131 0592	GSM Compatible SSF	1/O	1/O	1/O	1/O
FAJ 131 0150	Hold Communication	1/B	1/B	1/B	1/B
FAJ 131 0634	Hotline	1/O	1/O	1/O	1/O
FAJ 131 0803	HSS Bypass		1/O	1/O	1/O
FAJ 131 0632	In Conference Control	1/O	1/O	1/O	1/O
FAJ 131 0573	Incoming Communication Barring Rule Based	3/O	3/O	3/O	3/O
FAJ 131 0547	International and National Toll Restriction	1/B	1/B	1/B	1/B
FAJ 131 0697	Japanese Charging	2/O	2/O	2/O	3/O
FAJ 131 0493	Legal Intercept	4/O	4/O	4/O	4/O
FAJ 131 0577	Location Based Number Translation	4/O	5/O	5/O	5/O



FAJ 131 0500	Malicious Communication Identification	2/B	2/B	2/B	2/B
FAJ 131 0525	Malicious Communication Rejection	1/B	1/B	1/B	1/B
FAJ 131 0543	Mr-interface	2/O	2/O	2/O	2/O
FAJ 131 0540	Multiple Languages Support	2/O	2/O	2/O	2/O
FAJ 131 0661	Multiple Subscriber Number	2/O	2/O	2/O	2/O
FAJ 131 0165	Network Announcement	3/O	3/O	3/O	3/O
FAJ 131 0698	Network Provided Location Information Retrieval in MMTel AS	1/B	1/B	1/B	1/B
FAJ 131 0699	Network Provided Location Information retrieval in SCC-AS	1/B	1/B	1/B	1/B
FAJ 131 0696	Network Provided Ringback Tone	1/O	1/O	1/O	1/O
FAJ 131 0187	Number Normalisation	1/B	1/B	1/B	1/B
FAJ 131 0575	Number Portability	2/O	2/O	2/O	2/O
FAJ 131 0631	Number Translation	1/B	1/B	1/B	1/B
FAJ 131 0164	Offline Charging	6/O	7/O	8/O	9/O
FAJ 131 0499	Online Charging	4/O	5/O	6/O	7/O
FAJ 131 0184	Operator Anonymous Communication Rejection	1/O	1/O	1/O	1/O
FAJ 131	Operator Black	1/O	1/O	1/O	1/O



0183	Lists				
FAJ 131 0527	Operator White Lists	1/O	1/O	1/O	1/O
FAJ 131 0151	Originating Identity Presentation	4/B	4/B	4/B	4/B
FAJ 131 0152	Originating Identity Restriction	2/B	2/B	2/B	2/B
FAJ 131 0572	Outgoing Communication Barring Rule Based	2/O	2/O	2/O	2/O
FAJ 131 0528	Parlay-X	3/O	3/O	3/O	3/O
FAJ 131 0428	Precondition	1/O	2/O	2/O	3/O
FAJ 131 0490	Priority Call	2/O	2/O	2/O	2/O
FAJ 131 0574	Scheduled Conference	4/O	4/O	4/O	4/O
FAJ 131 0163	Self Administration via Service Codes	7/B	8/B	8/B	8/B
FAJ 131 0186	Self Administration via Ut-interface	6/O	6/O	6/O	6/O
FAJ 131 0567	Service Centralization and Continuity Application Server	3/B	3/B	3/B	3/B
FAJ 131 0569	Service Domain Selection	2/O	2/O	2/O	2/O
FAJ 131 0564	Service Profile	2/O	2/O	2/O	2/O
FAJ 131 0580	Session Transfer to Own Device	3/O	4/O	4/O	4/O
FAJ 131 0502	Short Number Dialing	1/O	1/O	1/O	1/O
FAJ 131 0568	Single Radio VCC	3/O	4/O	4/O	4/O
FAJ 131	SIP Trunking	1/B	2/B	2/B	2/B



0797	AS Base				
FAJ 131 0561	Subscriber Credit Notification	1/O	1/O	1/O	1/O
FAJ 131 0526	Support for IPv6	3/B	3/B	3/B	3/B
FAJ 131 0570	Terminating Access Domain Selection	4/O	4/O	4/O	4/O
FAJ 131 0153	Terminating Identity Presentation	1/B	1/B	1/B	1/B
FAJ 131 0154	Terminating Identity Restriction	2/B	2/B	2/B	2/B
FAJ 131 0167	Text Chat	1/O	1/O	1/O	1/O
FAJ 131 0162	Three Party Call	2/B	2/B	2/B	2/B
FAJ 131 0672	Time Zone Handling	1/B	1/B	1/B	1/B
FAJ 131 0166	Video Communication	1/O	1/O	1/O	1/O
FAJ 131 0549	Video Fallback to Audio	1/O	1/O	1/O	1/O
FAJ 131 0635	Wholesale	2/O	2/O	2/O	2/O
FAJ 131 0791	WiFi Calling	1/O	1/O	1/O	2/O
FAJ 901 0060	WiFi Calling MMTel			1/O	2/O



3 General - BASIC FEATURES



3.1 Abbreviated Dialing

Feature Identity:	FAJ 131 0501/1 R1A, Rev. A
Feature Type:	Basic in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

3.1.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

3.1.2 Summary

The Abbreviated Dialing service enables a subscriber to call an assigned, stored number by dialing a short digit sequence. Abbreviated numbers between 0-9 and 00-99 are supported.

Price Object INF 901 5045/1, MMTel Base.

3.1.3 Benefits

End-user

Abbreviated Dialing simplifies the dialing for long or frequently-used telephone numbers for an end-user.

Operator

Operators are able to charge a premium for the service and so increase revenues.

3.1.4 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, with or without a adhoc identity presentation SSC or carrier select code added, numbers for every user exist in the user data (stored number is defined [<SSC>][<CSC>]<ND>, SSC and CSC are optional).. Provision is done on the CAI3G interfaces. Update of the user data is done with CAI3G and on the Ut interface.

Sent type of URI (sip: or tel:), the output from MTAS, is the same type of URI (sip: or tel:) stored.

There are two levels of data management that are performed by the operator for a MTAS service:

- Node level configuration
- Provision Subscriber Service data

Node level configuration is performed by the operator via the Muta using LDAP protocol and it allows the operator to customize the service function by using managed object attributes specified for the function. For example, this includes configuration of service code command syntax and parameters, administrative state of the function etc.

Provisioning Subscriber Service data is performed by the operator via EMA using CAI3G protocol and it allows the operator to manage service data for a subscriber. This, for example, includes definition of whether the service is granted to the subscriber, withdrawn from a subscriber etc.



Service data is managed via XDMS that provides the Ut interface (XCAP over HTTP) to the user and CAI3G to the operator. XDMS uses Sh (Diameter) to update the HSS. The user accesses the XDMS directly and the operator accesses it via EMA. The user may access the list, of abbreviated and stored numbers, and change services settings via the Ut interface.



3.2 Basic Barring

Feature Identity: FAJ 131 0161/1 R3A, Rev. A

Feature Type: Basic in 12A to 16B (FD)

Technology:

3.2.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

MGC:

FAJ 131 0306 ISUP and SIP Interworking: ACR and CB Supplementary Service

Terminal impacts and dependencies

Not Specified

3.2.2 Summary

This feature provides the end-user and the operator with the possibility to select from operator-defined Barring Programs to bar the end-users outgoing voice communication. The outgoing communication restriction in Barring Programs is always based on destination.

This is an Ericsson specific service.

Price Object INF 901 5045/1, MMTel Base.



3.2.3 Benefits

End-user

The end-user will be able to control his/her own outgoing traffic.

Operator

The operator will be able to offer Outgoing Barring services to its end-users. The operator can also control the outgoing traffic from the end-user e.g. on demand from the end-user or for limited subscriptions.

3.2.4 Description

The operator can define so-called Barring Programs. 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 barring programs defined by the operator can be controlled either by the end-user or the operator.

The Operator Controlled Barring is sometimes referred to as OCB.

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

For example:

The Request URI 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 relatively long list of relatively complex node-level Barring Program from which the end-user or operator selects 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 single scheme the end-user and the operator can have one program at the same time.

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 or operator will select one or more programs at the same time in order to create an individual outgoing barring plan by combining the offered programs.

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 audio announcements is optional and configurable by the operator.

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

MTAS provides the operator a set of node-level configuration options which are listed below. The node-level configuration is performed via LDAP-interface.

- CommuniBarring enable/disable
There is one common administrative state of the Communication Barring function in MTAS. The operator can enable/disable all barring sub-features on node-level (i.e. Barring Programs, OCB, ICB, ACR, etc).
 - Communication Barring enable/disable
There is one common administrative state of the Communication Barring function in MTAS. The operator can enable/disable all barring sub-features on node-level (i.e. Barring Programs, OCB, ICB, ACR, etc).



- Announcement enable/disable
The operator can enable/disable the audio announcement when a communication is barred.
- "Audio only" Announcement
The operator can specify which audio announcement is to be played to the originating party when a communication is barred.
- Video Announcement enable/disable
The operator can enable/disable video announcement when a communication is barred.
- "Video only" Announcement
The operator can specify which video announcement, without audio, is to be played to the originating party when a communication is barred.
- Audio/Video Announcement - Audio part
The operator can specify which audio, associated with a video announcement, to be played to the originating party when a communication is barred.
- Audio/Video Announcement - Video part
The operator can specify the video part of the audio/video announcement.

The user-level configuration is performed by the end-user via the Ut-interface and accessed by the operator via EMA (Ericsson Multi Activation) using CAI3G.

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

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

- Number of barred outgoing communications
 - Number of times an announcement was not successfully initiated

3.2.4.1

Standards

3GPP TS 24.611, IETF RFC 2396



3.2.5 Enhancement

New in MTAS 12A is the possibility to include carrier select prefix in barring rules.



3.3 Basic Voice Communication

Feature Identity: FAJ 131 0149/1 R2A, Rev. A

Feature Type: Basic in 16B (FD)

Technology: IMS

3.3.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

3.3.2 Summary

The basic voice communication feature considers set-up and clear down of 2-party voice sessions.

Price Object INF 901 5045/1, MMTel Base.

3.3.3 Benefits

Traditional voice communication service can be offered to subscribers.

3.3.4 Description

Variants of the basic 2-party session involving support for 100rel (SIP Provisional Message Reliability), early media and SIP forking are covered by the feature.

The operator is provided with a number of node-level configuration parameters. The configuration is performed using LDAP. Some configurable options as examples are:

- “No reply” timer for communication sessions
- Max. number of parallel communication sessions that a user can have



A number of performance counters are also provided to evaluate the usage and quality of service, like:

- Number of initiated sessions
- Number of audio type media streams
- Number of sessions that was not initiated due to internal exceptional events
- Number of sessions that was not initiated due to external exceptional events

3.3.4.1 Standards

3GPP TS 22.173 Stage 1, 3GPP TS 24.173 Stage 3, 3GPP TS 26.114, IETF RFC 3261, IETF RFC 3264

3.3.5 Enhancement

Enhancement in 16B:

Support for Long Duration Call Supervision.



3.4 Communication Diversion - Operator Blacklist

Feature Identity:	FAJ 131 0430/1 R2A, Rev. B
Feature Type:	Basic in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

3.4.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

3.4.2 Summary

The operator will be able to put certain destination numbers on a blacklist to avoid communication diversion to these numbers.

This is an Ericsson specific service.

Price Object INF 901 5045/1, MMTel Base.

3.4.3 Benefits

Operator

Communication Diversion to certain destination numbers can be disallowed

3.4.4 Description

In general, the Communication Diversion service in MTAS is rule-based, i.e. the end-user can create rules with different conditions and the rules are evaluated to test if their respective condition(s) are true.

Please refer to section *Communication Diversion (CDIV) Service* included in Communication Forwarding - Unconditional for a general description of the rule based mechanism.



The operator can define a blacklist to which it is not possible to perform communication diversion. The blacklist has the same format as the Outgoing Black List. When a call is diverted, MTAS bars the call if the new destination matches the blacklist, responding 603 Decline. Each time a diversion rule is inserted or amended, MTAS only allows the change if the target does not match the blacklist.

The Black List - Communication Diversion is managed by the operator via LDAP-interface.

For additional details, like network announcement support, please refer to "OCB Destination" feature.

3.4.4.1 Standards

3GPP TS 24.604, OMA OMA-TS-XDM_Core-V1_0, IETF RFC 4244, IETF RFC 2327

3.4.5 Enhancement

The Communication Diversion (CDIV) supplementary services are aligned with TISPA R2 and have been updated for additional licenses, hop-by-hop Ack, inclusion of Communication Forwarding Not Reachable (CFNRc), interaction with Call Admission Control (CAC) and Online Charging, addition of From-Header and standards changes for History-Info.



3.5 Communication Diversion - to Voice Mail

Feature Identity:	FAJ 131 0429/1 R4A, Rev. A
Feature Type:	Basic in 15A to 16B (FD)
Technology:	GSM, WCDMA, LTE, Fixed Broadband, Wireless Broadband

3.5.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

3.5.2 Summary

End-user can forward the incoming communication to Voice Mail.

Price Object INF 901 5045/1, MMTel Base.

3.5.3 Benefits

End-user

The end user does not have to add the address of his/her Voice Mail system. The address to Voice Mail is configured in MTAS and automatically provided at diversion to Voice Mail.

Operator

Traditional PSTN/ISDN Call Forwarding to Voice Mail service can be offered to subscribers.

Voicemail servers with a separate address per user, and voicemail servers with a single address for all users, can be supported by a single MTAS.

The successful call ratio grows which results in increased revenue.



3.5.4 Description

In general, the Communication Diversion service in MTAS is rule-based, i.e. the end-user can create rules with different conditions and the rules are evaluated to test if their respective condition(s) are true.

Please refer to section *Communication Diversion (CDIV) Service* included in Communication Forwarding - Unconditional for a general description of the rule based mechanism.

Communication diversion can be performed to Voice Mail. In MTAS it is possible for the operator to define the default address to Voice Mail, and to supply a specific address for each user, or any subset of users.

The operator can activate/deactivate the Voice Mail service for the subscriber and specify whether each subscriber uses the default Voice Mail address or a specific Voice Mail address by providing the XML structure to the user's Subscriber Data.

The end-user configures a special target in his/her diversion rules via the Ut-interface and the operator accesses them through EMA (Ericsson Multi Activation) using CAI3G-interface.

The served user can also manage some type of his/her Communication Forwarding service via Supplementary Service Codes (SSC). For the list of supported Communication Forwarding types please refer to "Self Administration via Service Codes" feature.

The operator can configure the behavior of the Voice Mail feature. The following node-level options are available which are configured via LDAP:

- The default address of the Voice Mail server for diverted communication.
- The default address for Voice Mail retrieval.

3.5.4.1 Standards

3GPP TS 24.604, OMA OMA-TS-XDM_Core-V1_0, IETF RFC 4244, IETF RFC 2327

3.5.5 Enhancement

Following has been enhanced in 14B:

Allowing users to divert their calls to Voice Mail using Supplementary Service Codes. Voice mail number does not need to be given by the user.



The Following has been added in 15A:

Dropped calls can be diverted to Voice Mail.

Possibility to have a Voice Mail retrieval number separate from the Voice Mail diversion number.



3.6 Communication Diversion Notification - Caller

Feature Identity:	FAJ 131 0160/1 R2A, Rev. B
Feature Type:	Basic in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

3.6.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

3.6.2 Summary

At communication forwarding the caller can be notified that a diversion has been invoked.

Price Object INF 901 5045/1, MMTel Base.

3.6.3 Benefits

End-user

The caller will be informed, during session establishment, about the communication diversion. He/she can decide to continue/interrupt the call.

Operator

Announcement service, similar to the PSTN/ISDN Call Forwarding notification, can be offered to end-users.

3.6.4 Description

In general, the Communication Diversion service in MTAS is rule-based, i.e. the end-user can create rules with different conditions and the rules are evaluated to test if their respective condition(s) are true.



Please refer to section *Communication Diversion (CDIV) Service* included in Communication Forwarding - Unconditional for a general description of the rule based mechanism.

In case of Communication Forwarding, MTAS will send a SIP response 181 (Call Is Being Forwarded) towards the caller in order to inform about the communication diversion.

The response is sent on the existing SIP dialogue.

The announcement is sent in-band using the early media session. MTAS controls an external MRFP over Mp-interface, in accordance with the TISpan profile.

A notification should be sent or not is a user-level parameter and is configured by the end-user.

Additionally, MTAS can initiate an audio or video announcement towards the caller. This announcement option is a node-level parameter which is configured by the operator.

3.6.4.1 Standards

3GPP TS 24.604 V8.4.0, 3GPP TS 24.628 V8.2.0, OMA OMA-TS-XDM_Core-V1_0, RFC 4244, IETF RFC 3265, IETF RFC 4660, IETF RFC 4661, IETF RFC 3428.

3.6.5 Enhancement

The Communication Diversion (CDIV) supplementary services are aligned with TISpan R2 and have been updated for additional licenses, hop-by-hop Ack, inclusion of Communication Forwarding Not Reachable (CFNRc), interaction with Call Admission Control (CAC) and Online Charging, addition of From-Header and standards changes for History-Info.



3.7 Communication Diversion Notification - Reminder

Feature Identity:	FAJ 131 0159/1 R2A, Rev. B
Feature Type:	Basic in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

3.7.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

3.7.2 Summary

If the communication forwarding service is active for a specific user he/she can decide if the system should send him/her a notification text message whenever he/she initiates a communication. An example of this message could be: "You have active diverts".

Price Object INF 901 5045/1, MMTel Base.

3.7.3 Benefits

End-user

By receiving notification from the system, end-user will not forget to deactivate the service.

Operator

New kind of value added notification service, similar to traditional PSTN/ISDN Call Forwarding Active, can be offered to subscribers.



3.7.4 Description

In general, the Communication Diversion service in MTAS is rule-based, i.e. the end-user can create rules with different conditions and the rules are evaluated to test if their respective condition(s) are true.

Please refer to section *Communication Diversion (CDIV) Service* included in Communication Forwarding - Unconditional for a general description of the rule based mechanism.

In case of Communication Forwarding MTAS can notify the served user (forwarding user) with a text message that the communication forwarding service is active by sending a SIP MESSAGE to the served end-user.

The MESSAGE request will be sent out-of-dialog.

A notification should be sent or not is a user-level parameter and is configured by the end-user.

The operator can define the contents of the text message that is sent to remind a served user (i.e. B-party), when an outgoing call is made, that diversions are active. The default value is: "You have active divers".

3.7.4.1 Standards

3GPP TS 24.604 V8.4.0, 3GPP TS 24.628 V8.2.0, OMA OMA-TS-XDM_Core-V1_0, RFC 4244, IETF RFC 3265, IETF RFC 4660, IETF RFC 4661, IETF RFC 3428.

3.7.5 Enhancement

The Communication Diversion (CDIV) supplementary services are aligned with TISPAN R2 and have been updated for additional licenses, hop-by-hop Ack, inclusion of Communication Forwarding Not Reachable (CFNRc), interaction with Call Admission Control (CAC) and Online Charging, addition of From-Header and standards changes for History-Info.



3.8 Communication Diversion Notification - Served-user

Feature Identity:	FAJ 131 0158/1 R2A, Rev. B
Feature Type:	Basic in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

3.8.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

3.8.2 Summary

At communication forwarding the served user (i.e. B-party) can be notified that a diversion has been occurred. In many market the similar PSTN feature is known as "Ring Splash" service. In MMTel, according to the standard, a text message is sent to the user. However, depending on the client, a tone can at reception of the text message be generated to the end-user. An example of a text message could be: "Communication Diverted".

Price Object INF 901 5045/1, MMTel Base.

3.8.3 Benefits

End-user

Even if the call forwarding service is activated the end-user will be informed whenever an incoming communication request was received so he/she can take immediately actions (e.g. make a call).

Operator

New kind of value added notification service can be offered to subscribers.

It will differentiate the operator from the competitors.



3.8.4 Description

In general, the Communication Diversion service in MTAS is rule-based, i.e. the end-user can create rules with different conditions and the rules are evaluated to test if their respective condition(s) are true.

Please refer to section *Communication Diversion (CDIV) Service* included in Communication Forwarding - Unconditional for a general description of the rule based mechanism.

In case of Communication Forwarding MTAS can notify the served user (forwarding user) with a text message that a communication forwarding occurred by sending a SIP MESSAGE to the served end-user.

This MESSAGE request is sent out-of-dialog.

A notification should be sent or not is a user-level parameter and is configured by the end-user.

The operator can define the contents of the Text message that is sent to inform a served user (i.e. B-party) that a call addressed to him/her has been diverted elsewhere. The default value is: "Communication Diverted".

Notification by tone sending is possible to achieve from clients that, upon reception of the text message, reacts by generating a tone.

Enhancements

The Communication Diversion (CDIV) supplementary services are aligned with TISPAN R2 and have been updated for additional licenses, hop-by-hop Ack, inclusion of Communication Forwarding Not Reachable (CFNRc), interaction with Call Admission Control (CAC) and Online Charging, addition of From-Header and standards changes for History-Info.

3.8.4.1 Standards

3GPP TS 24.604 V8.4.0, 3GPP TS 24.628 V8.2.0, OMA OMA-TS-XDM_Core-V1_0, RFC 4244, IETF RFC 3265, IETF RFC 4660, IETF RFC 4661, IETF RFC 3428.

3.8.5 Enhancement

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3.9 Communication Forwarding - Busy

Feature Identity:	FAJ 131 0156/1 R2A, Rev. B
Feature Type:	Basic in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

3.9.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

MGC:

FAJ 131 0304 ISUP and SIP Interworking: Call Diversion Supplementary Service

Terminal impacts and dependencies

Not Specified

3.9.2 Summary

End-user can forward the incoming communication if his/her status is busy.

Price Object INF 901 5045/1, MMTel Base.

3.9.3 Benefits

End-user

When communicating with another party, the incoming calls are forwarded to another destination so the end-user will not miss any important calls.



Operator

Traditional PSTN/ISDN Busy Call Forwarding service can be offered to subscribers.

The successful call ratio grows which results increased revenue.

3.9.4 Description

In general, the Communication Diversion service in MTAS is rule-based, i.e. the end-user can create rules with different conditions and the rules are evaluated to test if their respective condition(s) are true.

Please refer to section *Communication Diversion (CDIV) Service* included in Communication Forwarding - Unconditional for a general description of the rule based mechanism.

In case of Communication Forwarding - Busy (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.

At communication forwarding the SIP response 181 (Call Is Being Forwarded) is sent to the caller.

In addition to the general configuration options described in "Communication Forwarding - Unconditional" feature the operator can set the CFB behavior on node-level via LDAP.

MTAS provides a configurable parameter which is used to determine the INVITE method responses for which the CF Busy condition shall be true. The default "486" is the TISPAN R1 compliant value.

The available options are:

- 486 - Busy Here
 - 600 - Busy Everywhere
 - 603 - Decline

Enhancements

The Communication Diversion (CDIV) supplementary services are aligned with TISPAN R2 and have been updated for additional licenses, hop-by-hop Ack, inclusion of Communication Forwarding Not Reachable (CFNRC), interaction with Call Admission Control (CAC) and Online Charging, addition of From-Header and standards changes for History-Info.



3.9.4.1 Standards

3GPP TS 24.604, OMA OMA-TS-XDM_Core-V1_0, IETF RFC 4244, IETF RFC 2327

3.9.5 Enhancement



3.10 Communication Forwarding - No Reply

Feature Identity: FAJ 131 0157/1 R4A, Rev. A

Feature Type: Basic in 12A to 16B (FD)

Technology:

3.10.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

MGC:

FAJ 131 0304 ISUP and SIP Interworking: Call Diversion Supplementary Service

Terminal impacts and dependencies

Not Specified

3.10.2 Summary

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

Price Object INF 901 5045/1, MMTel Base.

3.10.3 Benefits

End-user



End-user will not miss any important calls, not even when he/she is unable to answer the call.

Operator

Traditional PSTN/ISDN Call Forwarding No Reply service can be offered to subscribers. The successful call ratio grows which results increased revenue.

3.10.4 Description

In general, the Communication Diversion service in MTAS is rule-based, i.e. the end-user can create rules with different conditions and the rules are evaluated to test if their respective condition(s) are true.

Please refer to section *Communication Diversion (CDIV) Service* included in Communication Forwarding - Unconditional for a general description of the rule based mechanism.

In case of Communication Forwarding - No Reply (CFNR) MTAS starts a Timer at the reception of the SIP response 180 (Ringing). If the timer expires the communication is forwarded to the target address.

At communication forwarding the SIP response 181 (Call Is Being Forwarded) is sent to the caller.

The operator can activate/deactivate the CDIV No Answer Timer service for the subscriber by providing the XML structure to the user's Subscriber Data.

The end-user configures his/her CDIV No Answer Timer via the Ut-interface and the operator accesses it through EMA (Ericsson Multi Activation) using CAI3G-interface.

The served user can also manage his/her CDIV No Answer Timer service via Supplementary Service Codes (SSC). For the list of supported Communication Forwarding types please refer to "Self Administration via Service Codes" feature.

In addition to the general configuration options described in "Communication Forwarding - Unconditional" feature the operator can set the CFNR behavior on node-level via LDAP, i.e. the operator can set the default time interval within which the served user must respond before a session is forwarded, which applies to those end-users who have not set their CDIV No Answer Timer.

3.10.4.1 Standards

3GPP TS 24.604, OMA OMA-TS-XDM_Core-V1_0, IETF RFC 4244, IETF RFC 2327

**3.10.5****Enhancement**

The Communication Diversion (CDIV) supplementary services are aligned with TISpan R2 and have been updated for additional licenses, hop-by-hop Ack, inclusion of Communication Forwarding Not Reachable (CFNRc), interaction with Call Admission Control (CAC) and Online Charging, addition of From-Header and standards changes for History-Info.



3.11 Communication Forwarding - Unconditional

Feature Identity:	FAJ 131 0155/1 R2A, Rev. B
Feature Type:	Basic in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

3.11.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

MGC:

FAJ 131 0304 ISUP and SIP Interworking: Call Diversion Supplementary Service

Terminal impacts and dependencies

Not Specified

3.11.2 Summary

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

Price Object INF 901 5045/1, MMTel Base.

3.11.3 Benefits

End-user



This service can be used advantageously in different situations (e.g. when traveling, during meetings, etc.). By forwarding his/her incoming communication to another destination, the end-user will not miss any important calls.

Operator

Traditional PSTN/ISDN Unconditional Call Forwarding service can be offered to subscribers.

3.11.4 Description

Communication Diversion (CDIV) Service

The Communication Diversion (CDIV) supplementary service consists of a number of diversion features that will enable a served user to have the network redirect communication to another user.

This sub-chapter is intended to describe the general mechanism for all the CDIV related features. The following MTAS features are included:

Basic features

- Communication Forwarding - Unconditional
 - Communication Forwarding - Busy
 - Communication Forwarding - No Reply
 - Communication Diversion Notification - Served-user
 - Communication Diversion Notification - Reminder
 - Communication Diversion Notification - Caller
 - Communication Diversion - to Voice Mail
 - Communication Diversion - Operator Blacklist

Optional features

- Communication Forwarding - Origination
 - Communication Forwarding - Time
 - Communication Forwarding - Media
 - Communication Forwarding - Not Logged-in



- Communication Forwarding - Presence
- Communication Forwarding - Anonymous
- Communication Deflection

In general, the CDIV service in MTAS is rule-based, i.e. the end-user can create rules with different conditions and CDIV determines if the communication shall be diverted by evaluating these rules. The mechanism is very flexible where rules are built up with different conditions and a forward-to action. They can be combined in many ways to express if a communication shall be diverted or not and if notifications shall be sent.

The rule sets include one or more rules, each rule having one or more conditions as illustrated in the figure below:

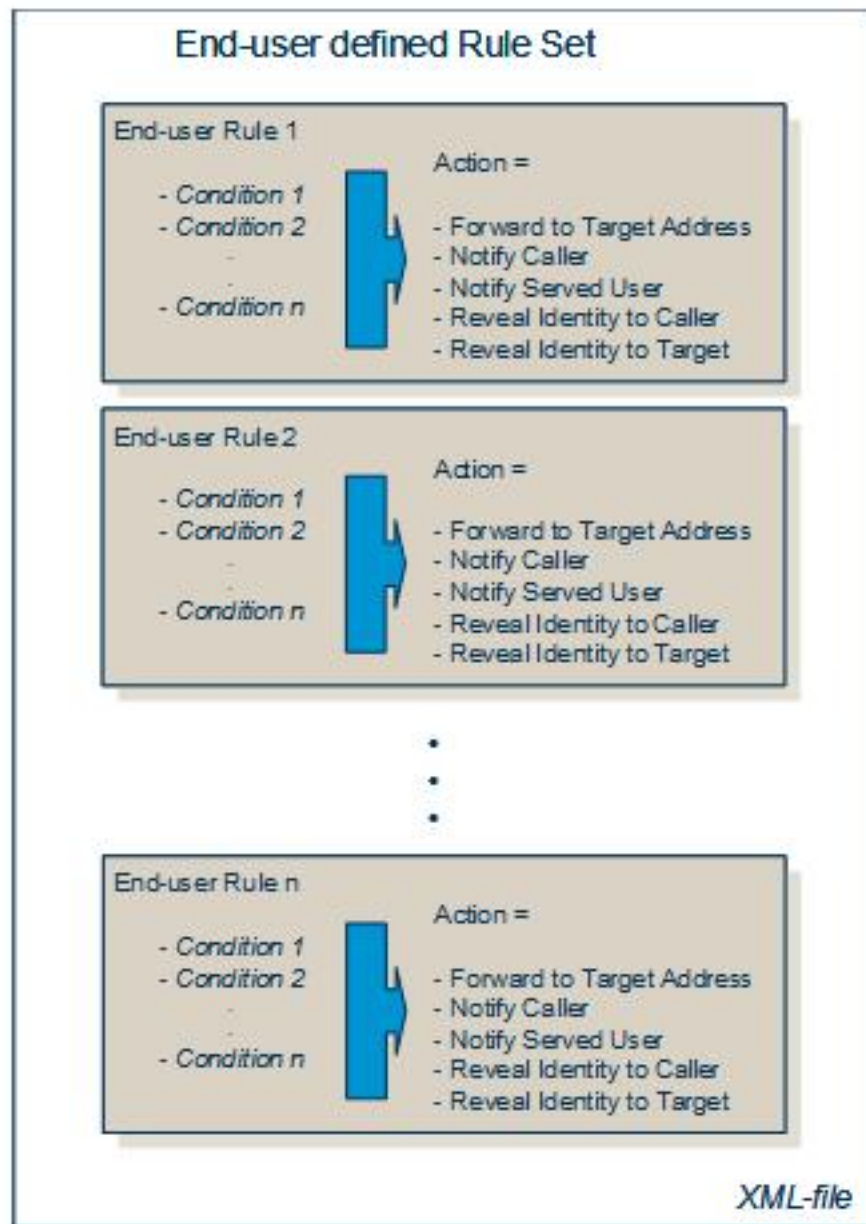


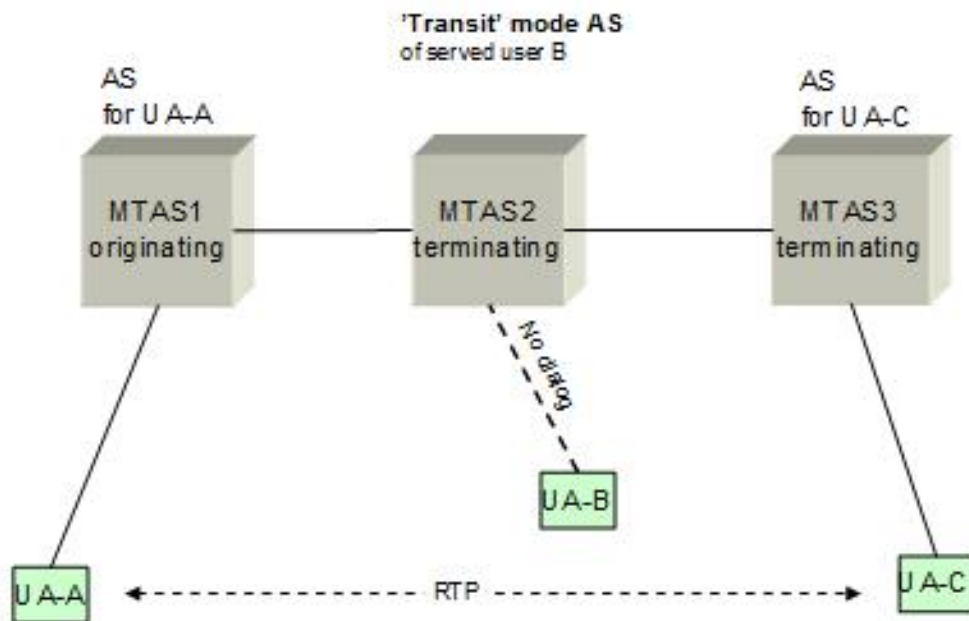
Figure 1 – Forwarding rule sets, examples

Each rule in the full set of rules is evaluated, from top to bottom.



Regarding charging of CDIV services, the terminating AS of the served user becomes a transit AS when the communication is diverted. The served user's outgoing originating services are executed when served user B's AS sets up the dialog to the diverted to user C.

As the figure below illustrates, the served user's terminating AS will be kept in the message path, and shall change from a terminating AS mode to a 'transit' mode.



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.

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

The following service specific AVPs are applicable to diversion charging:

- Supplementary Service Information - indicating the type of diversion or deflection.
 - Requested Party Address - identifying the original called party.



- Related ICID - used to correlation purposes.

For configuration, the operator can activate/deactivate the CDIV service for the subscriber and control which individual conditions and actions they are allowed to use by providing the XML structure to the user's Subscriber Data.

The end-user configures his/her diversion rules via the Ut-interface and the operator accesses them through EMA (Ericsson Multi Activation) using CAI3G-interface.

The served user can also manage some type of his/her Communication Forwarding service via Supplementary Service Codes (SSC). For more information please refer to "Self Administration via Service Codes" feature.

The operator can configure the general behavior of the Communication Forwarding feature. The following node-level options are available which are configured via LDAP:

- Communication Forwarding enable/disable
The operator can enable/disable the Communication Diversion in MTAS.
 - Max number of Communication Forwarding in the network
The operator can set the maximum number of times the same communication is allowed to be forwarded.
 - Audio Announcement enable/disable
The operator can enable/disable the Communication Forwarding service network audio announcement.
 - "Audio only" Announcement
The operator can specify which audio announcement to be played to the calling party when a call is forwarded.
 - Video Announcement enable/disable
The operator can enable/disable the Communication Forwarding service network video announcement.
 - "Video only" Announcement
The operator can specify which video announcement, without audio, to be played to the calling party when a call is forwarded.
 - Audio/Video Announcement - Audio part
The operator can specify which audio, associated with a video announcement to be played to the calling party when a call is forwarded.



- Audio/Video Announcement - Video Part
The operator can specify the video part of the audio/video announcement.
- Text for Communication Diversion Notification - Reminder
The operator can define the content of the text message that is sent to remind a served user (i.e. B-party), when an outgoing call is made, that diversions are active. The default value is: "You have active diverts".
- Text for Communication Diversion Notification - Served-user
The operator can define the contents of the text message that is sent to inform a served user (i.e. B-party) that a call addressed to him/her has been diverted elsewhere. The default value is: "Communication Diverted".
- Default Address for Voice Mail
The operator can define the address of the Voice Mail server for all users that do not have a specific Voice Mail server address.

For detailed information on the CDIV supplementary service regarding rules evaluation, condition elements and forward-to actions, please refer to Functional Specification 1/155 17-CXA 109 86/4, Communication Diversion in MTAS.

Communication Forwarding - Unconditional (CFU)

In case of Communication Forwarding - Unconditional (CFU), the first rule does not contain condition therefore each incoming communication will be forwarded.

The CFU only operates on the initial INVITE method and its responses.

The Dial Tone Management service interacts with the CDIV service rule CFU. When a request is made to activate or deactivate CFU, then this can result in a DTM notification being sent.

Please refer to Communication Diversion (CDIV) Service included under 3.7 Communication Forwarding - Unconditional for a general description of the rule based mechanism which is common for all CDIV features.

3.11.4.1

Standards

3GPP TS 24.604 V8.4.0, 3GPP TS 24.628 V8.2.0, OMA OMA-TS-XDM_Core-V1_0, RFC 4244.



3.11.5 Enhancement

The Communication Diversion (CDIV) supplementary services are aligned with TISpan R2 and have been updated for additional licenses, hop-by-hop Ack, inclusion of Communication Forwarding Not Reachable (CFNRc), interaction with Call Admission Control (CAC) and Online Charging, addition of From-Header and standards changes for History-Info.



3.12 Communication Waiting

Feature Identity:	FAJ 131 0423/1 R2A, Rev. B
Feature Type:	Basic in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

3.12.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

MGC:

FAJ 131 0419 ISUP and SIP Interworking: CW Supplementary Service

Terminal impacts and dependencies

Not Specified

3.12.2 Summary

Communication Waiting informs a subscriber that is busy in an ongoing call that another subscriber is trying to set up a session with the busy subscriber.

Price Object INF 901 5045/1, MMTel Base.

3.12.3 Benefits

End-user

The end user can engage in a call and still be reachable by other users.



Operator

The rate of successful session establishments will increase.

3.12.4

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).

With the introduction of multimedia telephony, we see a change in the end-user behavior which e.g. results in changes in the traditional concept of a busy user. In MTAS 3.1 there are two variants of CW available; one that is terminal based and one network based. Both variants are described in this feature.

Terminal based CW

The main difference between the two variants of Communication Waiting is that in the terminal based implementation, the signaling always reaches the terminal. MTAS is not involved in deciding if the end user is busy as it is solely up to the terminal to make that decision.

This enables an end-user to be reachable for incoming communication despite being occupied in e.g. a real time chat session without having the network make the busy decision.

The function is triggered by the initial SIP INVITE method and actuated by sending an INVITE to the served user. It is executed at the served user's MTAS when a terminating communication is attempted.

For each terminating communication attempt the CW function on the MTAS checks whether the served user has an active subscription to the CW service. If so then it provides a Communication Waiting Active (CWA) indication in the INVITE that is sent to the served user's UA. The MTAS does not take into account whether the served user is busy or free at the point where it sends the INVITE. Note that the CW function takes no account of the type of communication session being requested when considering whether or not to add a CWA indication.

The CW function then waits for a response to this CWA indication.

If the CW function receives a response from the served user that CW has been used (that is the served user has been alerted to the waiting communication) it starts a timer on the alerting phase of the waiting communication. It also records the fact that CW has been used by the served user's UA for charging purposes.



The CW function then optionally plays an announcement to the caller to inform them that the served user is being alerted to the waiting communication. The announcement is played once only. If announcements are enabled, this process takes into account the type of session being requested. If the type is voice or video then an audio or video announcement is played, otherwise no announcement is played.

The CW function then waits for a response to the waiting communication which is now in an alerting phase. This may come from the caller or the served user or the timer may expire.

When a response is received, the CW function processes the response and takes the appropriate action. For example if the served user accepts the waiting communication during the announcement then the timer and announcement are stopped and the session between the caller and served user proceeds as for a normal session. The CW function records the outcome of the CW invocation for performance management purposes.

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.

Network based CW

The network based Communication Waiting implementation does not come bring any fundamental differences to the end-user. In this scenario, it is up to the MTAS to decide whether the end-user is busy or not. This variant is more reflecting the traditional network behavior in case of a busy subscriber, and is addressing operators prioritizing backward compatibility with legacy PSTN services.

The CW function works in conjunction with the CAC function, which determines if the served user is in the 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 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)

If CW has been used and the served user has an active subscription to CW the CW function will start an operator configurable alerting timer, and may optionally play an operator configurable announcement to the caller, to inform the caller that the called party has been informed that there is a waiting communication. The use of CW is recorded for charging and performance management purposes.

Upon receipt of a final response or expiry of the alerting timer, the CW announcement and the timer are stopped, if required, and the outcome of the CW usage is recorded for performance management purposes.

The UA may make use of the HOLD service in order to place an existing session on hold while the waiting communication is answered.

3.12.4.1 Standards

3GPP TS 24.615 v 8.0.0

3.12.5 Enhancement

The terminal based feature for CW is extended with a network based variant, bringing alignment with 3GPP CW specification for determination of CW used when ANDUB sent.



3.13 Dial Tone Management

Feature Identity: FAJ 131 0422/1 R3A, Rev. A

Feature Type: Basic in 11B (FD) to 16B (FD)

Technology:

3.13.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

3.13.2 Summary

The Dial Tone Management function enables an end user terminal to be kept informed whether a special-condition-tone to indicate activation of CDIV should be played or not at initiation of a new session establishment.

Price Object INF 901 5045/1, MMTel Base.

3.13.3 Benefits

End-user

The end user is informed when making a call that CDIV is activated by the special-condition-tone being played

3.13.4 Description

The DTM function allows MTAS to notify a SIP UA of the dial tone information of an end-user, whenever the dial tone information changes. The dial tone information reflects whether the end-user has communication diversion CFU active or not.



An end-user's dial tone information can change when the end-user's communication diversion CFU subscription data is changed via a Supplementary Service Code, via the Ut or CAI3G interface on MTAS.

The operator can configure the DTM function regarding:

- Administrative state (locked/unlocked)
 - Routing to I-CSCF hostname

The following counters exist:

- The number of notify signals sent.
 - The number of notify signals that are unsuccessfully sent due to internal error.
 - The number of notify signals that are unsuccessfully sent due to external error.

3.13.4.1 Standards

ETSI TS 183 043 V1.1.1, IETF RFC 3265

3.13.5 Enhancement

The dial tone management function has been enhanced to support recent standardized function according to PES standard (ETSI TS 183 043 V2.3.1) or the earlier pre-standard version introduced in MTAS 3.0. Note that MTAS have added the new standard version and kept the old implementation in order to be backwards compatible. There is now a configuration parameter indicating which mode that is used (pre-standard or standard mode).



3.14 Flexible Identity Presentation

Feature Identity: FAJ 131 0571/1 R1A, Rev. A

Feature Type: Basic in 12A to 16B (FD)

Technology:

3.14.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

3.14.2 Summary

This feature makes it possible for a user which subscribes to the Flexible Communication Distribution service to choose which originating identity that shall be presented to the terminating side.

Price Object INF 901 5045/1, MMTel Base.

3.14.3 Benefits

End-user

Users are offered the possibility to use any of the identities that are connected to his subscription in a flexible way. If, for example, and end user have a common subscription with one mobile device/number and PC-client with a fixed line number then the end user is able to configure his subscription so that also an outgoing call made on the PC-client can use the mobile identity for A-number presentation.

Operator

This feature can generate additional revenues for an operator through subscription to the service.



The Flexible Identity Presentation service allows the operator to differentiate by offering attractive bundled subscriptions.

3.14.4 Description

The Flexible Identity Presentation allows the operator to configure which originating identity that should be presented in case the end user subscribes to the Flexible Communication Service and have several different identities connected to his subscription.

The service does not allow for regular end user self administration as that would make it possible for the end user to use this service for different kinds of fraud (presenting the identity of the country's prime minister for example). It is therefore important that the identity can be trusted and the only way of achieving that is to only allow the operator to set the attributes configuring this service. So no support for Supplementary Service Codes or Ut-interface is available. However it is possible to develop a portal which presents only those identities that are connected to the end user's subscription as alternatives and therefore removes the possibilities for fraudulent usage. The end user configures per phone number which identity that shall be used for an outgoing call. The portal can then update the subscriber profile using the normal provisioning interface (CAI3G). In this way the end user is still able to choose identity in a flexible way but it is still possible to correctly identify which subscription that made a certain call and therefore fraud is not possible.

3.14.4.1 Standards

3GPP TS 24.229, 3GPP TS 23.228



3.15 Hold Communication

Feature Identity:	FAJ 131 0150/1 R1A, Rev. B
Feature Type:	Basic in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

3.15.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

MGC:

FAJ 131 0305 ISUP and SIP Interworking: Communication Hold Supplementary Service

Terminal impacts and dependencies

Not Specified

3.15.2 Summary

This feature enables an end-user to suspend active media stream(s) of an established multimedia session, and resume the media stream(s) at a later time.

Price Object INF 901 5045/1, MMTel Base.

3.15.3 Benefits

End-user



This feature enables an end-user to handle calls more efficiently and flexible.

The following added value is provided to the end user compared to the corresponding feature in PSTN:

- Flexibility for the end-user to decide if only one media stream to another user shall be stopped or if all streams in a session shall be stopped, i.e. only stop audio or only stop video or stop both audio and video streams.
 - Both audio and video announcements can be given to the held party.

Operator

For the network operator this feature can increase revenues as charges can be made for several sessions connected to one user.

3.15.4

Description

The HOLD feature enables the end-user in a session to temporarily request the opposite user to stop sending media and at a later time resume sending of media. The user initiating the HOLD has the possibility to perform the operation either on an individual stream in the session or on all streams in the session.

The HOLD function is triggered by an UPDATE SIP message from a user trying to hold or resume communication. Note that RE-INVITE is also supported, but not recommended.

Depending on the configuration, a network announcement from an external MRFP is played during HOLD to the held user.

The RESUME function is invoked at RE-INVITE or UPDATE. Similar to the HOLD, RESUME can be performed either on only one stream or on a group of streams.

Announcements are stopped if RESUME is detected in the session.

The HOLD feature supports both audio and video announcements.

The audio announcement also can be music to provide a "Music on Hold" service. Note that the announcement to be played is a node-level configurable parameter, i.e. the content configuration per user or group is not supported. For details see next chapter.

For video announcements, video only or video + audio announcements can be configured.



MTAS utilizes an external MRFP in order to play in-band announcements.

The HOLD or RESUME operations will not be inhibited if the announcement device is unable to send announcements. However, the inability will be registered by a counter to keep track of the performance.

The configurable parameters of HOLD/RESUME service are described below. Please note that each of the listed parameters is set on node-level by the operator via LDAP-interface.

- Audio Announcement enable/disable
The operator can decide if an audio network announcement shall be played during communication HOLD. If enabled and the held stream is an audio stream, an audio announcement will be played to a held user.
 - "Audio only" Announcement
The operator can specify which audio announcement is to be played to the held users. As described in the previous chapter the announcement also can be music to provide a "Music on Hold" service. The operator can choose whether a normal announcement or music or a combination of them shall be played on hold.
 - Video Announcement enable/disable
The operator can decide if a video network announcement shall be played. If enabled and the held stream is a video stream, a video announcement will be played to the held users.
 - "Video only" Announcement
The operator can specify which video announcement, without audio, is to be played to the held users.
 - Audio/Video Announcement - Audio part
The operator can specify which audio, associated with a video announcement, to be played to the held users.
 - Audio/Video Announcement - Video part
The operator can specify the video part of the audio/video announcement.

Performance counters are provided to evaluate the usage and quality of service for the HOLD service. Examples are:

- Number of initiated Holds
 - Number of successfully initiated Holds
 - Number of initiated Resume



3.15.4.1

Standards

3GPP TS 22.610



3.16 International and National Toll Restriction

Feature Identity:	FAJ 131 0547/1 R1A, Rev. A
Feature Type:	Basic in 11A to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

3.16.1 Attention

Commercial attention

Specific to some markets like North America

Dependencies

No internal technical dependencies have been defined for this Feature

3.16.2 Summary

The International and National Toll Restriction feature allows the operator to provide services which restricts international or national long distance calls.

Price Object INF 901 5045/1, MMTel Base.

Price Object INF 901 5045/1, MMTel Base.

3.16.3 Benefits

End-user

By using toll restriction the end user avoids making expensive international or long distance calls on his/her subscription.

Operator

The operator is able to charge extra for the service as well as he is able to satisfy the needs of his customers as toll restriction is a frequently asked for service in several markets.



3.16.4

Description

The operator is able to define through configuration certain aspects of the dial plan within his national network. Any call outside the national network is an international call. Within the national network it is also possible to define what is considered a local versus a long distance calls. In most markets this is done through the definition of area codes but for some markets (typically very large countries) area codes could cover large areas so that a local call is only defined as a call within certain number ranges within an area code. The International and National Toll Restriction feature facilitates these configurations and introduces two new end user services "National Toll Restriction" and "International Toll Restriction" which can be provisioned to the end user and also configured by the end user using a portal. The end user may choose the two services ("National Toll Restriction" and "International Toll Restriction") independently.

In case a called is barred as a result of "National Toll Restriction" or "International Toll Restriction" then a special identifier can be added in charging output to indicate the reason for the barring.



3.17 Malicious Communication Identification

Feature Identity:	FAJ 131 0500/1 R2A, Rev. A
Feature Type:	Basic in 14B (FD) to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

3.17.1 Attention

Commercial attention

For legal requirements

Dependencies

No internal technical dependencies have been defined for this Feature

3.17.2 Summary

MTAS offers the Malicious Communication Identification (MCID) service to its subscribers. The MCID service makes it possible for a subscriber to register a recent incoming communication as malicious.

Price Object INF 901 5045/1, MMTel Base.

3.17.3 Benefits

End-user

Malicious Communication Identification is invaluable in helping to trace and put a stop to malicious communication towards an end-user.

Operator

There is in many countries a regulatory requirement on operators to provide an MCID service.

MCID may be charged by administrative fees.



3.17.4 Description

MTAS can use Diameter messages to convey details of the malicious communication to the Communication Details Server (CDS) or output the information in a local file.

MTAS 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. For temporary mode MCID, MTAS 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 MTAS node. MCID has two modes, permanent and temporary.

- In permanent mode, the details of all incoming communications to the served user are reported to the CDS.
- In 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 to the CDS.

The sub?functions for MCID are:

Register malicious caller temporary mode:

- The action taken by a served user to register an incoming communication as malicious. Registration is done using SSCs.

Register malicious caller permanent mode:

- The action taken by the MTAS node to register an incoming communication to a served user who has permanent mode MCID.

Store Communication Details:

- MTAS stores information about all incoming communications.

Manage service data for registered subscribers:

- The main function implemented is subscriber data management for registered subscribers. When a de-registration is received or when the registration timer expires, the IRS and transparent data is purged immediately along with any incoming communication details stored.

Configure service:



- The MCID service is enabled at node level

3.17.4.1 Standards

3GPP TS 24.616 V8.2.0

3.17.5 Enhancement

In MTAS 14A the possibility of writing the MCID information to a local file is added. With this approach neither the DIAMETER interface nor the Call Detail Server (CDS) is required to offer this service.



3.18 Malicious Communication Rejection

Feature Identity:	FAJ 131 0525/1 R1A, Rev. B
Feature Type:	Basic in 11A to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

3.18.1 Attention

Commercial attention

This is a feature used for legal requirements, alternative feature for non legal requirements is Dynamic Blaklist.

Dependencies

No internal technical dependencies have been defined for this Feature

3.18.2 Summary

Malicious Communication Rejection

Price Object INF 901 5045/1, MMTel Base.

3.18.3 Benefits

End-user

Malicious Communication Rejection prevents incoming calls from malicious callers.

Operator

There is in many countries a regulatory requirement on operators to provide an MCID service. Operators can then enhance the value of the MCID service by letting the end user use the information given by MCID to prevent unwanted calls.

MCR may be charged by administrative fees.

**3.18.4****Description**

End users may activate the service either via portal or via supplementary services codes. In case the service is active, MTAS will check the originating identity of an incoming call to see if it matches any entry in a list of malicious callers. The list of malicious identities is generated by the MCID service. In case there is a match between the incoming callers identity and an entry in the list of malicious callers the call will be rejected.

3.18.4.1**Standards**

3GPP TS 24.611 V8.2.0



3.19 Network Provided Location Information Retrieval in MMTel AS

Feature Identity: FAJ 131 0698/1 R1A, Rev. A

Feature Type: Basic in 14B (FD) to 16B (FD)

Technology:

3.19.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

SBG:

FAJ 131 0505/6 3GPP R8 P-CSCF

HSS:

FAJ 901 504/2 R2A CS/PS/EPS User Data Access

Terminal impacts and dependencies

Not Specified

3.19.2 Summary

Network Provided Location Information (NPLI) is a standardised function for fetching location information. NPLI can be used by other features and functions to provide location based services and prevent fraud related to location.

3.19.3 Benefits

End User



The End User gets access to correct location information which is used to provide various types of location based services.

Operator

The operator gets a correct location information, preventing fraud, and is able to offer various location based services.

3.19.4 Description

Upon request by a service, MMTel AS is able to request Network Provided Location Information (NPLI), a Cell Global Identity (CGI), for a mobile subscriber from HSS/HSS-FE using Sh interface. Requesting NPLI utilizes the Sh-Pull mechanism.

In the request, MTAS indicates to HSS that location is to be retrieved from MME in the PS domain. Following a response from HSS, MTAS extracts the NPLI information and insert it in the PANI header. If NW PANI exists in the received SIP message, MTAS is able to insert NPLI on top and if NW PANI has not been received, MTAS generates a new NW PANI. MTAS uses the topmost NW PANI which has preference over UE PANI and other NW PANIs possibly added by P-CSCF.

Upon request by a service, originating-MTAS can invoke NPLI retrieval to include the PANI in the SIP INVITE for user A while terminating-MTAS can invoke NPLI retrieval to include the PANI in the SIP response (first 18x or 200) for user B.

3.19.4.1 Standards

3GPP TS 29.328



3.20 Network Provided Location Information retrieval in SCC-AS

Feature Identity: FAJ 131 0699/1 R1A, Rev. A

Feature Type: Basic in 14B (FD) to 16B (FD)

Technology:

3.20.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

SBG:

FAJ 131 0505/6 3GPP R8 P-CSCF

HSS:

FAJ 901 504/2 R2A CS/PS/EPS User Data Access

Terminal impacts and dependencies

Not Specified

3.20.2 Summary

Network Provided Location Information (NPLI) is a standardised function for fetching location information. NPLI can be used by other features and functions to provide location based services and prevent fraud related to location.

3.20.3 Benefits

End User



The End User gets access to correct location information which is used to provide various types of location based services.

Operator

The operator gets a correct location information, preventing fraud, and is able to offer various location based services.

3.20.4 Description

SCC AS is able to request Network Provided Location Information (NPLI), a Cell Global Identity (CGI), for a mobile subscriber from HSS/HSS-FE using Sh interface. Requesting NPLI utilizes the Sh-Pull mechanism.

Following a response from HSS, MTAS extracts the NPLI information and insert it in the PANI header. If NW PANI exists in the received SIP message, MTAS is able to insert NPLI on top and if NW PANI has not been received, MTAS generates a new NW PANI. MTAS uses the topmost NW PANI which has preference over UE PANI and other NW PANIs possibly added by P-CSCF.

3.20.4.1 Standards

3GPP TS 29.328



3.21 Number Normalisation

Feature Identity:	FAJ 131 0187/1 R1A, Rev. B
Feature Type:	Basic in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

3.21.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

3.21.2 Summary

MTAS supports SIP URI and tel URI normalisation for originating sessions, e.g. converting the SIP or tel URI into an international E.164 form.

This is an Ericsson specific service.

Price Object INF 901 5045/1, MMTel Base.

3.21.3 Benefits

End-user

End-users do not have to remember the format (SIP or Tel) used when configuring subscriber data. As MTAS works with normalized numbers, the right behavior is guaranteed irrespective of format used.

3.21.4 Description

SIP and tel URIs can be presented to the MTAS via a number of interfaces, these being:

- Ut-interface



- ISC interface from the S-CSCF
- MTAS/CAI3G interface

The MTAS Number Normalisation feature is capable of normalising 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 a 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 normalisation MTAS updates the input-URI with the normalised number or Null if normalisation is not possible.

The number normalisation output may be a:

- Global E.164 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

MTAS provides the operator a number of node-level configuration parameters which affects the behavior of the Number Normalisation. The parameters are managed by LDAP. Examples are:

- Profile name in string for which the Number Normalisation data will be defined.
255 possible profiles may be configured. By default the profile should be a country name.
 - Warning text string that defines the nature of number Normalisation failure
 - String of the rules context consisting of digits or domain name.

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



- Number of input string which is not recognised as a valid tel or SIP URI or is deemed to be a malformed tel or SIP URI
 - Number of failed attempts to normalise a NSN, OSN, or normal tel or SIP URI number because no rules contexts could be found to match against the contexts provided

3.21.4.1 Standards

ITU-T E.164, RFC 3966, RFC 2396



3.22 Number Translation

Feature Identity: FAJ 131 0631/1 R1A, Rev. B

Feature Type: Basic in 13A to 16B (FD)

Technology: IMS

3.22.1 Attention

Commercial attention

Not applicable

Dependencies

This Feature has no dependencies.

3.22.2 Summary

The Number translation function translates dialed local or global numbers based on translation rules defined by the operator.

Price Object INF 901 5045/1, MMTel Base.

3.22.3 Benefits

End-user

N/A. This is an operator service.

Operator

The operator is handle dial plan issues in a more flexible manner.

3.22.4 Description

The Number translation function translates dialed local or global numbers based on translation rules defined by the operator.



The number translation function is similar to the number translation function but is configured and executed independently from number translation. Number translation and number normalization are executed at different point in the call setup phase. Number normalization is executed after dialed number mapping while number normalization is executed after User Equal Phone Error Correction and Dialed String analysis are performed.

Number translation is controlled by a set of regular substitution rules which are configured by operator.



3.23 Originating Identity Presentation

Feature Identity: FAJ 131 0151/1 R4A, Rev. A

Feature Type: Basic in 14B (FD) to 16B (FD)

Technology:

3.23.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

MGC:

FAJ 131 0299 ISUP and SIP Interworking: CLIP/CLIR PSTN Services

Terminal impacts and dependencies

Not Specified

3.23.2 Summary

This feature makes it possible for a terminating user to see the identity of the originating user at the initiation of a communication.

If Identity Restriction is requested then a number of headers will be removed or anonymized.

Any request to restrict the identity is ignored if the Override option is active.

Price Object INF 901 5045/1, MMTel Base.



3.23.3 Benefits

End-user

Users find it useful to know who is calling or to be able to ignore the call, note the number and call back later.

The OIR Override feature can be particularly important for the security authorities.

Operator

This feature can generate additional revenues for an operator through subscription to the service.

The OIR Override is in many markets a mandatory regulatory service.

3.23.4 Description

Originating Identity Presentation (OIP) presents the originating user's identity to the terminating users. The OIP feature is executed on behalf of the terminating user and is invoked on the terminating MTAS.

The main case for OIP is that the terminating user has OIP and then no actions are performed in the MTAS. In the alternative case where the terminating user does not have OIP MTAS will remove identity information from the messages.

A SIP message triggers the execution of the feature, e.g. INVITE.

The OIR Override (Originating Identity Restriction Override) is part of the OIP feature. The 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 feature is usually only enabled for specific users.

In case that the served user has the OIR Override the MTAS is responsible to remove all "Privacy" header fields to ensure that the identity information is not removed by the network. All other headers containing identity are passed transparently.

There is one common administrative state of the Identity Presentation function in MTAS. The operator can enable/disable all presentation sub-features on node-level (i.e. OIP, OIR, TIP, TIR). This node-level configuration is performed by the operator via LDAP.

MTAS enables the operator to configure the service on user-level with EMA (Ericsson Multi Activation) using CAI3G-interface:



- OIP Activation (activate/deactivate)
 - OIR Override Activation (activate/deactivate)

MTAS provides the end-user the following configuration option:

- OIP Activation (activate/deactivate)

The end-user can configure his/her setting via the Ut-interface.

MTAS provides a number of performance counters to evaluate the usage and quality of service. These counters are:

- Number of successful invocations without Override
 - Number of successful invocations with OIR Override
 - Number of times the OIP service was invoked and the action was to restrict the originating identity

Enhancement

In addition to the standardized way to present identities MTAS is also able to present the content of the 'From header'.

3.23.4.1

Standards

3GPP TS 24.607, IETF RFC 3323, IETF RFC 3325

3.23.5

Enhancement

In MTAS 14A support for lack of CLI has been added. MTAS can provide information about the reasons why the identity is not present such as the call comes from a pay phone, or that call originates in a network not providing identity information at the interconnect.



3.24 Originating Identity Restriction

Feature Identity:	FAJ 131 0152/1 R2A, Rev. B
Feature Type:	Basic in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

3.24.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

MGC:

FAJ 131 0299 ISUP and SIP Interworking: CLIP/CLIR PSTN Services

Terminal impacts and dependencies

Not Specified

3.24.2 Summary

This feature makes it possible for an originating user to restrict the presentation of the identity to the terminating user.

The service exists in two modes, permanent and temporary. In the temporary mode it is possible for the served user to determine whether privacy is requested on calls or not, and to switch privacy on and off on a per call basis. In permanent mode this is not possible.

Price Object INF 901 5045/1, MMTel Base.



3.24.3 Benefits

End-user

This feature gives the end-user control over who can identify his/her identity.

An example of where the feature can be applied is a Call Center with agents, often working from home, where OIR will prevent the called customer to call back to the Call Center agent as the identity is not presented.

Operator

This feature can generate network operator revenues from subscriptions or from activation per call.

3.24.4 Description

Originating Identity Restriction (OIR) restricts the originating user's identity from being presented to the terminating users. The MTAS responsibility is to indicate in the signaling that the originating user wants to restrict the presentation of the identity.

The OIR feature is executed on behalf of the originating user and is invoked on the originating and transit MTAS.

The main case for OIR is that the originating user has OIR active. The originating MTAS will in that case add "Privacy" header fields in accordance with the originating user's service settings.

A SIP message triggers the execution of the feature, e.g. INVITE.

MTAS supports two kinds of identity restriction levels at OIR:

- „Restrict asserted identity"
It corresponds to "id-level" + "user-level" privacy
 - "Restrict all private information"
It corresponds to "id-level" + "user-level" + "header-level" privacy

In case of "Restrict asserted identity" the following SIP headers are affected: Call-ID, Subject, Call-info, Organization, User-Agent, Reply-To, In-Reply-To, From, Server, Warning, Referred-By, P-Asserted-Identity, P-Preferred-Identity.

In case of "Restrict all private information", in addition to the previously listed headers, the following SIP headers are affected: Via, Contact and Record-Route.



MTAS supports OIR in two modes:

- Permanent mode
For an originating user that subscribes to the OIR service in permanent mode, MTAS updates the Privacy header based on the subscription option.
 - Temporary mode
The OIR in temporary mode can be initiated on a per call basis by using SSCs (Supplementary Service Codes). The SSC can be interpreted either as "Restriction" or "No Restriction",

Example)

*31*DN# will indicate "Restriction"

#31*DN# will indicate "No Restriction"

(DN = Directory Number of the called party)

There is one common administrative state of the Identity Presentation function in MTAS. The operator can enable/disable all presentation sub-features on node-level (i.e. OIP, OIR, TIP, TIR). This node-level configuration is performed by the operator via LDAP-interface.

MTAS enables the operator to configure the service on user-level with EMA (Ericsson Multi Activation) using CAI3G-interface:

- OIR Activation (activate/deactivate)
 - Mode (permanent mode/temporary mode)
 - Restriction (restrict asserted identity / restrict all private information)

MTAS provides the end-user the following configuration options:

- OIR Activation (activate/deactivate)
 - Default behavior (presentation restricted / presentation not restricted)
 - Activate/deactivate temporary mode identity presentation

The end-user can configure his/her settings via the Ut-interface and via SSC. Note, that the latter supports the "OIR Temporary Mode" only.



A number of performance counters are provided by MTAS to evaluate the usage and quality of service, like:

- Number of successful invocations with OIR in permanent mode
 - Number of sessions where the user, with OIR in temporary mode, decides on a per call basis to allow presentation
 - Number of sessions where the user, with OIR in temporary mode, has the presentation of the identity restricted on a per call basis

3.24.4.1 Standards

3GPP TS 24.607, IETF RFC 4244

3.24.5 Enhancement

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3.25 Self Administration via Service Codes

Feature Identity:	FAJ 131 0163/1 R8A, Rev. A
Feature Type:	Basic in 15B (FD) to 16B (FD)
Technology:	WCDMA, LTE, ANSI, IMS

3.25.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

FAJ 131 0186 Self Administration via Ut-Interface (for generic SSC commands)

Other node impacts and dependencies

Not Applicable

Terminal impacts and dependencies

Not Specified

3.25.2 Summary

Self Administration via Service Codes enables users to gain access to and control supplementary services by using Service Code Commands.

MTAS provides two kinds of supplementary service code commands:

- System defined SSC commands
- Generic SSC commands that are configurable

Price Object INF 901 5045/1, MMTel Base.



3.25.3 Benefits

End-user

Thanks to this flexible configuration method end-users can easily customize their communication supplementary services.

Operator

The feature off-loads the operator from end-user administration resulting decreased OPEX.

By using generic SSC commands an operator can configure SSC's already existing in legacy network when migrating users to IMS. New innovative SSC commands can also be introduced that for instance can define a combination of different services.

3.25.4 Description

The service is independent of user access type and includes all necessary procedures to activate, deactivate, disable, interrogate and invoke a supplementary service. A service code command is received in MTAS within an INVITE message.

MTAS provides two kinds of supplementary service code commands:

- System defined SSC commands
- Generic SSC commands

System defined SSC commands represent SSC definitions having a system defined managed object class per supported supplementary service and SSC action. System defined SSC commands are available for a pre-defined set of supplementary services.

Generic SSC commands provide a flexible mechanism to customize definitions of SSC commands without the syntax restriction as of system defined SSC commands. Generic SSC commands can be used for all supplementary services supported.

MTAS provides solution for a user to gain access to, and control of, the following supplementary services:

System defined SSC Commands:

The main use cases of the system defined SSC commands are:



- Activation of supplementary services
- Deactivation of supplementary services
- Interrogation of supplementary services
- Invocation/Disabling of supplementary services on a per call basis
- Modification of PIN
- Data Management
- Interrogation / Revocation of queued service requests (CC service only)

MTAS supports system defined SSC commands for control of the following supplementary services:

- Communication Forwarding Unconditional (CFU)
- Communication Forwarding on Busy (CFB)
- Communication Forwarding on No Reply (CFNR)
- Communication Forwarding Do Not Disturb (DNDCF)
- Communication Forwarding Unconditional to Voice Mail (CFUVM)
- Communication Forwarding on Busy to Voice Mail (CFBVM)
- Communication Forwarding on No Reply to Voice Mail (CFNRVM)
- Communication Forwarding on Busy and No Reply to Voice Mail (CFBNRVM)
- Communication Forwarding Do Not Disturb to Voice Mail (DNDCFVM)
- Originating Communication Barring (OCB)
- Anonymous Communication Rejection (ACR)
- Do Not Disturb Communication Barring (DNDCB)
- Modification of PIN
- Originating Identification Restriction (OIR) in temporary mode (Dynamic



- ad hoc Identification Presentation/Restriction)
- Originating Identity Presentation (OIP)
- Originating Identity Restriction (OIR)
- Terminating Identity Presentation (TIP)
- Terminating Identity Restriction (TIR)
- Communication Diversion All (CDiv All)
- CNIP function within Originating Identity Presentation.
- Communication Waiting (CW)
- Malicious Communication Identification (MCID)
- Malicious Communication Rejection (MCR)
- Dynamic Black List (DBL)
- Abbreviated Dialing
- Communication Completion (CC)
- Flexible Communication Distribution Divert Primary (FCDDP)
- Explicit Communication Transfer (ECT)
- Invocation of Voicemail (VM)
- Communication Forwarding Not Logged in (CFNL)
- Call Forwarding Conditional (CFCOND)
- Communication Forwarding Not Logged in to Voice Mail (CFNL)
- Communication Forwarding on Busy and No Reply to Voice Mail (CFBNRVM)
- Hotline
- Call Return
- Multiple Subscriber Number (MSN)



MTAS supports each of the three major code schemes specified by ITU-T Recommendations, E.131:

- 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.

Below some examples are listed which are prepared according to the ETSI coding:

*61*PIN#	Activation of Comm. Forwarding on No Reply (CFNR) with PIN
#21#	Deactivation of Comm. Forwarding Unconditional (CFU)
*34*PIN*BP#	Activation of a Barring Program (BP) with PIN
#31*DN per	Disabling of Originating Identification Restriction (OIR) on a call basis (DN: Destination Number)

There are two levels of configuration that are performed by the operator: node-level and user-level:

- Node-level configuration is performed by operator using LDAP protocol and it allows the operator to customize the feature. For example, this includes configuration of service code command syntax and parameters, administrative state of the function, voice announcement codes, etc.
 - User-level configuration is performed by operator through EMA (Ericsson Multi Activation) using CAI3G-interface and it allows the operator to manage service subscription rules for a subscriber. This, for example, includes definition of whether the service is granted to the subscriber, the initial password if required, etc.

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
- etc.

Generic SSC commands:

The Generic SSC (GenSSC) commands provide a flexible mechanism to define SSC commands representing the same set of supplementary service operations as is available to the user via the Ut self-service provisioning interface. Generic SSC commands (GSCC) are defined in an XML document to perform operations on the subscriber service data.

The main use cases of the generic SSC commands are:

- Interrogation of service data
- Update of service data
- Deletion of service data

3.25.4.1**Standards**

ETS 300 738, ITU-T E.131, ETSI TS 183 043, ANSI X3.4-1986, IETF RFC 4825, RFC 2616

3.25.5**Enhancement**

Enhancements in 13A:

Supplementary service code commands added for:

- Hotline
- Call Return

Enhancement in 13A FD1:



- Supplementary service code commands added for Multiple Subscriber Number (MSN)
- Introduction of Generic SSC Commands

Enhancement in 15B:

- support for dynamic announcements in CDIV related Supplementary Service Code commands. The announcement can contain a configurable fix announcement specific to the command and a dynamic part including the forward-to destination number.
-



3.26 Service Centralization and Continuity Application Server

Feature Identity: FAJ 131 0567/1 R3A, Rev. A

Feature Type: Basic in 15A to 16B (FD)

Technology:

3.26.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

3.26.2 Summary

The Service Centralization and Continuity Application Server is a base license for the SCC-AS as such and contains certain basic capabilities for the SCC-AS such as SIP protocol handling and performance counters. At least one of the optional features for the Service Centralization and Continuity Application Server is required to provide services with commercial interest.

Price Object INF 901 5045/44.

3.26.3 Benefits

End-user

This is an operator feature.

Operator

The Service Centralization and Continuity Application Server provides the possibility to offer IMS centralized Services and Single Radio VCC which are key components for a VoLTE solution. For further information see information for the optional features "Single Radio VCC function", "Service Domain Selection" or "Terminating Domain Selection".

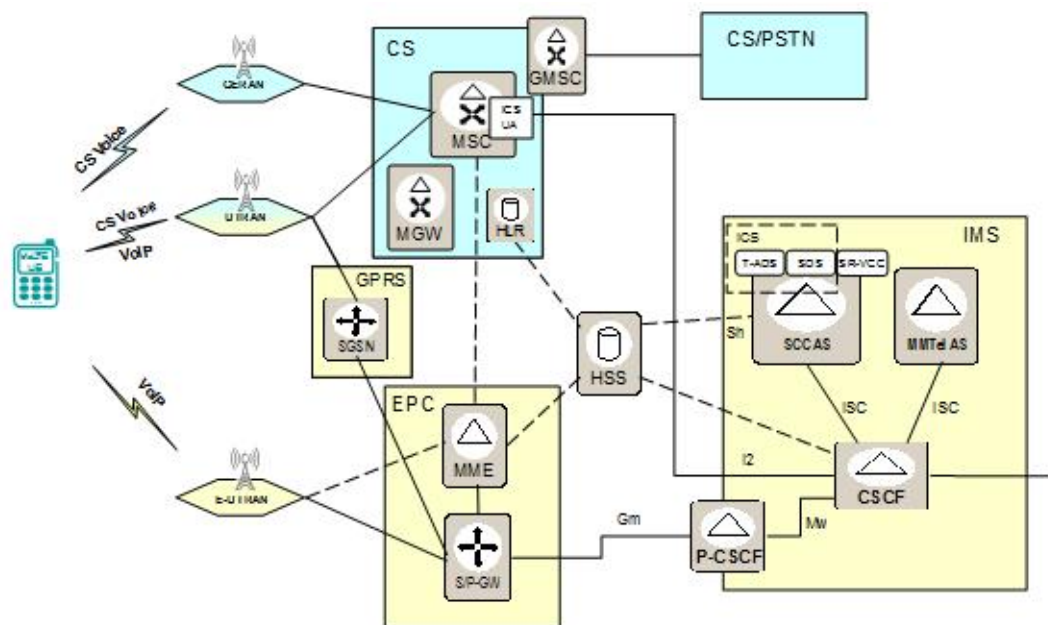


3.26.4

Description

This is an enabler for all the optional features related to the Service Centralization (IMS centralized Services) and Continuity Application Server (Voice Call Continuity). It contains all common parts for the optional features such as SIP handling and common performance countersn and also supports geo redundancy.

SCC AS is a central node in IMS to support IMS services to VoLTE mobiles camping in networks with a mix of CS and continuous LTE coverage, where it can be expected that the mobile moves between LTE and 2G/3G accesses.



Solution overview

The functionality provided by SCC AS is the support for IMS Centralized Services (ICS) which is needed for a single telephony service engine (IMS) for ICS User terminals accessing the network both over LTE PS and 2G/3G CS, and SR-VCC support needed for the transfer of ongoing sessions when the mobile moves from LTE PS to 2G/3G CS coverage.



SCC AS supports geographical redundancy meaning that the IMS system can be configured with two SCC ASs. If the primary SCC AS is restarting or unavailable the secondary SCC AS can be used in the system. The UEs contact information, used by ICS and SR-VCC services, is maintained through third party registrations or from regevent notification and stored internally in SCC AS.

3.26.5 Enhancement

Enhancement in 13A FD1:

- Support for geographical redundancy introduced.

Enhancement in 15A:

- Adding information to CS that this session is subject to IMS Centralised Services "+3gpp.ics" making it possible to handle CS service interaction in MSS such as supression of late call forwarding and announcements.



3.27 SIP Trunking AS Base

Feature Identity:	FAJ 131 0797/1 R2A, Rev. A
Feature Type:	Basic in 15B (FD) to 16B (FD)
Technology:	ETSI, IMS, TSS

3.27.1 Attention

Commercial attention

Not applicable

Dependencies

No external technical dependencies have been defined for this Feature

3.27.2 Summary

ST AS handles managing of access between the operators IMS Network and enterprise PBXs where static and dynamic mode connectivity is supported.

In addition the ST AS serves the operator with Regulatory and Supplementary services.

3.27.3 Benefits

Migration of PBX business to the next generation network of packet switched SIP controlled world of IMS.

3.27.4 Description

3.27.4.1 SIP Trunking Connection and Control functions

These are functions supporting the operator connecting the IP-PBXs to the IMS Network.

- Managing the ST AS administrative state
- License Handling



- Feature tag handling
- Policing of concurrent media streams
- Route registration and de-registration
- PBX originating call
- PBX terminating call including route selection
- ST AS failover procedure

3.27.4.2 Supplementary and regulatory services

3.27.4.2.1 SIP Trunking Identity Presentation

- Originating Identity Presentation (OIP)
- Originating Identity Restriction (OIR)
- Terminating Identity Presentation (TIP)
- Terminating Identity Restriction (TIR)

3.27.4.2.2 SIP Trunking Communication Barring

- Rule Based Barring
- Anonymous Communication Barring (ACR)
- Operator Black Lists
- Operator White Lists
- Operator Barring Programs
- Configurable and Rule Based Fixed and Variable Announcements

3.27.4.2.3 SIP Trunking Call Admission Control

- Limiting the number of PBX Trunk Route originating, terminating and total number of sessions



3.27.4.2.4 SIP Trunking Communication Diversion (CDIV)

- Communication Forwarding - Unconditional (CFU)
- Communication Forwarding - Not Reachable (CFNRc)
- Communication Forwarding - Not Logged In (CFNL)
- Communication Diversion Notification - Served-user
- Communication Diversion - Rule based
- Communication Deflection - Communication re-routed from SIP Trunking AS without keeping ST AS in the loop.

3.27.4.2.5 SIP Trunking Number Normalisation

- Normalisation of SIP and Tel URI; handling of P-Asserted Identity, phone context, Global E.164, Service Numbers et cetera

3.27.4.2.6 SIP Trunking Malicious Communication Identification

- Capture, Interrogation and external storage of MCID information for a limited number of extensions behind the PBX SIP Trunk Route

3.27.4.2.7 SIP Trunking Carrier Select

- Carrier Pre-Select Rn
- Carrier Select Rn

3.27.4.3 Standards

ETSI TS 182 025 V3.3.1 (2011-03) Business Trunking Architecture and functional description
3GPP TS 32.240 - Charging Management; Charging architecture and principles
IETF RFC 3892 - The Session Initiation Protocol (SIP) Referred-By Mechanism
IETF RFC 3966 - The tel URI for telephone Numbers
ITU-T E.164 - The international public telecommunication numbering plan

3.27.5 Enhancement

The following additions are done for MTAS 15B:



- Terminating Identity Presentation (TIP)
- Terminating Identity Restriction (TIR)
- Communication Deflection



3.28 Support for IPv6

Feature Identity: FAJ 131 0526/1 R3A, Rev. B

Feature Type: Basic in 14B (FD) to 16B (FD)

Technology:

3.28.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

3.28.2 Summary

MTAS offers the possibility for the operator to use IPv6 interfaces. MTAS can also use either IPv4 or Ipv6 for communication with other IMS nodes e.g. CSCF, HSS and MRFP. This means that MTAS can be deployed in an IPv6 enabled network.

Price Object INF 901 5045/1, MMTel Base.

3.28.3 Benefits

End-user

As the world is running out of IPv4 addresses it will be possible to communicate with more people.

Operator

There are many countries where there are very few available IPv4 addresses compared to the size of the population. In these countries IPv6 is becoming a must. Within the operators own network MTAS can also inter-work with other traffic handling nodes using IPv6 protocol.



3.28.4 Description

Devices with IPv6 addresses can be used both on traffical and O&M interfaces.

MTAS can also interwork with other traffic handling nodes using IPv6 protocol.

It is possible to define which IP version (IPv4 or IPv6) that can be used per interface. It is therefore not necessary to use the same version for all interfaces and this facilitates introduction of IPv6 in more complex network where some nodes may only support IPv4.

3.28.5 Enhancement

Enhancement in MTAS 13A FD1:

- IPv6 also supported on O&M interfaces



3.29 Terminating Identity Presentation

Feature Identity: FAJ 131 0153/1 R1A, Rev. A

Feature Type: Basic in 11A to 16B (FD)

Technology:

3.29.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

3.29.2 Summary

This feature makes it possible for an originating user to see the identity of the terminating user at the initiation of a communication. If Identity Restriction is requested then a number of headers will be removed.

If the Override option is active then any request to restrict the identity is ignored.

3.29.3 Benefits

End-user

The feature is useful for those situations, for example Communication Diversion, when the terminating user is not the same as the called user.

The TIR Override feature can be particularly important for the security authorities.

Operator

This feature can generate additional revenues for an operator through subscription to the service.



3.29.4 Description

The TIP feature is executed on behalf of the originating user and is invoked on the originating MTAS.

The main case for TIP is that the originating user has TIP and then no actions are performed in the MTAS. In the alternative case where the originating user does not have TIP MTAS will remove identity information from the messages.

If the user has TIR Override, the "Privacy" header values "none", "user", "header" and "id" are removed and no other headers are anonymized.

If the served user has the TIP feature but not TIR Override and the request contained a "Privacy" header then this is used to remove the appropriate SIP header.

If the served user does not subscribe to TIP then if the request contained a "Privacy" header then this is used to remove the appropriate SIP header.

A SIP message triggers the execution of the feature, e.g. INVITE.

The 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.

In case that the served user has the TIR override as part of the TIP settings the MTAS is responsible to remove all "Privacy" header fields to ensure that the identity information is not removed by the network. All other headers containing identity are passed transparently.

There is one common administrative state of the Identity Presentation function in MTAS. The operator can enable/disable all presentation sub-features on node-level (i.e. OIP, OIR, TIP, TIR). The node-level configuration is performed by the operator via LDAP-interface.

MTAS enables the operator to configure the service on user-level with EMA (Ericsson Multi Activation) using CAI3G-interface:

- TIP activation (activate/deactivate)
 - TIR Override Activation (activate/deactivate)

MTAS provides the end-user the following configuration options:

- TIP Activation (activate/deactivate)

The end-user can configure his/her settings via the Ut-interface.



MTAS provides a number of performance counters to evaluate the usage and quality of service. These counters are:

- Number of successful invocations with TIR Override
 - Number of times the TIP service was invoked and the action was to restrict the Terminating identity
 - Number of successful invocations of the service with the option TIR override disabled when the action is to not restrict the identity



3.30 Terminating Identity Restriction

Feature Identity:	FAJ 131 0154/1 R2A, Rev. B
Feature Type:	Basic in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

3.30.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

3.30.2 Summary

This feature makes it possible for a terminating user to restrict the presentation of the identity to the originating user.

The service exists in two modes, permanent and temporary. In the temporary mode it is possible for the served user to determine whether privacy is requested on calls or not, and to switch privacy on and off on a per call basis. In permanent mode this is not possible.

Price Object INF 901 5045/1, MMTel Base.

3.30.3 Benefits

This feature gives the end-user control over who can identify his/her number.

This feature is an important service for protecting an end-user's integrity and when legal considerations restrict the presentation of the terminating identity.

Operator

This feature can generate additional revenues for an operator through subscription to the service.



It also enables an operator to attract Call Center companies who needs to protect the anonymity of Call Center agents working from home.

3.30.4 Description

The TIR feature is executed on behalf of the terminating user and is invoked on the transit and 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 fields "privacy: user; id".

A SIP message triggers the execution of the feature, e.g. INVITE.

MTAS supports one identity restriction level at TIR: „Restrict asserted identity". It corresponds to "id-level" privacy + "user-level" privacy.

The following SIP headers are affected: Call-ID, Subject, Call-info, Organization, User-Agent, Reply-To, In-Reply-To, From, Server, Warning, Referred-By, P-Asserted-Identity, P-Preferred-Identity.

MTAS supports TIR in two modes:

- Permanent mode
For a terminating user that subscribes to the TIR service in permanent mode, MTAS updates the Privacy header.
 - Temporary mode

There is one common administrative state of the Identity Presentation function in MTAS. The operator can enable/disable all presentation sub-features on node-level (i.e. OIP, OIR, TIP, TIR). The node-level configuration is performed by the operator via LDAP-interface.

MTAS enables the operator to configure the service on user-level with EMA (Ericsson Multi Activation) using CAI3G-interface:

- TIR Activation (activate/deactivate)
 - Mode (permanent mode/temporary mode)

MTAS provides the end-user the following configuration options:

- TIR Activation (activate/deactivate)
 - Default behavior (presentation restricted/presentation not restricted)
 - Activate/deactivate temporary mode identity presentation



The end-user can configure his/her settings via the Ut-interface.

A number of performance counters are provided by MTAS to evaluate the usage and quality of service, like:

- Number of successful invocations with TIR in permanent mode
 - Number of sessions where the user, with TIR in temporary mode, decides to on a per call basis allow presentation
 - Number of sessions where the user's, with TIR in temporary mode, identity is restricted on a per call basis

3.30.4.1 Standards

3GPP TS 24.608, IETF RFC 4244

3.30.5 Enhancement

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3.31 Three Party Call

Feature Identity:	FAJ 131 0162/2 R2A, Rev. A
Feature Type:	Basic in 12A to 16B (FD)
Technology:	Fixed Broadband

3.31.1 Attention

Commercial attention

This is a feature for fixed access, use Ad-hoc Conference for mobile.

Dependencies

No internal technical dependencies have been defined for this Feature

3.31.2 Summary

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 behavior from previous Three Party Call feature from MTAS 3.0 is still available within the Conference feature which supports ad-hoc (dial-out) conferencing according to TISPAN R1.

3.31.3 Benefits

End-user

Easy to use. Fast and effective method for collaboration.

User behavior for existing PSTN service remains the same.

Operator

Traditional PSTN/ISDN Three Party Conference service can be offered to subscribers.



Three Party Call service may provide additional income per subscriber either by generating more traffic or by Value Added Service fees.

3.31.4 Description

The feature allows a user who is involved in two separate 2-party sessions with another two users to bring these two users in a 3PTY session. Both 2-party sessions have to be put on "hold", before the request for the 3PTY session is issued. The 3PTY originator has a possibility to resume back to 2-party session with any of the 3PTY participants.

The 3PTY service is implemented on the originating MTAS of the user who initiates a 3PTY session. The service is triggered in the MTAS by reception of an initial INVITE that has the Request-URI set to the 3PTY factory URI. The 2-party sessions which need to be included in the 3PTY session are identified by the Session-ID parameters included in the URI list provided in the INVITE which requests creation of the 3PTY session.

The 3PTY service maintains dialogs to the involved participants, implements policies and controls the MRFP. During the establishment phase and throughout the 3PTY session, the 3PTY service interworks with other services.

An important part of the 3PTY service is the MRFP, but this function is viewed as an external actor from the MTAS point of view. The MRFP is a device that is capable of mixing (and also transcoding if needed) media streams.

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

3.31.5 Enhancement

The feature contains updates to reflect PSTN user behavior as described by alternative 3 in TISPA PES, where ongoing calls can be transferred into a conference and back as separate 2-party sessions.



3.32 Time Zone Handling

Feature Identity:	FAJ 131 0672/1 R1A, Rev. B
Feature Type:	Basic in 14B (FD) to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

3.32.1 Attention

Commercial attention

Not applicable

Dependencies

It is possible to set UHTZ per user in Ericsson Charging System (However, it can not be expected to be supported in all OCSs)

3.32.2 Summary

This feature enables the possibility to provision service time zone per user

Price Object INF 901 5045/1, MMTel Base.

3.32.3 Benefits

End-user

This feature enables an end-user to handle individual treatment of served time zone tied to different features like time based services like call diversion and also for correct announcements of time in the call return feature.

Operator

For the network operator having a network spanning multiple time zones this feature makes it possible to provision a user home timezone that can be included in charging AVP's if needed for charging purposes. For instance time dependant charging can be based on the home time zone and not the actual timezone from where the call is made.



3.32.4 Description

Each subscriber can be provisioned with an individual home time zone area, User Home Timezone, UHTZ that also includes compensation for Daylight Saving Time. The time-zone-area is in the form "*Area/Location*" and must be included in the list of time zones in IANA Time Zone Database. Example "Europe/Stockholm"

MTAS can also include this UHTZ in a charging AVP, UHTZ-Offset AVP that indicates the home time zone with compensation for Daylight Saving Time for the user.

3.32.4.1 Standards

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4 General - OPTIONAL FEATURES



4.1 Ad-hoc Conference

Feature Identity: FAJ 131 0185/1 R6A, Rev. A

Feature Type: Optional in 16A to 16B (FD)

Technology: IMS

4.1.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

MRS:

FAJ 121 2413/3 MRFP Basic
FAJ 121 2017/3 Audio Conferencing
FAJ 121 2705/2 MRFC

Optional features:

MRS:

FAJ 121 2019/2 Video Basic
FAJ 121 2423/2 Video Conferencing
FAJ 121 2708/1 Content Sharing

Terminal impacts and dependencies

Not Specified

4.1.2 Summary

The Conference feature allows end-users to start an audio and video conference and to invite other participants to the conference.



Price Object INF 901 5045/3, Conference.

4.1.3 Benefits

End-user

Easy to use. Fast and effective method for collaboration. Access to a new range of multi-party services.

Operator

Ad-hoc PSTN/ISDN Multi Party Conference service can be offered to subscribers.

Conference service may provide additional income per subscriber either by generating more traffic or by Value Added Service fees.

4.1.4 Description

MTAS supports ad-hoc (dial-out) conferencing according to TISPAN R1. Ad-hoc conferencing means that the conference is created automatically, “on the fly”, without any need for reservation, scheduling, etc.

When a conference is created, the conference creator (the user creating or starting the conference) establishes a session to the conference server entity of MTAS using SIP INVITE message. The conference server is acting as a User Agent Server.

An additional participant can be invited to the conference by instructing the conference server to establish a session with the new participant. In this step the SIP REFER request is used, as defined by RFC 3515. In this case, the conference server is acting as a User Agent Client.

An external Media Resource Processor Function (MRFP) receives the media streams, combines media of the same type and redistributes the result to each participant. The MRFP is controlled by the conference server’s internal Media Resource Function Control (MRFC) through the Mp-interface (H.248).

Url-list are supported which improves call setup times for larger Ad-hoc conferences

The conference is terminated when the conference creator leaves the conference. The other conference participants may leave the conference at any time without affecting the rest of the conference.



Conference notifications give the possibility through subscriptions to present conference events to the user. It can be presented when users are Entering, Leaving or put on Hold/resumed.

The maximum number of the participants in a conference is 32 including the conference creator.

The supported media types are audio and video.

The conference server has the capability to support conferences with multiple media types, such as simultaneous audio and video conferences. However, after the conference has been established between the conference creator and the conference server, the server does not support any modification (adding/removing media components, modifying media codec type) neither from the conference creator nor the conference participant(s).

The only parameters that are available to be changed are the media port number and the media direction.

MTAS supports the „Dial - Out” method. The conference creation is very simple and user-friendly. The initiator (A-party) invites the other participants to the conference using MTAS. From this point MTAS takes care about the rest part of the conference establishment. The technical details are described as follows:

After successful creation of a session between the conference creator and the conference server, the creator instructs the conference server to establish a session with the new conference participant. The conference creator sends a SIP REFER request, as defined by RFC 3515, to the server asking to invite the new participant to the conference using the contact information provided in the request. The server in turn sends an SIP INVITE request towards the new participant.

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 the conference creator may invite another participant to the conference.
 - Only the conference creator can remove participants from the conference.
 - Any participant may leave the conference at any time without this affecting the rest of the conference.
 - If the conference creator leaves the conference, the conference is terminated.



- Anyone can change the port and direction of their media.

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

The Conference feature supports offline charging using Diameter based Rf-interface.

The user creating the conference 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.

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.

MTAS provides a number of node-level configuration options which are configured by the operator via LDAP-interface. The following configuration options are available:

- Conference enable/disable
 - A number of conference server specific parameters (e.g. conference factory URI, prefix of the user part of the conference URI, the ISC port number of the S-CSCF where the INVITE from the focus is to be routed to, etc.)

The operator is able to configure the following user-level parameters with EMA (Ericsson Multi Activation) using CAI3G-interface:

- Conference enable/disable per user
 - Maximum number of participants (including the subscriber) per user



There are a number of counters to assist performance management for ad-hoc Three Party Call. This includes monitoring of MTAS node behavior, QoS measurements as well as to provide a number of statistics. The following counters are examples:

- The number of successfully created Conference
 - The number of unsuccessful Conference creation attempts due to node internal reasons
 - The number of unsuccessful Conference creation attempts due to node external reasons
 - The number of currently active Conference

4.1.4.1 Standards

3GPP TS 24.147 v7.7.0, 3GPP TS 24.605, 3GPP TS 29.333, RFC 4353 (February 2006), RFC 4579 (August 2006), RFC 3515, RFC 3891 (September 2004), RFC 3892 (September 2004), RFC 5057 (November 2007)

4.1.5 Enhancement

New in MTAS 13A:

It is now possible to either colocate conference focus and conference factory or have them as separate entities.

P-Served-User and Orig AS chaining has been added

Url-list is added which improves call setup times for larger Ad-hoc conferences

Content sharing using video stream

New in MTAS 15A:

Auto answer detection, to avoid connecting to a users voice mail.

New in MTAS 15B:

Improved presentation information in the conference notifications, like progress information for invitations call set up, Display Name, several anonymous participants can be distinguished from each other.

New in MTAS 16A: Support for segmented variable announcement.



4.2 Add/Drop Media

Feature Identity:	FAJ 131 0169/1 R1A, Rev. B
Feature Type:	Optional in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

4.2.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.2.2 Summary

MTAS supports adding and removing of additional media flow into an existing session between two MMTel end-users, for example the addition of real-time text to an existing audio session. Another example is when a 2-Party audio/video call is modified to remove the video part.

Price Object INF 901 5045/9, Multimedia Communication

4.2.3 Benefits

End-user

The feature enables an end-user to change the communication behavior during an established communication by adding or removing different types of media which will enhance his/her communication experience.

Operator

This new type of communication behavior can generate additional revenue sources, attract new subscriptions and decrease subscriber churn rate.



4.2.4 Description

When adding media, UE-A indicates that it wish to add an extra media flow by sending a RE-INVITE message. UE-B is able to reject the change if it is not acceptable.

Removal of a media flow can be achieved by using an UPDATE request or RE-INVITE message. There is unlikely to be a need to alert the B-Party to obtain permission for the change.

The supported media types are: audio, video, real-time text.

4.2.4.1 Standards

3GPP TS 22.173 Stage 1, 3GPP TS 24.173 Stage 3, 3GPP TS 26.114, IETF RFC 3261, IETF RFC 3264, IETF RFC 3311



4.3 Address Policing

Feature Identity:	FAJ 131 0548/1 R1A, Rev. A
Feature Type:	Optional in 11A to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

4.3.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.3.2 Summary

The address policing feature allows the operator to define rules for destination addresses. It is then possible to identify and take actions if destination addresses does not follow the expected rules.

Price Object INF 901 5045/38, Address policing.

4.3.3 Benefits

End-user

End user will benefit of this feature as it is possible for the operator to inform the end user of for example incorrectly dialed phone numbers.

Operator

Operator can both provide a better end user experience and at the same enforce dial plans.



4.3.4

Description

The Address Policing service ensures that the format of the destination address entered by a caller complies with rules imposed by the operator. The following format checks are supported in MTAS 11A:

- A long-distance mobile number cannot be entered without the National Dialing Prefix (NDP).

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



4.4 Advice of Charge

Feature Identity: FAJ 131 0496/1 R1A, Rev. A

Feature Type: Optional in 11A to 16B (FD)

Technology:

4.4.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

MGC:

FAJ 131 0476 SUP and SIP Interworking: Advice of Charge Supplementary Service

Terminal impacts and dependencies

Not Specified

4.4.2 Summary

The Advice of Charge (AOC) supplementary service allows the served user to be informed of SIP session related charging information. The charging information provided to the served user, and the time at which the information is provided, is dependent on the AOC service type.

Price Object INF 901 5045/16, Advice of Charge.



4.4.3 Benefits

End-user

AOC enables end-users to control the cost of their communication more effectively. Charging information such as tariffs can be received either at setup of the communication, during the communication or at the end of a completed communication. AOC Information can be received from either OCS or from external source.

4.4.4 Description

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.

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).

4.4.4.1 Standards

3GPP TS 32.280 (V8.1.0), 3GPP TS 24.647 (V8.1.0), ETSI TS 183 043 (V2.3.1)



4.5 Anonymous Communication Rejection

Feature Identity:	FAJ 131 0182/1 R2A, Rev. A
Feature Type:	Optional in 14B (FD) to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

4.5.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

MGC:

FAJ 131 0306 ISUP and SIP Interworking: ACR and CB Supplementary Service

Terminal impacts and dependencies

Not Specified

4.5.2 Summary

The feature enables an end-user to automatically reject anonymous incoming communication.

Price Object INF 901 5045/2, MMTel Extended.

4.5.3 Benefits

End-user

The end-user will be able to control his/her anonymous incoming traffic.



He/She can also combine this origin-based barring with other conditions (Time and Media).

Operator

The operator will be able to offer Anonymous Communication Rejection service to its end-users.

It will differentiate the operator from the competitors.

4.5.4 Description

Anonymous Communication Rejection (ACR) is a kind of Incoming Communication Barring, which bars anonymous callers.

The incoming communication is rejected by a SIP response with result code 433 (Anonymity Disallowed).

Please note, that 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.

Thanks to the flexibility of the MTAS rule-based barring feature, the ACR can be combined in many ways with other types of incoming barring to express whether a communication should be barred or not. In this mode, the following barring rule examples can be easily created:

- Bar all anonymous video communication except between 08.00-11.00 2009-10-14
(In this example the conditions are based on ACR + Time)
 - Bar every type of anonymous communication except video between 08.00-11.00 2009-10-14
(In this example the conditions are based on ACR + Media + Time)

MTAS provides the following node-level configuration options which are managed by the operator via LDAP-interface.

- Communication Barring enable/disable
There is one common administrative state of the Communication Barring function in MTAS. The operator can enable/disable all barring sub-features on node-level (i.e. Barring Programs, OCB, ICB, ACR).
 - Audio Announcement enable/disable
The operator can enable/disable the audio announcement when a communication is barred when ACR rejects a communication.



- "Audio only" Announcement
The operator can specify which audio announcement is to be played to the originating party when a communication is barred due to ACR.
- Video Announcement enable/disable
The operator can enable/disable the video announcement when a communication is barred due to ACR.
- "Video only" Announcement
The operator can specify which video announcement, without audio, is to be played to the originating party when a communication is barred due to ACR.
- Audio/Video Announcement - Audio part
The operator can specify which audio, associated with a video announcement, to be played to the originating party when a communication is barred due to ACR.
- Audio/Video Announcement - Video part
The operator can specify the video part of the audio/video announcement.

The user-level configuration is performed by the end-user via the Ut-interface and via SSC (Supplementary Service Codes) and accessed by the operator via EMA using CAI3G-interface.

The following performance counter is provided by MTAS to evaluate the usage and quality of service:

- Number of barred incoming anonymous communications

4.5.4.1 Standards

3GPP TS 24.611, IETF RFC 4745, IETF RFC 3323, IETF RFC 4566, ETSI ES 183 028, IETF RFC 2396, IETF RFC 3966, <http://www.w3.org/TR/2004/REC-xmlschema-2-20041028/>

4.5.5 Enhancement

The feature has been enhanced in MTAS 14A to handle reasons for lack of CLI. In this way calls from for example pay phones or networks not providing CLI at the interconnect point can be handled in a better way.



It is also possible in 14A to configure the feature to charge for the announcement indicating to the A-subscriber that he has been barred.



4.6 AS Controlled Forking

Feature Identity:	FAJ 131 0544/1 R1A, Rev. A
Feature Type:	Optional in 11A to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

4.6.1 Attention

Commercial attention

Not applicable

Dependencies

The following node features are required in order to use IMS system level feature Service Convergence. For more information, see IMS Feature Description.

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

FAJ 131 0492/1 R2A Flexible Communication Distribution

Other node impacts and dependencies

CSCF:

FAJ 131 0554/1 R2A Service Convergence

The following node features are optional for IMS system level feature Service Convergence

SBG:

FAJ 131 0453 SIP Message Manipulation & Protocol repair

Terminal impacts and dependencies

Not Specified

4.6.2 Summary

The AS controlled forking feature allows MTAS to recognize individual devices



Price Object INF 901 5045/33, AS Controlled Forking

4.6.3 Benefits

End-user

End users will benefit from being able to use the FCD service (parallel and sequential ringing) also towards individual devices.

Operator

Operators will be able to offer more attractive services related to parallel and sequential ringing where it is also possible to address individual devices and not only 'targets' (phone numbers).

4.6.4 Description

MTAS uses a mechanism based on RFC 3840 which allows MTAS to influence forking which still takes place in CSCF. The effect is that it will be possible to define parallel or sequential ringing patterns towards individual devices for subscribers of the FCD (Flexible Communication Distribution) service.



4.7 Business Mobility Base

Feature Identity:	FAJ 131 0685/1 R1A, Rev. C
Feature Type:	Optional in 14B (FD) to 16B (FD)
Technology:	GSM, WCDMA, Fixed Broadband, Wireless Broadband, IMS

4.7.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

FAJ 131 0685 Business Mobility Base

Other node impacts and dependencies

Not Applicable

Terminal impacts and dependencies

Not Specified

4.7.2 Summary

Business Mobility Base is a base license enabling integration of business users on PLMN access of GSM terminals into an IMS environment.

The telephone service can be delivered by the IMS domain in order to provide evolved service delivering more value for the end users including end users with a non-SIP phone.

In order to deliver a useful service at least one of the following optional services on top of this base package must also be ordered:

- Business Mobility Dynamic Service Domain selection



Price Object INF 901 5045/67

4.7.3 **Benefits**

End-user

This is an operator feature.

Operator

Operator can deliver IMS services also to Business users not having a SIP phone.

For further information see information for the optional features listed in summary.

4.7.4 **Description**

This is an enabler for the optional features available to the Business Mobility Base. It contains all common parts for the optional features such as SIP handling and common performance counters and supports also geo redundancy.

Business Mobility Base is an enabler to provide IMS services to mobile phones on 2G/3G accesses. This provides support for IMS Centralized Services (ICS) which is needed for a single telephony service engine (IMS) for ICS User terminals accessing the network both over LTE PS and 2G/3G CS.

SCCAS supports geographical redundancy meaning that the IMS system can be configured with two Business Mobility ASs. If the primary AS is restarting or unavailable the secondary AS can be used in the system.



4.8 Business Mobility Dynamic Service Domain Selection

Feature Identity:	FAJ 131 0684/1 R1A, Rev. B
Feature Type:	Optional in 14B (FD) to 16B (FD)
Technology:	GSM, WCDMA, IMS

4.8.1 Attention

Commercial attention

Requires Business Mobility Base

Dependencies

The following feature is always required:

- MTAS: FAJ1310685 - Business Mobility Base

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

FAJ 131 0685 Business Mobility Base

Other node impacts and dependencies

Not Applicable

Terminal impacts and dependencies

Not Specified

4.8.2 Summary

The Service Domain Selection function selects IMS as service domain for Business Users Mobile CS UE and VoLTE UE originating at 2G/3G CS access.

Price Object INF 901 5045/69



4.8.3 Benefits

End-user

This is an operator feature.

Operator

Provides the operator with the capability to handle enterprise CS devices in IMS solutions. IMS based enterprise services can thereby be offered also to users using CS handsets with a possible future possibility to migrate these users to IP access like VoLTE keeping still delivering the same end user service.

4.8.4 Description

MTAS provides a standard compliant (3GPP R10) implementation for SCC-AS. Both I2 and Mg interfaces are supported. The Service Domain selection is part of IMS Centralized Services (ICS) which provides one service engine handling multiple accesses (CS and PS). Having only one service engine makes it easier to handle service interactions and provide a consistent end user experience irrespective of used access.

The Service Domain Selection function can anchor calls in IMS based on conditions configured in the dynamic SDS function.

Conditions that can be used for dynamic SDS are:

- the media type of the call,
- if subscriber is Roaming or not,
- if the called party number qualifies as a Local number and
- if called party number entered with an Escape code.

The result of dynamic SDS is then to either continue the call in CS domain or to anchor in IMS with an IMRN.

4.8.4.1 Standards

3GPP TS 24.292



4.9 Call Admission Control

Feature Identity:	FAJ 131 0495/1 R1A, Rev. B
Feature Type:	Optional in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

4.9.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.9.2 Summary

The Call Admission Control (CAC) feature allows the operator to define different subscription types with unique policies, e.g. a subscription with multiple numbers under one and the same subscription. The related policy can state for example max number of simultaneous calls.

Price Object INF 901 5045/17, Call Admission Control.

4.9.3 Benefits

End-user

End-users can be offered tailor made subscriptions based on usage.

Operator

The feature gives the operator control over the amount of traffic in the network. This can be used to guarantee Quality of Service and it will also enable differentiation through attractive tariff models.

CAC prevents fraud as it will be possible for the operator to control the number of parallel calls over the same broadband connection.



4.9.4

Description

The User Call Admission Control (CAC) supplementary service 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 Group CAC supplementary service provides two options for the specification of the call limits for the CAC Group:

- specification of the call limits in the xml data for the CAC group
 - specification of the limits in a CAC profile on the MTAS with the CAC Group xml data indicating the profile to be used.

Specification of the call limits in the xml data enables the operator to restrict:

- the number of all sessions all the users in the group are involved in
 - the number of all originating sessions all the users in the group are involved in
 - the number of all terminating sessions all the users in the group are involved in
 - the number of active sessions all the users in the group are involved in
 - the number of active originating sessions all the users in the group are involved in
 - the number of active terminating sessions all the users in the group are involved in



Specification of the call limits in a CAC profile on the MTAS enables the operator to restrict based upon combinations of the following:

- calls to or from specific destinations, based upon number or domain matching (full or partial matching)
 - the direction of the call, originating, terminating or all calls

The User and Group CAC services check the user and group counts against the appropriate limits, as configured for the user and group. Group CAC limits may be configured in either the Group CAC xml data or a CAC Profile indicated to be used for the CAC Group.

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.

4.9.4.1

Standards

RFC 3261, RFC 4511, RFC 4566



4.10 Call Return

Feature Identity:	FAJ 131 0633/1 R2A, Rev. B
Feature Type:	Optional in 14B (FD) to 16B (FD)
Technology:	Fixed Broadband

4.10.1 Attention

Commercial attention

Not applicable

Dependencies

This Feature has no dependencies.

4.10.2 Summary

Call return is a supplementary service that has been historically used mainly in wireline networks. It allows the end user to get information of the most recent incoming call attempt and gives the opportunity to call back to the caller of that call attempt. In certain markets this is the most commonly used supplementary service.

Price Object INF 901 5045/57, Call Return.

4.10.3 Benefits

End User

The end user will get a service which allows him to get information about the last missed call and is offered an easy to use mechanism for calling back to that person.

Operator

The operator will benefit from call return in increased traffic and therefore higher ARPU. In many markets this is a legacy service and the operator will therefore be able to offer his legacy features also in a MMTel/Broadband VoIP scenario.



4.10.4 Description

The Call Return service saves information about the last incoming call and provides the end user with the possibility to obtain this information using a Supplementary Service Code (SSC). The user is also offered the possibility to make a call back to the user of the incoming call.

The Call Return service can be configured to be triggered by either all calls or by unanswered calls only.

The saved last incoming call will be kept by MTAS until either a new incoming call is received or the user is de-registered. During this time the user can use the CR service to make a call back to the caller at any time. The information about the last incoming call is kept in MTAS and is not stored in HSS for characteristics reason. We would like to avoid writing towards HSS for every call.

If there is no last incoming call or the last incoming call includes a privacy header, the served user will be informed accordingly with an announcement and the call return will fail.

4.10.4.1 Standards

Call Return is not a standardised service in IMS/MMTel.

4.10.5 Enhancement

Enhancement in MTAS 13A FD1: The announcement of the last incoming call includes besides calling number also time and date for the last incoming call.



4.11 Calling Party Category

Feature Identity:	FAJ 131 0563/1 R1A, Rev. B
Feature Type:	Optional in 11B (FD) to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

4.11.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.11.2 Summary

This feature allows the operator to assign certain Calling Party Category values for different subscribers. Certain Calling Party Categories may result in certain call treatment for certain users like for instance payphone users.

Price Object INF 901 5045/42

4.11.3 Benefits

End-user

End user could be provided with different type of subscriptions.

Operator

Operator could assign special categories to certain subscribers and give those users certain treatment. One example of a category is payphones users.

This is very useful in PSTN transformation scenarios.



4.11.4 Description

CPC is an operator service. CPC value is assigned for originating sessions in the P-Asserted-Identity. CPC value is also added in relevant diversion scenarios. The CPC value can be mapped to ISUP in PSTN interworking scenarios.

Examples of standard values for CPC are:

- ordinary
- test
- operator
- payphone
- unknown
- mobile-hplmn
- mobile-vplmn

Calling party category is also provided in charging information

4.11.4.1 Standards

3GPP TS 24.229 R8



4.12 CAPv2 SSF

Feature Identity:	FAJ 131 0688/1 R1A, Rev. B
Feature Type:	Optional in 14B (FD) to 16B (FD)
Technology:	GSM, WCDMA

4.12.1 Attention

Commercial attention

Not applicable

Dependencies

The following feature is always required:

- MTAS: FAJ1310592 - GSM Compatible SSF

4.12.2 Summary

CAPv2 SSF provides support for the CAPv2 protocol on top of the GSM Compatible SSF feature. This makes it possible to support legacy IN services already available in existing service layer when introducing Voice over LTE with MMTel.

Note that in the longer term, Camel and IN services will not have a role in IMS and will be replaced by service logic residing in IMS or on modern service platforms on top of Parlay-X northbound interface.

Price Object INF 901 5045/64

4.12.3 Benefits

End User:

This is an operator feature.

Operator:

Makes it possible to make use of legacy IN SCP's supporting the CAPv2 protocol.

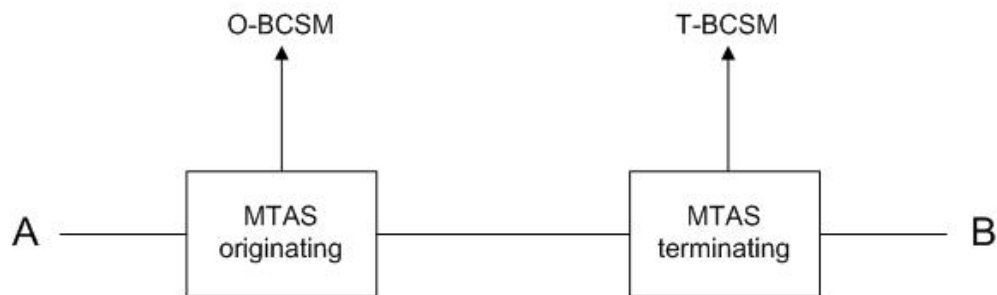


4.12.4

Description

The CAP v2 SSF function MTAS makes it possible for an CAPv2 capable Service Control Function, SCF, in the Service Layer to influence an MMTel call by sending CAP operations to MTAS. The support is based on 3GPP TS 09.78 v7.1.0, Customised Applications for Mobile network Enhanced Logic (CAMEL); CAMEL Application Part (CAP) specification.

CAMEL services will only be invoked for calls between Tel URIs and/or Tel URIs in SIP URIs with user=phone and location information will be provided only by originating MTAS. MTAS supports all Trigger Detection Points in the Originating Basic Call State Machine, O-BCSM, as well as in the Terminating-BCSM. A Basic Call will have two CAMEL service invocation points. One in the originating MTAS, and one in the terminating MTAS as shown in the below figure:



Also Diverted and follow on calls are supported.

A basic call can be enhanced by invoking a Service Layer application in an originating MTAS as well as in a terminating MTAS.

Some examples of possible use cases possible with CAPv2 are:

- CAMEL enhanced call, a Basic Call that is enhanced by CAP v2.
- CAMEL prepaid charging, it gives the CAMEL service control over the call duration.
- CAMEL post-paid charging, a CAMEL controlled call that is subject to CAMEL post-paid charging.
- CAMEL impacted MMTel charging, a call where the CAMEL service impacts offline charging in the originating MTAS



- CAMEL user interaction - announcement, a user interaction initiated by CAMEL where an announcement is played to the user on an early SIP dialogue.
- CAMEL user interaction - digit collection, a user interaction initiated by CAMEL where an announcement is played to the user and then the user provides DTMF digits that are collected by the MRFP.

Please note that the above uses-Cases listed below are examples of CAP v2 usage. MTAS supports the complete CAP v2 interface including all Camel detection points. The IN triggers are situated in the same position in the supplementary service chain as in existing MSC.

4.12.4.1 Standards

3GPP TS 09.78 v7.1.0



4.13 Carrier Pre-Select

Feature Identity:	FAJ 131 0426/1 R5A, Rev. A
Feature Type:	Optional in 11B (FD) to 16B (FD)
Technology:	

4.13.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

CSCF:

FAJ 131 0353 Carrier Routing

HSS:

FAJ 901 486 Application Data Repository

Terminal impacts and dependencies

Not Specified

4.13.2 Summary

The MTAS provides Carrier Pre-Select service which allows the operator to set up data on the end-user's behalf to override the default carrier and select a carrier based on call type, where call type is determined by the destination phone number.

Price Object INF 901 5045/13, Carrier Select / Pre-select.



4.13.3 Benefits

End-user

The end user can be served by a different operator than the default operator owning the network resources.

Operator

In many markets it is a regulatory requirement to be able to offer the end user other service providers/operators than the default service provider/operator.

4.13.4 Description

The MTAS provides Carrier Pre-Select service based on the cic and dai parameters and a Carrier Pre-Select Rn based service based on the rn parameter. These services allows the operator to set up data on the end-user's behalf to override the default carrier and select a carrier based on call type, where call type is determined by the destination phone number.

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.

Carrier Pre-Select cannot be unlocked if Carrier Pre-Select Rn is unlocked and vice versa.

Carrier Pre-Select

The 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 CPS 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>.

Carrier Pre-Select Rn

The MTAS uses the Country Code (CC) and Area Code (AC) stored in the subscription data to determine if the Called Party numbers are local or remote. If a match is found for the CC and AC in both the caller's and Called Party numbers, then the call is considered local, otherwise it is considered remote.

Congestion control is applied to Carrier Pre-Select Rn communication on receipt of a congestion error message. This is known as Crank back and gives the possibility to route calls to via other route in case of network congestion.

4.13.4.1 Standards

RFC 3261, RFC 4694, IETF draft-yu-tel-dai-05.txt endorsed by 3GPP R8.

4.13.5 Enhancement

Introduction of congestion control known as Crank back.



4.14 Carrier Select

Feature Identity: FAJ 131 0425/1 R3A, Rev. A

Feature Type: Optional in 12A to 16B (FD)

Technology:

4.14.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

CSCF:

FAJ 131 0353 Carrier Routing

HSS:

FAJ 901 486 Application Data Repository

Terminal impacts and dependencies

Not Specified

4.14.2 Summary

The MTAS provides a Carrier Select service, which allows an end-user to override the default carrier and signal which carrier to use for a particular call.

Price Object INF 901 5045/13, Carrier Select / Pre-select.

4.14.3 Benefits

End-user



The end user can be served by a different operator than the default operator owning the network resources.

Operator

In many markets it is a regulatory requirement to be able to offer the end user other service providers/operators than the default service provider/operator.

4.14.4 Description

The MTAS provides a Carrier Select service based on the cic and dai parameters, and also a Carrier Select Rn service solution based on the rn parameter. These services allow an end-user to override the default carrier and signal which carrier to use for a particular call.

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.

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:

- Release call with announcement
 - Reject call without announcement



- 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 with announcement
 - Release call without announcement
 - 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.

Carrier Select Rn

The Carrier Select Rn service allows an end-user to choose which carrier to select for a particular call. Carrier Select Rn overrides Carrier Pre-Select Rn.

The main function of the Carrier Select Rn service is to:

- Check that a user is allowed to use a carrier identified from a dialed Carrier Select Code (CSC) prefixed to the dialed number
 - Take the appropriate action which includes the addition of the rn parameter.

When enabled, Call-By-Call Block (also known as Call-By-Call Lock) prevents a subscriber from using call-by-call Carrier Select Rn. It is realized in MTAS as a subscriber who has not been provisioned with Carrier Select Rn.

Congestion control is applied to Carrier Pre-Select Rn communication on receipt of a congestion error message. This is known as Crank back and gives the possibility to route calls to via other route in case of network congestion.

4.14.4.1 Standards

RFC 3261, RFC 4694, IETF draft-yu-tel-dai-05.txt endorsed by 3GPP R8.

4.14.5 Enhancement

Introduction of congestion control known as Crank back.



4.15 Closed User Group

Feature Identity:	FAJ 131 0668/1 R1A, Rev. B
Feature Type:	Optional in 14B (FD) to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

4.15.1 Attention

Commercial attention

Not applicable

Dependencies

MGC support for CUG parameter mapping from SIP to ISUP and ISUP to SIP is needed. CUG parameter also needs to be transported to another IP network (SIP NNI).

4.15.2 Summary

Closed User Group (CUG) service is used to restrict the type of calls received by an IMS user. It can be used for instance in CallCenter scenarios.

Price Object INF 901 5045/65 Closed User Group

4.15.3 Benefits

End-user

This is an operator feature mainly used for call-center applications.

Operator

Makes it possible for an operator to provide a legacy CUG service also for IMS connected users.



4.15.4 Description

Closed User Group (CUG) service is used to restrict the type of calls received by an IMS user. It can be used for instance in CallCenter scenarios.

The service provided by MTAS enables the operator to provision groups of IMS users, whose communication profile for incoming communications is restricted to calls associated with closed user groups only. IMS users can be provisioned with one Closed User group (CUG) identity.

Terminating calls is accepted only when the CUG parameter is present and valid (same CUG as the terminating user has). CUG is supported on terminating side in MTAS and CUG parameter is included in forwarded calls (including CDIV to PSTN).

Call rejected by MTAS due to wrong CUG group ID includes a specific Cause-Code or SIP-Reason-Text for the rejected CUG calls.

For unsuccessful call attempts due to a missing or wrong CUG, the ACR can include an explicit indication about this.

4.15.4.1 Standards

3GPP TS 24.654 V10.1



4.16 Communication Completion Services

Feature Identity: FAJ 131 0497/1 R5A, Rev. A

Feature Type: Optional in 16B (FD)

Technology: IMS

4.16.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

MGC:

FAJ 131 0475 ISUP and SIP Interworking: Call Completion on Busy Supplementary Service

FAJ 131 0504 ISUP and SIP Interworking: Call Completion on No Reply Supplementary Service

Terminal impacts and dependencies

Not Specified

4.16.2 Summary

Communication Completion (CC) allows a caller who has attempted to make a call to a subscriber who is not available, to activate a Communication Completion (CC) request against that subscriber. Communication Completion comprises two services:

- Communication Completion Busy Subscriber (CCBS), which initiates a CC Call after a busy CCxx Called Party becomes free.



- Communication Completion by No Reply (CCNR), which initiates a CC Call after a call is next ended against a CCxx Called Party who did not reply to the original call attempt.
- Communication Completion on Not Logged-in (CCNL), which initiates a CC call after an unregistered Called Party becomes registered with the IMS network.

Price Object INF 901 5045/15, Communication Completion Services.

4.16.3 Benefits

End-user

The Communication Completion service is a very valuable service to the end-user. The frustration of manually having to reinitiate new call attempts several times due to busy condition or no reply or not logged in can be avoided.

Operator

Depending on the pricing policy, the Communication Completion service can generate revenue for the operator in two ways: through activation or subscription fees; and by increasing the number of completed calls.

4.16.4 Description

The Communication Completion service is divided into two main functions.

1) Communication Completion Agent

The CC Agent function is located at the originating MTAS 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.

2) Communication Completion Monitor

The CC Monitor function is located at the terminating MTAS 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 User / Group CAC service to determine when the CCxx Called Party becomes available to receive a CC Call. When the CCxx Called Party or other group user ends a call, the CC Monitor is triggered to check the current user availability and send a ready notification to the CC Agent.

For CCNL the CC Monitor monitors all successful registrations to determine when the called party becomes available to receive a CC call. When a user registers that has at least one active CCNL request in the CC B-queue, the CC Monitor is triggered to send a ready notification to the CC Agent.

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

During a CC Recall, the CC Agent can determine that the CCxx Caller is busy either through the User / Group 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 revoked.

When a call ends against a monitored user, 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 (assuming no outgoing call is made), all other terminating calls to that user are rejected with a busy response.

When a call in a group is ended making the group available again, terminating call attempts to any user of the group which does not have a maturing CC request (i.e. the idle guard timer is not running and notification has not been sent against the called user), are allowed to continue and are not rejected.

4.16.4.1 Standards

3GPP TS 24.642, RFC 3265, RFC3903, RFC3863

4.16.5 Enhancement

Enhancements in MTAS 12B:

- It is possible to re-trigger CCBS/CCNR in a CC call.
- The CC queue size limit can also be set per subscriber, not only on node.



- Some alignments to standards.

Enhancement in 13A FD1:

- Support for CCNL introduced
- Possible to play an announcement to A subscriber after unsuccessful initiation of Call Completion Call.
-

Enhancement in MTAS 16B:

Standard alignment with explicit release of early dialogs (SIP 199) for failed or non completed CCNR invocation opportunity.



4.17 Communication Deflection

Feature Identity:	FAJ 131 0175/1 R2A, Rev. B
Feature Type:	Optional in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

4.17.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.17.2 Summary

End-user can forward his/her current incoming communication to a new destination on a case-by-case basis.

Price Object INF 901 5045/2, MMTel Extended

4.17.3 Benefits

End-user

This service can be used advantageously in different situations (e.g. currently the user is not able to accept the call). By forwarding his/her actual incoming communication to another destination the end-user will not miss any important calls.

Operator

The value added forwarding service can be offered to subscribers.

It will differentiate the operator from the competitors.

The service supports new traffic cases which can increase the chargeable traffic.



4.17.4

Description

Communication Deflection (CD) is a flavor of Communication Forwarding service which does not require any rules but always forwards the communication when a 'deflection' redirect response is received from the served user. The response that is interpreted as 'deflection' is configurable on node-level by the operator.

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 Forwarding and Communication Deflection is that Forwarding 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.

It means that at an incoming call, the served user (B party) can decide whether to answer the call or to transfer it without answering it, and this is done by dialing a code. A typical use case could be a person that needs to transfer a particular call to the assistant.

At Communication Deflection, in addition to the SIP response and communication forwarding, the following actions take place:

- MTAS notifies the caller by playing an announcement
 - MTAS reveals the "Diverted-to" user's identity to the Caller
 - MTAS reveals the Served user's Identity to the "Diverted-to" party

In addition to the general configuration options described in "Communication Forwarding - Unconditional" feature the operator can set the Communication Deflection behavior on node-level via LDAP.

The following configuration options are supported:

- Communication Deflection enable/disable

Action at INVITE method responses 302

The parameter is used to determine if the INVITE method responses 302 should cause a redirect generated by the MTAS or if the 302 should be passed along towards the UA.

Please refer to Communication Diversion (CDIV) Service included under 3.7 Communication Forwarding - Unconditional for a general description of the rule based mechanism which is common for all CDIV features.



4.17.4.1 Standards

3GPP TS 24.604 V8.4.0, 3GPP TS 24.628 V8.2.0, OMA OMA-TS-XDM_Core-V1_0, RFC 4244

4.17.5 Enhancement

The Communication Diversion (CDIV) supplementary services are aligned with TISPAN R2 and have been updated for additional licenses, hop-by-hop Ack, inclusion of Communication Forwarding Not Reachable (CFNRc), interaction with Call Admission Control (CAC) and Online Charging, addition of From-Header and standards changes for History-Info.



4.18 Communication Diversion Rule Based

Feature Identity: FAJ 131 0586/1 R1A, Rev. A

Feature Type: Optional in 12A to 16B (FD)

Technology:

4.18.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.18.2 Summary

End-user can forward his/her incoming communication based on the Caller's preferences by means of a set of rules allowing the operator/end user to program the service

This feature in MTAS 12 consolidates multiple features used in MTAS 11B. In MTAS 12A no new functionality is added so the change is more 'editorial'.

Price Object INF 901 5045/2, MMTel Extended

4.18.3 Benefits

End-user

This service can be used advantageously in different situations (e.g. when traveling, during meetings, etc.). By forwarding his/her incoming communication based on customized detailed criterias, the end-user will not miss any important calls.

Operator

The value added forwarding service can be offered to subscribers.

It will differentiate the operator from the competitors.



The service supports new traffic cases which can increase the chargeable traffic.

4.18.4 Description

In general, the Communication Diversion service in MTAS is rule-based, i.e. the end-user can create rules with different conditions and the rules are evaluated to test if their respective condition(s) are true. It is possible to combine multiple conditions into one rule and one user may have more than one rule. Rules are evaluated in order. It is possible to combine rules included in this feature with the CDIV rules provided in the basic package.

Please refer to section *Communication Diversion (CDIV) Service* included in Communication Forwarding - Unconditional for a general description of the rule based mechanism.

Communication Diversion Rule Based includes the following conditions:

- Origination
- Time
- Media
- Not Logged In
- Not Reachable
- Presence

The Communication Diversion (CDIV) supplementary services are aligned with TISPA R2.

4.18.4.1 Standards

3GPP TS 24.604 V8.4.0, 3GPP TS 24.628 V8.2.0, OMA OMA-TS-XDM_Core-V1_0, RFC 4244



4.19 Communication Name Identity Presentation

Feature Identity: FAJ 131 0427/1 R3A, Rev. A

Feature Type: Optional in 15A to 16B (FD)

Technology: Fixed Broadband, IMS

4.19.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.19.2 Summary

This sub-function makes it possible for a Terminating User to see the display name associated with the identity of the Originating User as provided by an external Calling Name Server. If CNIP function is requested then the display-name portion of the P-Asserted-Identity and the From-Headers may be changed. The service exists in two modes, "always" and "interrogate-on-unavailability". In "interrogate-on-unavailability" the display name is only retrieved from the external database if the display name is missing from the identity provided in the SIP request from the Originating User in both the P-Asserted-Identity and the From headers. In "always" mode, the display name is always retrieved and the display name in the request is overwritten if it is already present on the Originating User's request.

Price Object INF 901 5045/11, Communication Name Identity Presentation.

4.19.3 Benefits

End-user

This feature makes it possible for the terminating user to see the name identity of the originating user at the initiation of a communication. Users find it useful to know the name of who is calling (especially if the originating number is not recognized) or to be able to ignore the call, note the name identity and call back later.



Operator

This feature can generate additional revenues for an operator through subscription to the service.

4.19.4

Description

The CNIP function is executed on behalf of the terminating user and in the terminating MTAS. The main case for CNIP is that the terminating user has OIP and CNIP and then MTAS will retrieve the calling name from the external Calling Name Server and update the identity information in the SIP-messages. If the terminating user does not have both OIP and CNIP or if the calling name could not be retrieved from the external database (for any reason) no modification of the message is performed by the CNIP function. The CNIP function has two modes of operation: "always" and "interrogate-on-unavailability". In "interrogate-on-unavailability" mode the CNIP function will only retrieve the calling name if there is no display-name portion present in any of the headers (P-Asserted-Identity- or From-header).

If the served user has OIP and CNIP enabled then:

- If OIP anonymizes the From header, and/or deletes the P-Asserted-Identity header then CNIP is not executed.
 - The CNIP function will check if there is a display-name already in at least one of the headers (P-Asserted-Identity or From) if there is and the CNIP mode is set to "interrogate-on-unavailability" no headers will be modified by the CNIP service.
 - If the request contains a global number in P-Asserted-Identity or From headers that global number will be used as parameter in the query to the external Calling Name Server. If a global number is present in both headers P-Asserted-Identity value takes precedence. If both sip and tel P-Asserted-Identity headers are present P-Asserted-Identity tel-format takes precedence. If there is no global number in any of the headers no query will be sent to external Calling Name Server and no name is retrieved nor replaced.
 - The global number of the called user is obtained from the list of identities present in the IRS data. If several global numbers are available, the first one in the list will be used as a parameter in the query to the external Calling Name Server. If it's not available it will not be included in the query to CNS.



- The CNIP function invokes the NameDb interface subsystem providing both the global number identity of the called user and the calling user to perform a query to the Calling Name Server. If the calling user's global number identity is not available then no request is made to the NameDb interface and the CNIP function makes no further attempt to modify the message.
- If the NameDb subsystem returns a name string, the first 64 characters of that string are written to the display-name in P-Asserted-Identity and From headers in quoted-form and following the escaping rules described in RFC 3261. If the headers already have a display-name that display-name gets overwritten. If the Name subsystem request fails for any reason then the CNIP function makes no further attempt to modify the message.

The service behavior is determined when the first message from the originating user is received and this behavior will be kept for the duration of the session. This means that the same header will be updated for each message.

There is one common administrative state of the Identity Presentation function in MTAS. The operator can enable/disable all CNIP on node-level. This node-level configuration is performed by the operator via LDAP.

MTAS provides the end-user the following configuration option:

- CNIP Activation (activate/deactivate)

The end-user can configure his/her setting via the Ut-interface.

MTAS provides a number of performance counters to evaluate the usage and quality of service. These counters are:

This counter represents the number of successful invocations of the CNIP service when the action is to not restrict the identity.

This counter represents number of invocations of the CNIP service where no entry could be found in the external database for the given calling id. The external database can be accessed over either a SOAP or SIP interface.

The name can also be provisioned in the subscriber profile.

4.19.4.1 Standards

3GPP TS 24.607, IETF RFC 3323, IETF RFC 3325



4.19.5 Enhancement

Enhancement in MTAS 13A:

It is optional to either provision the name in the subscriber profile or possible to use external database for getting more information for terminating calls. There is also a new interface option towards the external name database. Now it is possible to access the database using either SOAP or SIP interface.

Enhancement in MTAS 15A:

Retrieving additional configurable identity information from the calling party.



4.20 Customized Alerting Tone

Feature Identity: FAJ 131 0576/1 R1A, Rev. A

Feature Type: Optional in 12A to 16B (FD)

Technology:

4.20.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.20.2 Summary

Customized Alerting Tone is a commonly used service in mobile networks. It is sometimes called Personal Ringback Tone but the name used in 3GPP standards is CAT (Customized Alerting Tone).

The service allows an end user to define customized alerting tones.

Price Object INF 901 5045/51, Customized Alerting Tone.

4.20.3 Benefits

End-user

This is a popular feature by end users with high commercial value. End users are able to choose alerting tones to their liking.

Operator

Operators can continue to offer a popular end user service also when introducing MMTel and VoLTE..



4.20.4 Description

MTAS implements the service invocation part of the Customized Alerting Tone service. It is assumed that a CAT server (media server), which in many cases already exist in the network, contains existing end users media files. So provisioning of media files and content management is handled by the external CAT server. MTAS interfaces the CAT server using the Mr-interface (SIP). The Mp-interface towards the CAT server is not supported but a scenario where Mp-interface is used for regular media handling (multiparty calls and announcements) while the CAT service uses Mr-interface is supported.

It is assumed that the CAT server contains all the rules used to decide which alerting tone should be sent and therefore can this logic be reused from an existing legacy node. MTAS will generate charging information in case the service was used.

4.20.4.1 Standards

3GPP TS 22.182, 3GPP TS 24.182



4.21 Dynamic Black List

Feature Identity:	FAJ 131 0494/1 R3A, Rev. A
Feature Type:	Optional in 14B (FD) to 16B (FD)
Technology:	

4.21.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.21.2 Summary

Dynamic Black List (DBL) enables an end-user to, during or after the call, store incoming numbers in a black list to avoid unwanted communication. DBL is also applicable in case of anonymous communication

Price Object INF 901 5045/18, Dynamic Black List.

4.21.3 Benefits

End-user

The end-user can bar unwanted anonymous communication in a very selective way, without having to rely on the ACR feature for barring of all anonymous calls.

Operator

DBL is a value adding barring service that can be offered to subscribers which will differentiate the operator from the competitors.

Selective barring of anonymous communication will allow wanted anonymous communication to be set-up, thus securing chargeable traffic in the network.



4.21.4 Description

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

An end-user having the service active will be able to store an unwanted incoming number in 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 will bar all anonymous calls. When an anonymous call is stored in the black list, the number will remain invisible to the end-user, thereby guaranteeing the anonymity of the calling user.

4.21.5 Enhancement

New in MTAS 14A is the possibility to charge for the announcement towards the caller who is barred by the service. This is achieved with an announcement instead of early media.



4.22 Explicit Communication Transfer

Feature Identity: FAJ 131 0562/1 R2A, Rev. B

Feature Type: Optional in 12A to 16B (FD)

Technology:

4.22.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

MGC:

FAJ 131 0443 ISUP and SIP Interworking: ECT Supplementary Service

Terminal impacts and dependencies

Not Specified

4.22.2 Summary

The consultative Explicit Communication Transfer allows an end-user to transfer ongoing communication sessions to another destination after consultation with the subscriber that receives the session.

ECT consultative is supported in MTAS 11B release.

Price Object INF 901 5045/41



4.22.3 Benefits

End-user

End user will be able to transfer important communication sessions to the most appropriate receiver.

Operator

Existing PSTN/ISDN feature ECT can still be offered in PSTN transformation scenarios. The call duration for sessions can increase. Both call legs can be charged separately.

4.22.4 Description

The end user put a conversation on hold, calls another subscriber and when that session is established he/she can connect the two existing sessions and leave the conversation.

4.22.4.1 Standards

3GPP TS 24.629 version 8.3.0

4.22.5 Enhancement

ECT has been enhanced with the possibility for the B-party to check and terminate an ongoing transferred call.



4.23 File Sharing

Feature Identity:	FAJ 131 0168/1 R1A, Rev. B
Feature Type:	Optional in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

4.23.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

FAJ 131 0169 Add/Drop Media

Other node impacts and dependencies

SBG:

FAJ 131 0210 TCP-based media, RFC 4145
FAJ 131 0223 MSRP end-to-end interworking

MRS:

FAJ 121 2412 Border Gateway Function

Terminal impacts and dependencies

Not Specified

4.23.2 Summary

MTAS supports the handling of file sharing communication within MMTel between two end-users.

Price Object INF 901 5045/9, Multimedia Communication



4.23.3 Benefits

End-user

The feature enables an end-user to share different types of media e.g. pictures, movie clips, audio clips, etc. which will enhance his/her communication experience.

Operator

This new type of communication generates additional revenue sources.

By offering file sharing services, the operator can attract new subscriptions and decrease the subscriber churn rate.

4.23.4 Description

The sharing service involves the creation of a MSRP (Message Session Relay Protocol) session between two 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, etc. It can also be used to send text messages on a line by line basis.

For sharing, the MSRP session will typically be set up in parallel with an audio session. Supported media types are Image; Voice; Video and Text.

4.23.4.1 Standards

3GPP TS 22.340 V6.2.0 - IMS Messaging



4.24 Flexible AVP

Feature Identity:	FAJ 131 0636/1 R1A, Rev. B
Feature Type:	Optional in 13A to 16B (FD)
Technology:	LTE, Fixed Broadband, Wireless Broadband, IMS

4.24.1 Attention

Commercial attention

Not applicable

Dependencies

The following features are always required:

- MTAS: FAJ1310164 - Offline Charging
- MTAS: FAJ1310499 - Online Charging

4.24.2 Summary

Flexible AVP allows the operator to configure arbitrary SIP header field names that will be reported on either the Ro or Rf interface.

Price Object INF 901 5045/60, Flexible AVP.

4.24.3 Benefits

End User

N/A. This is an operator service

Operator

The operator is able to use arbitrary SIP header field names and report this on charging. This gives a high degree of flexibility in the charging area.



4.24.4

Description

The Flexible AVP function allows the operator to define SIP message header fields to be reported in AVPs of charging requests. If the configured SIP header field is found in the message, the Transaction-Info AVP will be used to report the content of the matching SIP header.



4.25 Flexible Communication Distribution

Feature Identity: FAJ 131 0492/1 R7A, Rev. B

Feature Type: Optional in 16A to 16B (FD)

Technology: IMS

4.25.1 Attention

Commercial attention

Not applicable

Dependencies

The following node features are required in order to use IMS system level feature Service Convergence. For more information, see IMS Feature Description.

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

FAJ 131 0544/1 R1A AS Controlled Forking

Other node impacts and dependencies

CSCF:

FAJ 131 0554/1 R2A Service Convergence

The following node features are optional for IMS system level feature Service Convergence

SBG:

FAJ 131 0453 SIP Message Manipulation & Protocol repair

Terminal impacts and dependencies

Not Specified



4.25.2 Summary

With Flexible Communication Distribution (FCD) it is possible to define call distribution rules towards different telephone numbers and devices based on parallel and/or sequential ringing. Both fixed and mobile numbers and devices can be used forming a one number service.

Price Object INF 901 5045/22, Flexible Communication Distribution.

4.25.3 Benefits

End-user

FCD enables a one number service, thereby increasing the flexibility and convenience for the user as he/she can decide how to handle incoming traffic over multiple devices.

The feature also enables the use of multiple phone numbers.

Operator

The feature can be offered to subscribers.

It will differentiate the operator from the competitors.

The feature increases the likelihood for successful communication resulting in increased chargeable traffic in the network.

4.25.4 Description

The Flexible Communication Distribution (FCD) service is an IMS terminating service with a serial and parallel mode to ring the targets in a serial or parallel mode. The served user can set rules and update the service from both clients and from web portal via the Ut interface. The FCD service shares rule set with other rule based services like CDIV and Barring.

These rules may be defined for both parallel and serial modes of ringing including combinations. The rules may have day and time defined, controlling the day and time when the rules are applied. The served user may also define the period of ring for the targets for the parallel and serial modes.

For a user provisioned with the FCD, an INVITE received on the unregistered terminating port is treated by the terminating Unregistered service as if the user were registered. If alerting the non-IMS served user's primary target fails, the Terminating Unregistered service treats the INVITE as if the user were unregistered.



An announcement is played to the caller whilst the FCD communication establishment is being carried out by MTAS.

The ring period for both parallel and serial modes is defined by the ring-period-type attribute in the FCD service data. The ring-period-type range is 5 to 120 seconds. For the parallel case the ring period is applicable to all served targets. For the serial case the ring period is applicable to individual served targets, and may be different for each target.

Flexible Communication distribution divert primary (FCDDP) and Communication Deflection from Primary (CDP) service are supported.

FCD Divert Primary (FCDDP) service unconditionally diverts the incoming call being distributed to the IMS Primary User to a configurable alternative target.

Communication Deflection from Primary (CDP) service processes the SIP 302 (Moved Temporarily) response, received from IMS Primary User, by deflecting the incoming communication distributed towards IMS Primary User and sending new INVITE to the URI target indicated in the Contact header.

FCD avoid answers from automatic answering machines by procedures demanding manual intervention.

Any user can include internal voicemail as one of FCD targets.

FCD has configurable reason headers to improve integration with UE and a better end user experience.

4.25.4.1 Standards

3GPP TS 29.163 V8.8.0, 3GPP TS 24.604 V8.4.0, ETSI TS 183 023, ETSI TS 24.243, RFC 4244, RFC 4745, RFC 4458, RFC 3261, RFC 4745, RFC 3966, <http://www.w3.org/TR/xmlschema-2>, OMA OMA-TS-XDM_Core-V1_0

4.25.5 Enhancement

FCD in MTAS 15B has introduced two enhancements. In the use case where a call has been treated with parallel ringing and the call is answered by one of the devices then the FCD service supports sending SIP CANCEL to all other device with a configurable reason header text. Default value is "Call completed elsewhere" which makes it possible for a device to have a correct local call log.

The second improvement is also related to parallel ringing. When a 486 Busy here is received from one UE then MTAS can respond with a Cancel with configurable response header. Default is "Busy everywhere. This behavior facilitates interworking with certain terminals and CS networks.



In MTAS 16A FCD is improved with a configuration option to allow only one secondary device to be involved in calls and to limit the number of calls on the secondary device. There is also an improvement to avoid voice clipping issues.



4.26 Flexible Service Format Selection

Feature Identity: FAJ 131 0579/1 R2A, Rev. B

Feature Type: Optional in 12B (FD) to 16B (FD)

Technology:

4.26.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.26.2 Summary

The Flexible Service Format Selection (FSFS) service makes it possible for other application servers before MTAS in the ISC chain to influence the set of services to be executed in MTAS for a specific session.

Price Object INF 901 5045/54, Flexible Service Format Selection

4.26.3 Benefits

End-user

End users will benefit from this feature as it makes it possible to extend the feature set beyond what MTAS supports and still avoid undesired feature interactions between services provided by MTAS and an applications server which is linked in before MTAS is the ISC chain.

Operator

Operators will benefit from this feature as it makes it possible to extend the feature set beyond what MTAS supports and still avoid undesired feature interactions between services provided by MTAS and an applications server which is linked in before MTAS is the ISC chain.



4.26.4 Description

The Flexible Service Format Selection service suppresses some of the existing MTAS services in a dynamic way by means of check of parameters in the incoming INVITE for each new session to decide what service is not needed in MTAS for the session.

A concrete example of usage could be a third party application server implementing a 'wakeup call service' which then sets a specific feature tag to identify that the call is a wakeup call. The terminating MTAS would then recognise the feature tag and omit ICB and CDIV services for this particular call. It would otherwise not make any sense to bar or forward to voicemail a wakeup call.

4.26.4.1 Standards

Prestandard

4.26.5 Enhancement

Enhancement in MTAS12B:

The service is enhanced to be able to suppresses both terminating and originating services in certain emergency related scenarios.

Suppression on terminating side includes 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.

For originating services, the FSFS service is enhanced to be able to suppress 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.

These enhancements are useful in certain emergency call and emergency callback scenarios.



4.27 Gateway Model

Feature Identity:	FAJ 131 0491/1 R2A, Rev. A
Feature Type:	Optional in 15A to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

4.27.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.27.2 Summary

The Gateway Model feature introduces a SIP signalling mechanism to secure interwork with devices or network only capable of handling single SIP dialogs.

Price Object INF 901 5045/23, Gateway Model.

4.27.3 Benefits

End-user

An end-user will to a larger extent be able to keep his/hers old SIP terminal.

Operator

Interconnect enabled with networks such as Telephony Softswitch (TSS), only capable of handling one single SIP dialog.



4.27.4 Description

The Gateway Model (GM) service allows MTAS to map events received from terminating user - on one or more dialogs - towards the originating user on one single dialog. The service can be used when the user cannot distinguish between early media (media before final answer for the initial INVITE) and regular media (media after final answer sent for the initial INVITE).

As there is only one dialog between Originating User 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.

4.27.4.1 Standards

3GPP TS 24.182 V9.2.0, RFC 3960

4.27.5 Enhancement

MTAS 15A introduce the possibility to invoke the Gateway Model per session, triggered by the "no-fork" directive.



4.28 GSM Compatible SSF

Feature Identity:	FAJ 131 0592/1 R1A, Rev. C
Feature Type:	Optional in 12B (FD) to 16B (FD)
Technology:	GSM

4.28.1 Attention

Commercial attention

Contact SPM

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

FAJ 131 0688/1 R1A CAPv2 SSF

Other node impacts and dependencies

Not Applicable

Terminal impacts and dependencies

Not Specified

4.28.2 Summary

GSM Compatible SSF is a base feature introducing an SSF function integrated in MTAS service logic. This makes it possible to support legacy IN services already available in existing service layer when introducing Voice over LTE with MMTel.

Note that in the longer term, Camel and IN services will not have a role in IMS and will be replaced by service logic residing in IMS or on modern service platforms on top of Parlay-X northbound interface.

In order for this feature to be useful it requires at least one additional feature for the actual IN protocol to be used.



Price Object INF 901 5045/50, GSM Compatible SSF

4.28.3 Benefits

End User:

This is an operator feature.

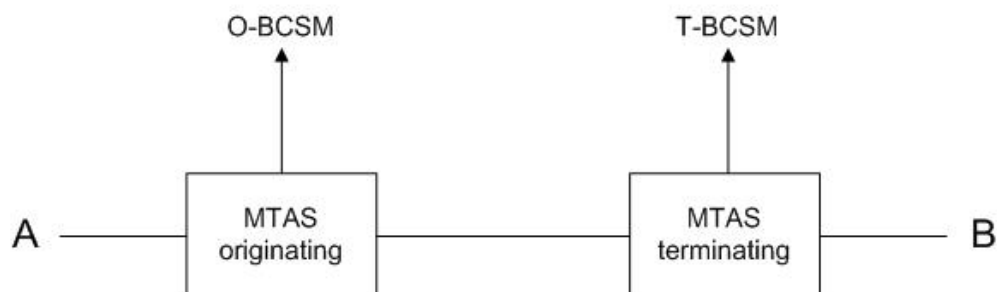
Operator:

The integrated SSF function in MTAS makes it possible to roll out VoLTE without impacting the existing service layer with IN SCPs that are widely spread in today's GSM and WCDM networks. This gives the mobile operators's core organisation the independence to upgrade the core network to support VoLTE without impacting the existing service layer that in most cases resides in a separate organisation.

4.28.4 Description

The SSF function in MTAS makes it possible for an Service Control Function, SCF, in the Service Layer to influence an MMTel call by sending CAP operations to MTAS. The support is based on 3GPP TS 09.78 v7.1.0, Customised Applications for Mobile network Enhanced Logic (CAMEL); CAMEL Application Part (CAP) specification.

CAMEL services will only be invoked for calls between Tel URIs and/or Tel URIs in SIP URIs with user=phone and location information will be provided only by originating MTAS. MTAS supports all Trigger Detection Points in the Originating Basic Call State Machine, O-BCSM, as well as in the Terminating-BCSM. A Basic Call will have two CAMEL service invocation points. One in the originating MTAS, and one in the terminating MTAS as shown in the below figure:





Also Diverted and follow on calls are supported.

A basic call can be enhanced by invoking a Service Layer application in an originating MTAS as well as in a terminating MTAS.

Some examples of possible use cases are:

- CAMEL enhanced call, a Basic Call that is enhanced by CAP.
- CAMEL prepaid charging, it gives the CAMEL service control over the call duration.
- CAMEL post-paid charging, a CAMEL controlled call that is subject to CAMEL post-paid charging.
- CAMEL impacted MMTel charging, a call where the CAMEL service impacts offline charging in the originating MTAS
- CAMEL user interaction - announcement, a user interaction initiated by CAMEL where an announcement is played to the user on an early SIP dialogue.
- CAMEL user interaction - digit collection, a user interaction initiated by CAMEL where an announcement is played to the user and then the user provides DTMF digits that are collected by the MRFP.

Please note that the above uses-Cases listed below are examples of CAP v2 usage. MTAS supports the complete CAP v2 interface including all Camel detection points. The IN triggers are situated in the same position in the supplementary service chain as in existing MSC.

In order to make this feature useful, please note that also the CAPv2 SSF feature is required.

4.28.4.1

Standards

3GPP TS 09.78 v7.1.0



4.29 Hotline

Feature Identity: FAJ 131 0634/1 R1A, Rev. B

Feature Type: Optional in 13A to 16B (FD)

Technology: LTE, Fixed Broadband, IMS

4.29.1 Attention

Commercial attention

Not applicable

Dependencies

This Feature has no dependencies.

4.29.2 Summary

Hotline is a commonly used supplementary service in legacy wireline network. It provides a mechanism to trigger a call to a predefined destination by lifting the receiver.

The MTAS implementation covers in addition to the traditional use case for Hotline also operator enforced Hotline which can be used to reroute any end user made call to a predefined number. In most cases this is used when the operator would like to reroute calls to customer care in case the end user has not paid his bill.

Price Object INF 901 5045/58, Hotline.

4.29.3 Benefits

End User

The end user can benefit from using the Hotline supplementary service. Both instant and delayed hotline is supported. Delayed hotline could be useful for example to access voice mail in a simple way and in this case the end user does not have to dial to the voice mail system to access voice mails.

Operator



The operator is able to offer both delayed and instant hotline services. Instant hotline is common for various types of courtesy phones in for example airports to call for taxi, rental car and hotels.

With the operator enforced hotline it is possible to get in contact with subscribers that have not paid their bills and at the same time prevent that these subscribers makes outgoing calls.

4.29.4 Description

Unconditional Hotline (Automatic Rerouting to Customer Care)

This service is used in case of customers that have not paid their bill. Their calls are automatically routed to hotline number when initiating a call. There is no special code in Request-URI as it is in Instant and Delayed Hotline. Each Request-URI in initial INVITE is replaced with hotline number. The implementation support the requirement language based routing where users can be routed based on their language profile. It is done in operator subscription level where operator can define hotline number base on his/her language preferences.

Instant Hotline

A call is initiated after off hook. The IMS client generates INVITE request with special code in Request-URI which allows recognizing the hotline service. In order to have a working service end to end it is therefore important to select an IMS client (in IAD or similar) that detects the off hook and is able to send a SIP INVITE with a special code that could be recognised by MTAS. This code is configurable and could for example look like "***". The special code is replaced with a preconfigured number included in the subscriber profile. Instant Hotline is an operator only configurable service.

Delayed Hotline

A call is initiated a few seconds after off hook. The timer is implemented in the user equipment (IMS client) and MTAS is not aware of it. This scenario is the same as Instant Hotline from traffic point of view. The only difference is in provisioning. The Delayed Hotline can be configured by user (Ut, SSC) and operator (CAI3G, node level configuration).



4.30 HSS Bypass

Feature Identity:	FAJ 131 0803/1 R1A, Rev. A
Feature Type:	Optional in 15B (FD) to 16B (FD)
Technology:	GSM, LTE, IMS

4.30.1 Attention

Commercial attention

Not applicable

Dependencies

The following feature is always required:

- MTAS: FAJ1310567 - Service Centralization and Continuity Application Server

4.30.2 Summary

SCC AS can be deployed without an interface to HSS(IMS) directly to HLR.

4.30.3 Benefits

Network Efficiency

4.30.4 Description

SCC AS can be deployed without an interface to HSS(IMS) meaning that the data normally fetched from HSS(IMS) by SCC AS during registration (IRS and C-MSISDN/MSISDN) is obtained from the 3PTY Registration message, and SCC AS acting as a GMSC in the T-ADS service when requesting the MSRN allocation for CS breakout directly to HLR over an ETSI MAP interface.



4.31 In Conference Control

Feature Identity:	FAJ 131 0632/1 R1A, Rev. E
Feature Type:	Optional in 14B (FD) to 16B (FD)
Technology:	LTE, Fixed Broadband, IMS

4.31.1 Attention

Commercial attention

Not applicable

Dependencies

The following feature is always required:

- MTAS: FAJ1310574 - Scheduled Conference

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

FAJ 131 0574 Scheduled Conference

Other node impacts and dependencies

Not Applicable

Terminal impacts and dependencies

Not Specified

4.31.2 Summary

"In conference control" offers a web interface for the conference owner to manage a scheduled conference.

Price Object INF 901 5045/61, In Conference Control.

4.31.3 Benefits

End User



The conference owner will be able to manage his booked conference and he is able to perform more operations on the conference than the normal user of the conference service. By doing so he is able to effectively manage the service over a web based interface significantly improving the usability of the scheduled conference service.

Operator

With the "In conference control" feature the operator is able to more effectively compete on the audio/video conference market by improving the usability.

4.31.4

Description

In conference control is an add on to scheduled conference in which a conference owner is able to create and manage his conference. The conference owner has additional rights compared to conference participants. In a moderator role he manages his conference using a web based interface called CCMP (draft-ietf-xcon-ccmp-11).

The following controls is supported for the moderator/conference creator role:

- Kick out participant(s)
- Dial-out to a first participant when the conference is not started which will automatically make the conference active
- Dial-out to other participant(s) when conference is active
- Lock/unlock conference for more participants to enter
- Mute/unmute audio of particular participant(s), one-way and both-way
- Mute/unmute video of particular participant(s), one-way and both-way
- Mute/unmute audio of all participants except moderator, one-way and both-way
- Mute/unmute video of all participants except moderator, one-way and both-way
- Terminate and delete active conference

The following controls is supported for all participants:

- Mute/unmute audio of own line, one-way
- Mute/unmute video of own line, one-way

In conference control is applicable together with off-line charging only.



4.32 Incoming Communication Barring Rule Based

Feature Identity:	FAJ 131 0573/1 R3A, Rev. A
Feature Type:	Optional in 15A to 16B (FD)
Technology:	GSM, WCDMA, LTE, Fixed Broadband, Wireless Broadband, IMS

4.32.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.32.2 Summary

This feature enables the operator and the end-user to restrict incoming communication, using an advanced, rule-based method.

Price Object INF 901 5045/2, MMTel Extended.

4.32.3 Benefits

End-user

The end-user will be able to control his/her incoming traffic by creating sophisticated barring rules.

Operator

The operator will be able to offer Incoming Barring services to its end-users to control incoming traffic from the network.

It will differentiate the operator from the competitors.



4.32.4 Description

The rule based incoming communication barring uses rule sets which includes:

- Origination
- Time
- Media

Thanks to the flexibility of the MTAS rule-based barring feature, the ICB Origination can be combined in many ways with other types of incoming barring to express whether a communication should be barred or not. In this mode, the following barring rule examples can be easily created:

- Bar all incoming real-time text communication except from *sip:+3617654321@ericsson.com;user=phone*
(In this example the conditions are based on Media + Origination)
 - Bar every type of communication from tel:+3617654321 except video
between 08.00-11.00 2009-10-14
(In this example the conditions are based on Origination + Media + Time)
 - The above rules can also be combined with the action to play a Do Not Disturb related announcement.

MTAS provides the following node-level configuration options which are managed by the operator via LDAP-interface.

- Communication Barring enable/disable
There is one common administrative state of the Communication Barring function in MTAS. The operator can enable/disable all barring sub-features on node-level (i.e. Barring Programs, OCB, ICB, ACR).
 - Audio Announcement enable/disable
The operator can enable/disable the audio announcement when a communication is barred due to ICB.
 - "Audio only" Announcement
The operator can specify which audio announcement is to be played to the originating party when a communication is barred due to ICB.



- Video Announcement enable/disable
The operator can enable/disable video announcement when a communication is barred due to ICB.
- "Video only" Announcement
The operator can specify which video announcement, without audio, is to be played to the originating party when a communication is barred due to ICB.
- Audio/Video Announcement - Audio part
The operator can specify which audio, associated with a video announcement, to be played to the originating party when a communication is barred due to ICB.
- Audio/Video Announcement - Video part
The operator can specify the video part of the audio/video announcement.

The user-level configuration is performed by the end-user via the Ut-interface and accessed by the operator via EMA using CAI3G.

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

- Number of barred incoming communications

Number of barred incoming diverted communications

4.32.4.1 Standards

3GPP TS 24.611, IETF RFC 4745, IETF RFC 3323, IETF RFC 4566, ETSI ES 183 028, IETF RFC 2396, IETF RFC 3966, <http://www.w3.org/TR/2004/REC-xmlschema-2-20041028/>

4.32.5 Enhancement

Following has been added in 15A:

Communication triggering video block barring can get automatic downgrade to audio.



4.33 Japanese Charging

Feature Identity: FAJ 131 0697/1 R3A, Rev. B

Feature Type: Optional in 16B (FD)

Technology:

4.33.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.33.2 Summary

Japanese charging is only applicable for operators offering telecommunication services in Japan. This feature covers two different functions called Interconnection Charge Billing System (ICBS) and Flexible Charging (FCH) Both these function are standardised by the Japanese standardization group Telecommunication Technology Committee (TTC).

Price Object INF 901 5045/71, Japanese Charging.

4.33.3 Benefits

End-user

The end-user will be able to get correct charging of his/her communication according to Japanese standard.

Operator

Charging and Accounting will be performed correctly and according Japanese standard.



4.33.4 Description

Interconnection Charge Billing System (ICBS)

Accounting information is sent between operators in both directions using SIP. The interconnect is always using ISUP which means that all interconnect will be over the MGC. The accounting information is stored and then sent in ACR start messages (Rf). Interconnect can take place even if the two subscribers are served by the same operator but are located in different cities.

MTAS has the possibility to determine in what cities the caller and callee is located in and based on analysis provide the necessary information to the CDF.

Flexible Charging (FCH)

Flexible charging is applicable for calls between a mobile subscriber and a subscriber served by NTT (fixed network).

FCH makes it possible for the operator of the terminating network to determine the call charge for calls, or call legs, originating in another network when call charge cannot be determined on basis of the called party number.

4.33.4.1 Standards

Telecommunications Technology Committee (TTC), Standard JJ-90.10 Inter-Carrier interface based on ISUP (ISDN User Part)

4.33.5 Enhancement

MTAS 14B has been enhanced with configuration of if the Japanese Charging information should be represented as SIP headers or SIP parameters.

In MTAS 16B this feature has been enhanced with

- support for the CFU use case at VoLTE Roaming,
- *ICBS parameter addition when triggering CFB due to busy ICS user in CS network, covering femto cell interworking*



4.34 Legal Intercept

Feature Identity: FAJ 131 0493/1 R4A, Rev. B

Feature Type: Optional in 13A to 16B (FD)

Technology: IMS

4.34.1 Attention

Commercial attention

Not applicable

Dependencies

LI-IMS is required for the output to standard interfaces. The full LI functionality is dependant on the IMS LI complete solution including other IMS products such as CSCF and SBC.

4.34.2 Summary

The Legal Intercept feature enables Lawful Enforcement Agencies (LEA) to lawfully intercept call data and call information data from the MMTel network for users when using the 3PTY, Ad-hoc conference, Scheduled Conference and ECT service.

Price Object INF 901 5045/19, Legal Intercept.

4.34.3 Benefits

Operator

Legal Intercept is in many markets a mandatory regulatory service.

4.34.4 Description

The scope with this feature is to, on a high level, describe the LI function related to the supplementary services executed within the MTAS. The scope is not to describe the LI function provided by the Ericsson based LI management solution (LI-IMS).



The LEA interfaces with an administrative function (ADMF) in the LI-IMS system when activating interceptions and delivery functions when collecting interception information.

Legal Intercept is handled by the MTAS that communicates with the ADMF. This is done over an X1 interface used for activating, deactivating and interrogation of LI targets in the MMTel network.

The MTAS monitors activated targets and sends intercepted related information, i.e. SIP messages and Conference Event information to the LI-IMS system. In parallel, a correlation number is created, one for each participant in the conference that shall be monitored. The correlation number(s) is sent as part of the IRI information. The LI information can be sent using IPsec.

The SBG-A node has support for LI and monitors activated targets on a per user level. The SBG sends intercepted related information to the LI-IMS system over the X2 and X3 interfaces. A correlation number (identical with the number created in the MTAS) is created in SBG-A which makes it possible to correlate the information delivered by MTAS with the one delivered by SBG-A.

4.34.4.1 Standards

ETSI TS 102, 232-5 V2.1.1, ETSI TS 102, 232-1 V2.1.1

4.34.5 Enhancement

New in MTAS 13A:

Support for scheduled conference



4.35 Location Based Number Translation

Feature Identity:	FAJ 131 0577/1 R5A, Rev. C
Feature Type:	Optional in 15B (FD) to 16B (FD)
Technology:	IMS

4.35.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.35.2 Summary

Location Based Number Translation is number translation service based on location intended primarily for the North America and other markets where you can dial according to local area dial plan based on location of your mobile.

Calls are routed based on location for different types of calls e.g. 7-digit dialing, 10-digit dialing, short codes, N11 and toll-free 8YY dialing.

Price Object INF 901 5045/52, Location Based Number Translation

4.35.3 Benefits

End-user

End users are able to continue using a set of commonly used legacy services when moving to VoLTE.

Operator

Operators can continue to offer a commonly used end user service also when introducing MMTel and VoLTE. Mobile operators can continue offer certain legacy services and comply with North American Numbering Plan and similar for those having local calling in their mobile dial plan.



4.35.4 Description

MTAS can determine location from information received in PANI and PAI headers. The following PANI formats are supported:

- CDMA 1x
- EVDO revA
- LTE

Based on this format MTAS is able to route calls towards certain service numbers to the correct destination using a number translation.

Several different types of service numbers are supported including:

- local number length dialing. When the end user is located in his home area code (in North America, Australia and other, wireline and wireless subscribers in the same area uses the same area code) the user does not need to make a local call using area code. The local 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. Configurable different announcements for local and long distance digit dialling. there is also Local calling area support for fixed devices.
- White list for not performing long distance check.
- Short codes. Phone numbers which are significantly shorter than regular phone numbers. The short-code can start with any number including “#” and “*” characters.
- N11. Certain service numbers.
- 8YY. 1-800 plus certain other number series. Sometimes called freephone or toll free.

4.35.4.1 Standards

3GPP TS 24.229

4.35.5 Enhancement

Enhancement in MTAS 14B:



New structure for Cell database using rating centers. Configurable announcement for 7 and 10 digit handling. White list for not performing long distance check. Local calling area support for fixed devices.

Enhancement in MTAS 15A:

Numbering plan number length independency, e g 8 digit local calling support.

Enhancement in MTAS 15B:

- Wildcarding in translation tables.
- Location based barring to certain destinations, e g for fraud prevention.
- Global Short Codes groups, covering several levels.



4.36 Mr-interface

Feature Identity: FAJ 131 0543/1 R2A, Rev. B

Feature Type: Optional in 13A to 16B (FD)

Technology: IMS

4.36.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.36.2 Summary

The Mr reference point is defined by 3GPP standard and it is the interface between an application server and an external MRFC. This interface is used for media resource handling e.g announcements and multiparty calls.

Price Object INF 901 5045/31, Mr-interface

4.36.3 Benefits

End-user

End users will benefit from a wide range of media resource related services like announcements.

Operator

Operators can use common MRFC nodes handling media resources not only for MMTel but also for other services. With the Mr-interface MTAS is able to interwork with MRFCs that are common to a large range of applications apart from MMTel. By using a MRFC which is common to several different applications the operator have the possibility of reducing OPEX.



4.36.4 Description

MTAS supports the Mr reference point towards an external MRFC. The protocol used for this interface is Netann (RFC 4240). In MTAS 11A the Mr interface supports all the features that is supported using the Mp interface (MRFC integrated in MTAS with a H.248 protocol towards a MRFP). This means that the Mr-interface supports announcements and services HOLD, Conference, 3PTY calls and CCBS. In case of CCBS MTAS supports a limited IVR capability. The IVR (prompt and collect) is implemented using voice XML (VXML).

Load balancing is supported in such a way that MTAS sends requests to the MRFC based on a configurable generic MRFC hostname. Then it is up to the DNS to resolve it to an IP address of a certain MRFC node. In some cases - like conferencing - subsequent requests have to be sent to the same MRFC. In these cases MTAS uses the IP address from the first MRFC response rather than the configured generic MRFC hostname.

It is possible to select MRFC located in different regions. The selection mechanism is based on SIP P-Access-Network-Info header added by SBG to INVITE. MTAS is able to select a MRFC associated with (presumed to be located at same site or close to) the SBG handling the call.

4.36.5 Enhancement

Enhanced in MTAS 13A:

It is possible to select MRFC located in different regions. The selection mechanism is based on SIP P-Access-Network-Info header added by SBG to INVITE. MTAS is able to select a MRFC associated with (presumed to be located at same site or close to) the SBG handling the call.

For network announcements functionality is added to support configurable options like delay, duration and repeat on Netann.



4.37 Multiple Languages Support

Feature Identity: FAJ 131 0540/1 R2A, Rev. B

Feature Type: Optional in 14B (FD) to 16B (FD)

Technology:

4.37.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.37.2 Summary

The Multiple Languages Support allows the operator to play announcements towards the end user in the end users preferred language. It is also possible to perform routing of calls based on language preference for instance in order to route calls to a call center supporting the end users preferred language.

Price Object INF 901 5045/39, Multiple Languages Support.

4.37.3 Benefits

End-user

End user will benefit of this feature as he will get all announcements in his/her preferred language.

Operator



Operator can both provide a better end user experience to all his subscribers by being able to play announcements in different languages and route calls according to the subscriber language preference. This is particularly valuable in countries with for example multiple official languages where it may be a regulatory requirement to offer announcements on all the official languages in a country as well as routing of service calls to a call center supporting the specific language. It could also be used by operators who would like to provide a better service to ethnic minorities.

4.37.4 Description

The end user will have an attribute in his subscriber profile indicating his/her language preference. All announcements will be played in the language indicated by the subscriber's language preference.

It is also possible to route calls to service numbers according to the users language preference and terminate the call towards a call center which is using his or her preferred language. Routing of calls can also use the language preference as an input criteria.

4.37.5 Enhancement

Enhancement in MTAS 13A FD1:

- Routing and number translation can also be based on subscriber language preference. This could for instance be used to route calls to service numbers to a call center supporting a specific language.



4.38 Multiple Subscriber Number

Feature Identity:	FAJ 131 0661/1 R2A, Rev. A
Feature Type:	Optional in 14B (FD) to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

4.38.1 Attention

Commercial attention

Not applicable

Dependencies

The following feature is always required:

- MTAS: FAJ1310571 - Flexible Identity Presentation

4.38.2 Summary

This feature enables the possibility for one end-user to have several numbers to be assigned to one subscription. The user can decide on per call basis which number shall be presented for the call made.

Price Object INF 901 5045/66, Multiple Subscriber Number

4.38.3 Benefits

End-user

This feature enables an end-user to have several identities tied to one telephone subscription.

Operator

The network operator can migrate users to IMS and still provide the legacy MSN service to the end-users.



4.38.4 Description

Multiple Subscriber Number feature allows additional numbers to be assigned to the served user. Served user should be able to select the MSN number when making outgoing calls using a Supplementary Service Code. The selected MSN number is used for identity presentation and charging purposes.

If no MSN identity is selected for the call the default Flexible Identity Presentation identity is used. If no MSN identity is selected and there's no default identity configured then the SIP message is left unmodified.

4.38.5 Enhancement

In MTAS 14A the following functionality is new:

All MMTel services are offered common to all numbers in a multi subscriber group except:

- Call Diversion. It shall be possible to set different targets depending on which alias that has been called (Terminating Served User)
- Supplementary Service Codes. It shall be possible to set different targets depending on served user



4.39 Network Announcement

Feature Identity: FAJ 131 0165/1 R3A, Rev. A

Feature Type: Optional in 12A to 16B (FD)

Technology:

4.39.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

MRS:

FAJ 121 2413/2 MRFP Basic

Terminal impacts and dependencies

Not Specified

4.39.2 Summary

The Network Announcement service offers a possibility to an operator to play an announcement to a calling user when fault situations occur during an MMTEL session establishment for example congestion or wrong number. This is enabled by SIP error codes being generated by other nodes than the MTAS, and these error codes are then mapped to relevant announcements.

Price Object INF 901 5045/8, Network Announcement.



4.39.3 Benefits

Enhanced service providing detailed information in case of network problems.

4.39.4 Description

MTAS includes an integrated Media Resource Control Function (MRFC) which controls an external MRFP through the Mp-interface (H.248).

The MRFC function handles the administration, peer association, and recovery mechanisms to control MRFP in order to provide MRF functionality to the MTAS services. Up to 32 MRFPs can be connected to an MRFC.

The Network Announcement service is triggered and an analysis is started when a SIP Error response is received from the terminating network during an MMTel session establishment. Input to the analysis are configuration data for the Network Announcement service and the media capabilities included in the SDP offer sent in the initial SIP INVITE.

If the outcome of the Network Announcement analysis is that an announcement shall be played to the calling party, an MRFP resource is used for playing the announcement. After the announcement has ended, the SIP error response received previously is sent to the calling party.

Announcements can be played for various reasons. For example an announcement may optionally depend on configuration data to be played when incoming or outgoing barring services are invoked.

For more information about announcements related to the standard supplementary services, please refer to the respective Functional Specification.

4.39.4.1 Standards

3GPP/TISPAN TS 183 028

4.39.5 Enhancement

In addition to supplementary services announcements, MTAS is now capable of interpreting standard SIP error codes generated from other nodes in the IMS network and map these to suitable announcements in case of errors during session establishment.



4.40 Network Provided Ringback Tone

Feature Identity: FAJ 131 0696/1 R1A, Rev. A

Feature Type: Optional in 14B (FD) to 16B (FD)

Technology:

4.40.1 Attention

Commercial attention

Not applicable

Dependencies

No technical dependencies have been defined for this Feature

4.40.2 Summary

Network Provided Ringback Tone allows the operator to provide a special ringback tone towards the calling party. This special ringback tone is typically used for marketing purposes.

This is an operator service and it should **not be confused** with Personal Ringback Tone (PRBT).

Price Object INF 901 5045/2, MMTel Extended

4.40.3 Benefits

Network Provided Ringback Tone offers the operator the possibility to use the ringback tone to send out various marketing messages towards the A-party which is typically a subscriber served by one of the competitors. In this way the operator is able to inform many new potential customers about for example attractive offers.



4.40.4 Description

The Network Provided Ringback Tone service allows the operator to configure whether or not to play the special ringback tone. The Network Provided Ringback Tone is played as early media and therefore the A-party is not charged for listening to the ringback tone. The same Ringback Tone is played towards all A-subscribers that are subject to the Network Provided Ringback Tone service.

The operator is able to configure aggregated sound patterns starting with for example an introduction sound, which is played only once, followed by other sound patterns played repeatedly.

4.40.4.1 Standards

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4.41 Number Portability

Feature Identity:	FAJ 131 0575/1 R2A, Rev. A
Feature Type:	Optional in 14B (FD) to 16B (FD)
Technology:	

4.41.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.41.2 Summary

The Number Portability (NP) service in MTAS performs if needed an ENUM query towards DNS/ENUM for number portability information and provides this information to charging system whether an appointed URI (either tel URI or embedded tel URI) is ported in the same operator's network or not. The charging system may be offline charging system or online charging system.

This feature is especially important when tariff can differ between if the call is made in operators own network or not. MTAS has the possibility to inform the caller using an announcement if he is going to make a call to a subscriber served by other operator resulting in a possible higher tariff.

Price Object INF 901 5045/49, Number Portability.

4.41.3 Benefits

End-user

Number portability makes it possible for an end user to change operator and keep his/her phone number. End users can be notified via an announcement that he/she is making a call to a person served by other operator. For these types of calls a higher tariff could be expected.

Operator



For the network operator this feature can keep existing charging structure with different tariffs for calls within operators own network compared to calls terminating in other operators network. This is important in prepaid scenarios when this analysis needs to be done in real time.

4.41.4 Description

Number Portability information can be provided both for outgoing and incoming communication.

For outgoing communication, originating MTAS can generate NP information for:

- SIP INVITE generated by user
- SIP INVITE generated by other service, for instance CDIV use cases
- SIP INVITE that is created as Call Out Of the Blue (COOB) request

For incoming communication, terminating MTAS can generate NP information for:

- SIP INVITE generated by user
- SIP INVITE generated by other service, for instance CDIV use cases
- SIP INVITE that is created as Call Out Of the Blue (COOB) request

4.41.4.1 Standards

IETF RFC 3482, IETF RFC 4694, IETF RFC 3966

4.41.5 Enhancement

New in MTAS 14A is the possibility to configure the node to play an announcement when a call is made to a subscriber served by other operator which could lead to higher tariff. The end user has the possibility to turn this functionality off.



4.42 Offline Charging

Feature Identity: FAJ 131 0164/1 R9A, Rev. A

Feature Type: Optional in 16B (FD)

Technology: IMS

4.42.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.42.2 Summary

MTAS collects detailed traffic information about the handled multimedia sessions and supports the interworking with external Charging Data Function (CDF) via standard Diameter-based Rf-interface.

Price Object INF 901 5045/7, Offline Charging

4.42.3 Benefits

End-user

The end-user can be provided with a detailed bill, specifying the different types of communication used.

Operator

The provided traffic information (e.g. called party ID, start time of the session, etc.) can be used for billing, capacity and trend analysis.

Based on the traffic information, it is possible to create various tariff structures and to generate detailed bills.



4.42.4

Description

The MTAS Offline Charging Function interacts with the services executing on the MTAS node and with the Charging Data Function (CDF). The Offline Charging Function interacts with the CDF over the Rf interface using Diameter Accounting Application messages.

Session based charging is used for charging successfully established communication sessions.

One time event charging is used for reporting of:

- Unsuccessful attempts to establish communication sessions.
 - Supplementary services.

For successfully established communication sessions, Accounting Request (ACR) messages are sent from MTAS to the CDF when the media session is established, updated and terminated.

Charging information is copied from SIP messages and is stored for use when generating charging messages. Additional information (e.g. timestamps) is also stored along with the information from the messages. The content of the stored information depends on the SIP message type (INVITE, BYE, etc.) and the MTAS role (i.e. originating MTAS or terminating MTAS).

As an example, the charging information stored from the message at initial INVITE received by MTAS would include the following:

- IMS Charging Identifier
 - Originating Inter-Operator Identifier
 - Primary CDF Address
 - Secondary CDF Address
 - Incoming SIP Call ID
 - Calling Party Address(es)
 - Requested Party Address
 - Originating Access Network Information

In addition to the charging information defined in 3GPP TS 32.260, the charging messages generated by MTAS can include further charging information in Ericsson vendor specific AVPs, including:



- Feature Tag(s)
 - Access Network Information
 - Address Presentation Information
 - SIP Ringing Timestamp
 - Subscription Identity
 - Supplementary Service Information (e.g. type of Communication Diversion)

4.42.4.1 Standards

3GPP TS 32.260, 3GPP TS 32.299, RFC3588, RFC3455

4.42.5 Enhancement

Enhancement in MTAS14A:

Enhanced possibilities to configure CCR updates. Possibility to include non-normalized CDIV addresses in Requested-Party-Address AVP. The later allows for charging system to see possible prefixes that may influence charging.

Enhancement in MTAS15B:

Support for Directory Assistants/Operator Communication Transfer.

Enhancements in MTAS 16A:

Option to output conference charging information as A to B plus A to C calls only.

IMPI provided at registration will be reported in offline charging.

Enhancements in MTAS 16B:

Narrowband Signalling Syntax (NSS) header and body content support in SIP supported as output to charging AVP.



4.43 Online Charging

Feature Identity: FAJ 131 0499/1 R7A, Rev. A

Feature Type: Optional in 16B (FD)

Technology: IMS

4.43.1 Attention

Commercial attention

Not applicable

Dependencies

No technical dependencies have been defined for this Feature

4.43.2 Summary

The MTAS Online Charging function is a charging mechanism where charging information can affect, in real-time, the service rendered, and therefore requires a direct interaction with the service.

Price Object INF 901 5045/21, Online Charging

4.43.3 Benefits

End-user

Online Charging enables the use of pre-paid services which will add cost control for an end-user. Note that this function only enables pre-paid. It is not the same as a full pre-paid service.

Operator

The competitive edge is increased by being able to offer cost control to end users. The risk for unpaid bills is decreased. Online charging enables the use of pre-paid services. Note that this function only enables pre-paid. It is not the same as a full pre-paid service.



4.43.4

Description

The Online Charging function consists of two sub-functions:

- Charge Service Online: handles the real-time interaction with the charging server to perform online charging for the services requested by the served user.
 - Manage Online Charging: configures online charging.

The Online Charging function within MTAS provides online session charging for the multimedia telephony sessions used to control the media components associated with the multimedia telephony communication service. The message sequences in the following sections focus on the online charging aspects and do not attempt to show complete sessions/dialogs. Please refer to relevant documentation of functions and services that uses the Online Charging function to get a better picture of the complete sequences.

The basic principles applicable to MTAS online charging are:

1. Charging is performed for the overall MMTel communication session.
2. The charging server is responsible for rating the requested communication service based on service information provided by MTAS, including SDP information describing the media session and components.
3. The Online Charging Function establishes a charging session with a charging server in order to perform real time charging. A charging session is only established if there is a valid unexpired Online Charging License available on the MTAS node.
4. The Online Charging Function initiates a charging session by sending a Diameter Credit Control Request (CCR) message, indicating 'Initial Request', to the charging server.
5. During a SIP session establishment, credit reservation is performed based on the media components identified from the INVITE.
6. If a SIP session is cancelled before the establishment is complete, the charging server is informed so that the reserved credit can be refunded to the user's account.
7. During an established SIP session, credit reservation will be revised on receipt of a 200 OK response to an UPDATE and on receipt of a 200 OK response to a Re-INVITE containing an SDP answer.



8. When the service units associated with the reserved credit have been consumed, the charging server is informed and further credit reservation is requested.
9. The charging server can specify a 'Time Quota Threshold' (TQT) when granting service units. This threshold indicates the time, in seconds, before the granted service units have been consumed, that the Online Charging Function should initiate a new credit reservation request.
10. The charging server can specify a validity time when granting service units. The validity time specifies the length of time for which the granted service units are valid, measured from the time that the CCA message is received.
11. When a request is received to terminate a SIP session (e.g. receipt of a BYE message), the charging server is informed of the number of used service units, so that the user's account can be updated accordingly.

4.43.4.1 Standards

3GPP TS 32.260, 3GPP TS 32.299, RFC 3588, RFC 4006

4.43.5 Enhancement

Enhancement in 14A:

Support for prompt and collect functionality is added. With this possibilities to handle use cases like account activation and setting/changing preferred language for a prepaid service is offered. **Note** that this requires that remaining part of the solution (not necessarily coming from Ericsson) like the charging system supports the same type of functionality.

It is possible to configure on a per subscriber basis if CCRs shall be sent for both originating and terminating calls or for only originating or terminating calls.

In addition the Ro interface has been enhanced with several detailed requirements from customers to handle interworking with their respective network and online charging system.

Enhancement in 15B:

Support for Directory Assistants/Operator Communication Transfer.

Extensive configuration to determine if the communication should be terminated or continued dependant on diameter error codes for the Charging Control Failure Handling.



Enhancements in 16A:

Suppression of terminating charging if the subscriber is in HPLMN or if the call is terminating on a fixed secondary device in order to massively reduce the signalling towards OCS.

Option to output conference charging information as A to B plus A to C calls only.

Enhancements in MTAS 16B:

Narrowband Signalling Syntax (NSS) header and body content support in SIP supported as output to charging AVP.



4.44 Operator Anonymous Communication Rejection

Feature Identity:	FAJ 131 0184/1 R1A, Rev. C
Feature Type:	Optional in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

4.44.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.44.2 Summary

This feature enables the operator to control (enable/disable) the anonymous communication in the network.

This is an Ericsson specific service.

Price Object INF 901 5045/2, MMTel Extended.

4.44.3 Benefits

End-user

End-users will be protected from disruptive or unwanted incoming calls.

Operator

The operator can filter out and block anonymous communication from its network.

4.44.4 Description

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.

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 audio announcements is optional and configurable by the operator.

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

In addition to the general configuration, described in ACR feature, the operator can to enable/disable the Operator ACR on node-level using the LDAP-interface.

For additional details please refer to the "ACR" feature.

4.44.4.1

Standards

3GPP TS 24.611, IETF RFC 4745, IETF RFC 3323, IETF RFC 4566, ETSI ES 183 028, IETF RFC 2396, IETF RFC 3966, <http://www.w3.org/TR/2004/REC-xmlschema-2-20041028/>



4.45 Operator Black Lists

Feature Identity:	FAJ 131 0183/1 R1A, Rev. B
Feature Type:	Optional in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

4.45.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.45.2 Summary

This feature enables the operator to create node-level Incoming/Outgoing Black List.

This is an Ericsson specific service.

Price Object INF 901 5045/2, MMTel Extended.

4.45.3 Benefits

End-users will be protected from disruptive or unwanted incoming calls.

Operator

The operator can filter out and block unwanted, illegal traffic from its network which is mandatory regulatory requirement in many countries.

It can be used to prevent the subscribers from the usage of extra expensive (premium) numbers.



4.45.4 Description

The operator can create an Incoming Black List (to filter out the communication into MTAS) and an Outgoing Black List (to filter out the communication originated from the MTAS).

Entries in the Outgoing Black List are matched with the "Request-URI" value of the INVITE message whereas in same use cases they are matched with the "Refer-To URI" values.

Entries in Incoming Black List are matched with the "P-Asserted-Identity" value of the INVITE message whereas in same use cases they are matched with the "Referred-By URI" value.

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.

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 audio announcements is optional and configurable by the operator.

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

The Black Lists are managed by the operator via LDAP-interface.

**4.45.4.1****Standards**

3GPP TS 24.611, IETF RFC 4745, IETF RFC 3323, IETF RFC 4566, ETSI ES 183 028, IETF RFC 2396, IETF RFC 3966, <http://www.w3.org/TR/2004/REC-xmlschema-2-20041028/>



4.46 Operator White Lists

Feature Identity:	FAJ 131 0527/1 R1A, Rev. B
Feature Type:	Optional in 11A to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

4.46.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.46.2 Summary

This feature enables the operator to create node-level Incoming/Outgoing White List.

This is an Ericsson specific service.

Price Object INF 901 5045/2, MMTel Extended.

4.46.3 Benefits

End-user

End-users will have the possibility to always be able to make calls to certain, by the operator, defined numbers or alternatively always be able to receive calls from certain destinations regardless of end user barring settings..

Operator

The operator is able to always allow certain calls. This could be useful for example in a scenario where the end user is not paying his bill. The operator can then for example bar all communication except to and from customer care center.



4.46.4 Description

The operator can create an Incoming White List (to filter out the communication into MTAS) and an Outgoing White List (to filter out the communication originated from the MTAS).

Entries in the Outgoing White List are matched with the "Request-URI" value of the INVITE message whereas in same use cases they are matched with the "Refer-To URI" values.

Entries in Incoming White List are matched with the "P-Asserted-Identity" value of the INVITE message whereas in same use cases they are matched with the "Referred-By URI" value.

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.

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 audio announcements is optional and configurable by the operator.

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

The White Lists are managed by the operator via LDAP-interface.

**4.46.4.1****Standards**

3GPP TS 24.611, IETF RFC 4745, IETF RFC 3323, IETF RFC 4566, ETSI ES 183 028, IETF RFC 2396, IETF RFC 3966, <http://www.w3.org/TR/2004/REC-xmlschema-2-20041028/>



4.47 Outgoing Communication Barring Rule Based

Feature Identity: FAJ 131 0572/1 R2A, Rev. A

Feature Type: Optional in 14B (FD) to 16B (FD)

Technology:

4.47.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.47.2 Summary

This feature enables the operator and the end-user to restrict outgoing communication, using an advanced, rule-based method.

Price Object INF 901 5045/2, MMTel Extended.

4.47.3 Benefits

End-user

The end-user will be able to control his/her own outgoing traffic by creating sophisticated barring rules.

He/She can also combine this destination-based barring with other conditions (Time and Media).

Operator

The operator will be able to offer outgoing barring services to its end-users and to control outgoing traffic from the network.

It will differentiate the operator from the competitors.



4.47.4

Description

The destination-based outgoing communication barring (OCB Destination) uses rule sets.

OCB Destination consists of two parts:

- An operator defined rule set
 - An end-user defined rule set

The rule sets include one or more rules, each rule having one or more conditions as illustrated in the figure below:

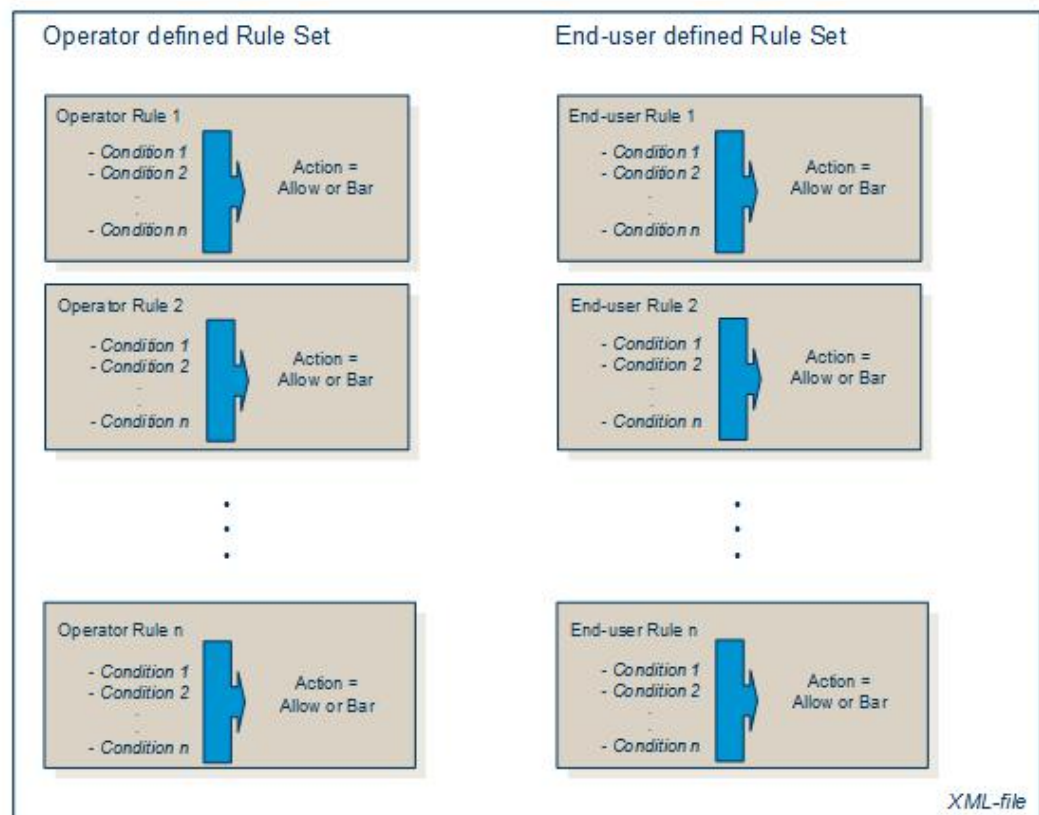


Figure 3 – Barring rule sets



The creator of the barring rule (i.e. the operator or the subscriber) can decide what action to do if the condition(s) matches: bar the communication or allow it.

In the OCB Destination, the condition always refers to the identity of the called party.

When a communication is barred a final SIP response 603 (Decline) is sent to the caller as an indication.

The evaluation method is characterized as follows:

- All the rules in the rule set are evaluated 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 continues normally, 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").
 - At first the operator defined rules are evaluated. Afterwards, the end-user's one.
 - Operator's barring rules override the end-user defined barring rules.

Thanks to the flexibility of the MTAS rule-based barring feature, the OCB service can be defined using different conditions including

- Destination
- Time
- Media

The following barring rule examples can be easily created:

- 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)

In addition to the SIP response 603 (Decline), an audio or video announcement can be played. Playing the audio announcements is optional and configurable by the operator via LDAP.

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

MTAS provides the following node-level configuration options which are managed by the operator via LDAP-interface.

- Communication Barring enable/disable
There is one common administrative state of the Communication Barring function in MTAS. The operator can enable/disable all barring sub-features on node-level (i.e. Barring Programs, OCB, ICB, ACR).
 - Audio Announcement enable/disable
The operator can enable/disable the audio announcement when a communication is barred due to OCB.
 - "Audio only" Announcement
The operator can specify which audio announcement is to be played to the originating party when a communication is barred due to OCB.
 - Video Announcement enable/disable
The operator can enable/disable video announcement when a communication is barred due to OCB.
 - "Video only" Announcement
The operator can specify which video announcement, without audio, is to be played to the originating party when a communication is barred due to OCB.
 - Audio/Video Announcement - Audio part
The operator can specify which audio, associated with a video announcement, to be played to the originating party when a communication is barred due to OCB.
 - Audio/Video Announcement - Video part
The operator can specify the video part of the audio/video announcement.



The user-level configuration is performed by the end-user via the Ut-interface and accessed by the operator via EMA using CAI3G.

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

- Number of barred outgoing communications
 - Number of times an announcement was not successfully initiated

4.47.4.1 Standards

3GPP TS 24.611, IETF RFC 4745, IETF RFC 3323, IETF RFC 4566, ETSI ES 183 028, IETF RFC 2396, IETF RFC 3966, <http://www.w3.org/TR/2004/REC-xmlschema-2-20041028/>

4.47.5 Enhancement

In MTAS 14A support for barring on REINVITE has been added.



4.48 Parlay-X

Feature Identity:	FAJ 131 0528/1 R3A, Rev. B
Feature Type:	Optional in 14B (FD) to 16B (FD)
Technology:	IMS

4.48.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.48.2 Summary

Parlay-X is a standardised interface towards an external application server implementing various value added services. There are several different 'interfaces' belonging to the Parlay-X standard. MTAS 11B provides three optional interfaces called 'Common', 'Call Direction' and 'Third Party Call'. These interfaces makes it possible to influence and initiate calls.

Price Object INF 901 5045/32, Parlay-X

4.48.3 Benefits

End-user

End users will benefit from a wide range of operator specific value added services implemented in external application servers.

Operator

Operators will benefit from the Parlay-X interface via a larger portfolio of attractive end user services. New value-added services can be offered without any design impact on MTAS. By using external Parlay-X gateways like Ericsson's SIG it will even be possible for the operator to let external service providers add services to the operators subscribers.



4.48.4

Description

MTAS supports Parlay-X Call Direction and Third Party Call interface.

Call Direction makes it possible to temporarily break out to an external application server during call setup that for instance can be used for a temporary number use case. The external application server is able to receive different call related parameters as input for its service execution. The external application server (here called VAS server) will based on the input information received from MTAS plus from possible other external interfaces execute one or more services and then return control to MTAS. The VAS server is able to influence the continued call handling in MTAS by for example instruct MTAS to reroute the call to a new destination or terminate the call.

Third Party Call interface can for instance be used for a click to dial use-case where the VAS server can request MTAS to initiate a session on behalf of a user. It is also possible for the VAS server to get the session status for the initiated call as well as order termination of the session.

The call can be initiated in two modes with two modes for Identity presentation and charging for the call:

- Basic mode where the user initiating the call uses the own phone for initiating the call. Caller identity presentation and charging is on behalf the for the originating leg for the call.
- Enhanced mode where the user initiating the call can take the call on any public phone, not only on the own phone belonging to the subscriber establishing the call. The caller identity is set to public identity of user initiating the call and the user initiating the call is charged for legs to both participants.

It is really the service logic in the VAS server that provides the end user or operator services. The Parlay-X feature in MTAS only facilitates the creation of such services.

Parlay-X triggers are stored in the subscriber profile allowing service execution in the MMTel service format.

4.48.4.1

Standards

3GPP TS 29.199-3 (Call Notification) 3GPP TS 29.199-2 (Third Party Call)



4.48.5 Enhancement

New in MTAS 13A:

- Support for Notifycal event message in Parlay-X call notification specification.
- Parlay-X triggers are from 13A stored in the subscriber profile allowing service execution in the MMTel service format.

New in MTAS 13A FD1:

- The call can be initiated in two modes with two modes for Identity presentation and charging for the call.



4.49 Precondition

Feature Identity: FAJ 131 0428/1 R3A, Rev. A

Feature Type: Optional in 16B (FD)

Technology:

4.49.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

MGC:

FAJ 131 0531 SIP Preconditions

Terminal impacts and dependencies

Not Specified

4.49.2 Summary

MTAS supports precondition set-up mechanism in order to allow for deploying MMTel over mobile access with use of precondition at session initiation.

Price Object INF 901 5045/12, Mobile Services.

4.49.3 Benefits

Operator

MMTel can be deployed in the mobile domain.



4.49.4 Description

The precondition set up mechanism is intended for end to end multimedia calls in a mobile network when using conversational bearers. MTAS can both act transparently in a subscriber to subscriber call or as an end-point server. Precondition set up mechanism provides that the conversational bearer resources are set up on the A-side before MTAS starts to send announcement to A. Please refer to the signaling flow below for a typical scenario where precondition is used.

If the communication is to continue to the B-side after an announcement, conferences reservation and other scenarios where a medias session has been established with MTAS as an end point, then preconditions are used in order to ensure optimal bearer capabilities in the radio network.

MTAS provides the operator with the following precondition related configuration parameter

The time limit imposed by MTAS on achieving the preconditions when attempting to play an announcement in early media.

0 has the special meaning that no timer is used to supervise the achievement of preconditions.

4.49.4.1 Standards

3GPP TS 24.229, RFC 3261, RFC 4566

4.49.5 Enhancement

Enhancements in 15B:

If the communication is to continue to the B-side after an announcement, conferences reservation and other scenarios where a medias session has been established with MTAS as an end point, then preconditions are used in order to ensure optimal bearer capabilities in the radio network.

Enhancements in 16B:

Support for NRBT with multiple terminals.



4.50 Priority Call

Feature Identity:	FAJ 131 0490/1 R2A, Rev. A
Feature Type:	Optional in 14B (FD) to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

4.50.1 Attention

Commercial attention

Not applicable

Dependencies

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

SBG:

FAJ 131 0681/2 Priority services

HSS:

FAJ 901 484 Subscription and Notification Support

FAJ 901 820 Priority Support in ISM

CSCF:

FAJ 131 0590 Priority Support Session

MRS:

FAJ 121 3367 Priority Call in BGF

FAJ 121 3656 Priority Call in MRF

SAPC:

FAJ 121 3289 Multimedia Priority Services

MGC:

FAJ 131 0326 ISUP and SIP Interworking: Priority Call Handling with SPC (Germany)

Terminal impacts and dependencies

Not Specified



4.50.2 Summary

This feature covers two services: Priority Call and Priority Service

The Priority Call feature assigns a priority indication in the outgoing requests originated by a subscriber provisioned with the service.

The Priority Service makes it possible to also prioritise (not just signal priority towards PSTN) certain calls within the IMS domain.

Price Object INF 901 5045/24, Priority Call.

4.50.3 Benefits

End-user

An end-user with assigned priority will have increased probability of successful through-connect in case of congestion in the PSTN or IMS network.

Operator

This feature provides compliance to a common regulatory requirement regarding the ability to give priority to important originating users, e.g. government functions.

4.50.4 Description

The Priority Call feature depends on the addition of a new service called 'operator-priority-call' to the operator part of the User Service Data. The presence of this service with a state of 'activated' means the user has the Priority Call service.

An originating session from a user with Priority Call involves the following steps:

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.

In case of a PSTN break-out call, the MGCF will map the emergency value to a corresponding priority flag to be used in the PSTN network in case of congestion.



4.50.4.1 Standards

IMS Core Network Government Industry Requirements for National Security/Emergency Preparedness NGN Priority Services, Issue 2.0, January 2013
Communications Resource Priority for SIP, RFC4412, IETF, February 2006

4.50.5 Enhancement

In MTAS 14A the priority service is added.

Priority Services is an IMS wide feature that ensures priority handling of communication within IMS. Eligibility for priority is granted by other node (CSCF/SBG). MTAS handles prioritization according to indication in incoming SIP messages. MTAS give priority treatment and exemption from certain restriction policies for prioritized communication sessions. Priority Services is a standard mechanism that uses different priority levels indicated in the Resource-Priority header.



4.51 Scheduled Conference

Feature Identity:	FAJ 131 0574/1 R4A, Rev. A
Feature Type:	Optional in 14B (FD) to 16B (FD)
Technology:	IMS

4.51.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

MRS:

FAJ 121 2413/3 MRFP Basic
FAJ 121 2017/3 Audio Conferencing
FAJ 121 2705/2 MRFC

Optional features:

MRS:

FAJ 121 2019/2 Video Basic
FAJ 121 2423/2 Video Conferencing
FAJ 121 2708/1 Content Sharing

Terminal impacts and dependencies

Not Specified



4.51.2 Summary

The Conference feature allows end-users to schedule a conference in advance via a CAS (Conference Administration Server). Invited participants can then participate in the conference by dialing a service number and enter a PIN code.

MTAS implements the role of the conference server. In addition to MTAS an IMS core including media server and a Conference Administration Server is needed.

Note: In MTAS 12A this feature has been renamed to Ad-hoc Conference to clearly distinguish this feature from the new optional feature Scheduled Conference which is introduced in MTAS 12A.

Price Object INF 901 5045/48, Scheduled Conference.

4.51.3 Benefits

End-user

Scheduled conference is a common service to use at work. Easy to use. Fast and effective method for collaboration.

Operator

Scheduled PSTN/ISDN Multi Party Conference service can be offered to subscribers.

Conference service may provide additional income per subscriber either by generating more traffic or by Value Added Service fees.

4.51.4 Description

The Scheduled Conference service provides a Conference Owner with the possibility to create conference resources in advance, distribute an invitation including the conference identity and have Conference Participants, IMS and non-IMS subscribers, dial-in to the conference at a scheduled time. This opposed to the Ad-Hoc Conference service where the conference resource is created once the Conference Creator initiates a traffic session.

The creator of the scheduled conference resource is referred to as the Conference Owner as opposed to Conference Creator which is the term used for ad-hoc conferencing. The reason for the difference in naming of these roles is mainly to emphasis that a scheduled conference traffic session can be initiated without the Conference Owner as opposed to Ad-Hoc which requires Conference Creator initiation.



The scheduled conference resource is created in MTAS by the CO via an external Conference Administration Server (CAS). The user interface to manage personal conference resources via either a web portal and calendar plug-ins is provided by the CAS. Integration with calendar to schedule the conference meeting, corporate directories to invite participants, SMS and E-mail servers to distribute the invitation are also the responsibility of this application. As viewed from MTAS the CAS is an external system for which MTAS provides conference traffic resources. MTAS interfaces the CAS via a CCMP interface. Conference policies can be defined in the CAS and includes:

- Nr of participants
- Allowed media
- Wait for owner

The scheduled conference service of MTAS allows an IMS subscriber to create, read and delete a conference resource (also referred to as a "conference focus") via the CAS.

The main use-cases of the scheduled conference function are:

- Conference owner creates a conference
- Conference owner reads conference status
- Conference owner deletes a conference
- Conference participant joins conference
- Conference participant leaves a conference

Scheduled Conference support Conference Event support, Single Stream Video Switching and Charging.

Conference Event support for Scheduled Conference service:

The Conference Event Support is introduced according to RFC4575, A Session Initiation Protocol (SIP) Event Package for Conference State.

The following use-cases are in scope of conference event handling.

- Participant/Moderator subscribes/un-subscribes/re-subscribes to conference event package
Conference Event Handler processes these requests and sends appropriate NOTIFY messages about the current state of the subscription



- Participant/Moderator joins/leaves conference
Participants subscribed to conference events will be notified about the change of conference state.
- Participant/Moderator holds/resumes conference media
These events also change the conference state. Similar to the previous use-case participants subscribed to conference events will be notified about the new state.
- Special "join" use-case when moderator joins conference in waiting state.
When the state of Conference Service transits from 'waiting' to 'connected' and all participants are connected to the focus in MRFP then participants subscribed to conference events shall be notified about the new state.

Support for Voice activated single-stream video switching as well as content sharing using video is supported in MTAS for multiparty video calls.

The video stream sent to each client is selected by MRFP based on most active speaker criteria. The format and quality of the video stream sent to each client shall be client-controlled as negotiated in SDP.

Scheduled conference service is also enhanced to support charging. Please refer to Offline Charging feature for more details.

AVPF early feedback profile supported for video conference.

Scheduled Conference offers operator assistance at use of wrong PIN code.

4.51.4.1 Standards

3GPP TS 24.147 v7.7.0, 3GPP TS 24.605, 3GPP TS 29.333, RFC 4353 (February 2006), RFC 4579 (August 2006), RFC 3515, RFC 3891 (September 2004), RFC 3892 (September 2004), RFC 5057 (November 2007)

4.51.5 Enhancement

Enhancement in MTAS14B:

Operator assistance when using wrong PIN code.



4.52 Self Administration via Ut-interface

Feature Identity: FAJ 131 0186/1 R6A, Rev. A

Feature Type: Optional in 15A to 16B (FD)

Technology: IMS

4.52.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.52.2 Summary

The Self Administration via Ut-interface feature provides the end-user a graphical user interface which enables to read and modify his/her MMTel service specific configuration data.

Price Object INF 901 5045/10, Self Administration via Ut-interface.

4.52.3 Benefits

End-user

Thanks to this flexible graphical user interface and configuration method end-users can easily customize their communication supplementary services.

The Ut-interface is available for mobile clients, as MMTel and GSM/WCDMA support the same reference point.

Operator

This feature off-loads the operator from end-user administration resulting in decreased OPEX.



4.52.4

Description

The Self Administration via Ut-interface feature enables an MMTel subscriber to read, modify and delete his/her MMTel service specific configuration data, for example settings belonging to his/her Communication Forwarding service.

The Ut-interface is relevant both for terminals supporting Ut as well as self administration via portal.

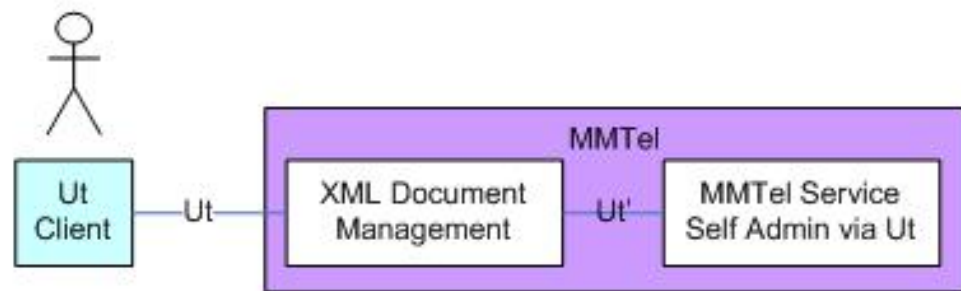


Figure 4 - Self Administration via Ut-interface

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



4.52.4.1 Standards

3GPP TS 24.623

4.52.5 Enhancement

The feature has been enhanced to support the new services implemented.

In MTAS 15A the feature has been enhanced with possibility for the operator to restrict the user to change specific rules through self provisioning. E.g. to be used for the operator to add a protected rule for Default Forwarding of Voicemail.



4.53 Service Domain Selection

Feature Identity:	FAJ 131 0569/1 R2A, Rev. B
Feature Type:	Optional in 12B (FD) to 16B (FD)
Technology:	

4.53.1 Attention

Commercial attention

Not applicable

Dependencies

The following feature is always required:

- MTAS: FAJ1310567 - Service Centralization and Continuity Application Server

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

FAJ 131 0567 SCC-AS

Other node impacts and dependencies

Not Applicable

Terminal impacts and dependencies

Not Specified

4.53.2 Summary

The Service Domain Selection function selects IMS as service domain for VoLTE UE originating at 2G/3G CS access.

Price Object INF 901 5045/46.



4.53.3 Benefits

End-user

This is an operator feature.

Operator

Provides the operator with the capability to handle CS devices in IMS solutions and is a key component in certain network migration scenarios from CS telephony towards VoLTE using IMS.

4.53.4 Description

MTAS provides a standard compliant (3GPP R10) implementation for SCC-AS. Both I2 and Mg interfaces are supported. The Service Domain selection is part of IMS Centralized Services (ICS) which provides one service engine handling multiple accesses (CS and PS). Having only one service engine makes it easier to handle service interactions and provide a consistent end user experience irrespective of used access.

If the I2 interface is used the MSC (or mobile softswitch) is equipped with an ICS client to register the CS attached VoLTE UE to IMS. In case the Mg interface is used then the SCC-AS does not need to interwork with enhanced MSCs.

4.53.4.1 Standards

3GPP TS 24.292

4.53.5 Enhancement

Enhancement in Originating Service Domain Selection, O-SDS, to also include the capability to receive Called Party Number in addition to the Called Party BCD number in the CAP IDP for O-SDS. This is mapped to Request URI in INVITE. Also location information Cell Global Identity (CGI) is copied from CAP IDP to PANI header

Enhancement in Terminating Service Domain Selection, T-SDS, is that SCC AS returns called party number prefixed with IMRN prefix in E.164 International format.

These enhancements are implemented in order for SCC-AS to support similar functionality as BCS Mobility.





4.54 Service Profile

Feature Identity: FAJ 131 0564/1 R2A, Rev. B

Feature Type: Optional in 13A to 16B (FD)

Technology: IMS

4.54.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.54.2 Summary

The Service Profile feature allows the operator to define different service profiles. A service profile is a template that lists a specific set of supplementary services that can be assigned to a group of subscribers.

It will be possible to define subscription packages for end users for example: Gold Silver and Bronze and then define the supplementary services included in each package as a service profile.

Price Object INF 901 5045/43

4.54.3 Benefits

End-user

This is an operator feature.

Operator

This is TCO saving feature. The provisioning of subscribers can be simplified and the memory consumption in HSS/CUDB will be reduced.

Simplifications in provisioning:



- Integration to existing provisioning systems can be simplified since it is enough to provision a subscriber with a service profile identity instead of a complete set of supplementary services.

- When the operator want to add an additional supplementary service to a group of subscriber assigned with one service profile it is enough to update the service profile document and all subscribers will automatically get the new service.

HSS/CUDB memory consumption will be reduced since it is enough to store the service profile ID the user is assigned to instead of the complete list of supplementary services.

4.54.4 Description

Each service profile is assigned with a supplementary service set that can be provisioned to a specific subscriber type. When an user activates a service it will only be the user specific configuration information that needs to be stored as transparent data in HSS for each subscriber, the overall service profile is still common for the subscription package.

4.54.5 Enhancement

New in MTAS 13A:

Support for new supplementary services Hotline and Call Return



4.55 Session Transfer to Own Device

Feature Identity:	FAJ 131 0580/1 R4A, Rev. A
Feature Type:	Optional in 15B (FD) to 16B (FD)
Technology:	IMS

4.55.1 Attention

Commercial attention

Not applicable

Dependencies

The following feature is always required:

- MTAS: FAJ1310544 - AS Controlled Forking

4.55.2 Summary

The Session Transfer to Own Device (STOD) allows the served user to transfer an ongoing communication session from one device to another device belonging to the same user. The devices registered under the served user can be alerted in a serial or parallel mode based on user settings. The served user can set rules via the Ut interface.

Price Object INF 901 5045/55, Session Transfer to Own Device.

4.55.3 Benefits

End-user

An end user is able to change device during an ongoing session to select a device with suitable capabilities or location. One example is a user can change to a video capable device during a session if he/she want to add video in the session using the add drop media function.

Operator



The operator can enhance the end user offering and differentiate from the competitors. Richer sessions can be offered resulting in increased chargeable traffic in the network.

4.55.4 Description

The session can be transferred either by actively pulling it from another device or by reinitiating the call to all original targets of the call.

The list of devices belonging to the service user and related targets contains up to 10 terminal identities and/or related users identities, and it is configured by the network operator or user. These identities are used in rules which specify either a serial or parallel distribution.

To pull the session, the user can dial a feature access code from another device, resulting in that the session is directly transferred to the new device and a reason header sent in the BYE on the original call leg.

Alternatively, the user triggers the service by putting the call on hold and then hangs up. As a result of this procedure the serving MTAS initiates serial or parallel session establishment towards the target devices and/or related users according to previously defined rules. The communication session will be transferred to the target first answering the call. While the parallel or serial distribution is being attempted the other party may, subject to configuration, be played a Call Hold announcement.

4.55.5 Enhancement

Enhancement in MTAS 13A FD1:
Session transfer also supported for originating calls.

Enhancement in MTAS 15A:

Session Transfer by the method of direct Call Pull from an associated device.

Enhancement in MTAS 15B:

For calls transferred using Call Pull mechanism MTAS supports configurable reason header. Default value for reason text is "Call has been transferred to another device". This improves end user experience as the device can use the information to produce a correct device call log. This is used in some VoLTE phones.



4.56 Short Number Dialing

Feature Identity:	FAJ 131 0502/1 R1A, Rev. A
Feature Type:	Optional in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

4.56.1 Attention

Commercial attention

Contact SPM

Dependencies

No internal technical dependencies have been defined for this Feature

4.56.2 Summary

The feature enables the use of short numbers among subscribers belonging to the same group.

Price Object INF 901 5045/27, Short Number Dialing

4.56.3 Benefits

End-user

This is attractive for the SME segment where employees are able to reach colleagues by dialing the extension of the full number.

Operator

Operators are able to charge a premium for the service and so increase revenues.



4.56.4

Description

The MTAS supports short number dialing. Members in a group are able to call each other by means of short numbers common to all members of the group. Only members of the same group are able to reach each other with the short numbers.

It shall be noted that for Short Number Dialing it is the operator that maintains the number plan, whereas in feature 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.

Users are provisioned with a SND number and a SND Domain in addition to their regular identities. A user can only be a member in one SND Domain. The SND Domain may only exist within one IMS network. There is no support over NNI for this feature.

A user with the SND service enabled is then able to dial a short number to reach other users which are part of the same SND Domain.



4.57 Single Radio VCC

Feature Identity:	FAJ 131 0568/1 R4A, Rev. B
Feature Type:	Optional in 15B (FD) to 16B (FD)
Technology:	

4.57.1 Attention

Commercial attention

Not applicable

Dependencies

The following feature is always required:

- MTAS: FAJ1310567 - Service Centralization and Continuity Application Server

The following node features are required in order to use IMS system level feature Session Continuity. For more information, see IMS Feature Description.

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

FAJ 131 0567 - Service Centralization and Continuity Application Server

Other node impacts and dependencies

SBG:

FAJ 131 0594 - Access Transfer Control Function

HSS:

FAJ 901 822 Enhanced 3rd Party Registration Support

FAJ 901 825 Single Radio VCC Support in SDA

FAJ 901 823 Single Radio VCC Support in ESM

FAJ 901 483 MS User Data Access

MRS:

FAJ 121 2412 Border Gateway Function

FAJ 121 2516 Access Transfer Gateway

**SAPC:**

FAJ 121 3290 SRVCC

The following node features are optional for IMS system level feature Session Continuity**CSCF:**

FAJ 131 0660 Extended Registration Event

Terminal impacts and dependencies

Not Specified

4.57.2 Summary

The SRVCC function allows voice call continuity between IMS over PS access and CS access for calls that are anchored in IMS when the UE is capable of transmitting/receiving on only one of those access networks at a given time.

Price Object INF 901 5045/45.

4.57.3 BenefitsEnd-user

This is an operator feature.

Operator

Provides the operator with voice call continuity between PS and CS access.

4.57.4 Description

MTAS provides a standard compliant (3GPP R12) implementation for Single Radio Voice Call Continuity.

When access transfer is initiated the UE that is the subject to the transfer is identified by its C-MSISDN value that is specified by the P-Asserted-Identity header of the INVITE request.

A UE may have multiple sessions but only one session is transferred using the SRVCC procedures: the one that has an active full-duplex speech media component that was most recently activated. Only the speech media is transferred; the other media components are terminated by SCC AS.



Before the access transfer SCC AS acts as a B2BUA: it forwards the SIP signalling between the source access leg and the remote leg. After successful access transfer the SIP signalling is forwarded between the target access leg and the remote leg.

There is a transient period during the access transfer when the source access leg and the target access leg coexist. During that transient period the served UE in PS domain has the possibility to revert the access transfer. This is initiated by a re-INVITE request sent by the served UE in PS domain.

During successful access transfer a new charging session is created for the target access leg and the original charging session is kept as well. The two charging sessions are terminated when the transferred session is terminated.

In addition to handle handover for established calls MTAS is able to handle calls in alerting and pre-alerting state. This enhancement requires support in the SBG/ATCF in order to work end to end.

4.57.4.1 Standards

3GPP TS 23.216, 3GPP TS 23.237, 3GPP TS 24.237

4.57.5 Enhancement

Enhancement in MTAS13A:

MTAS is able to handle calls in alerting state and provide a handover even for calls that has not yet been answered. Implementation is according to 3GPP R10. This enhancement requires support in the SBG/ATCF in order to work end to end.

Enhancement in MTAS15B:

MTAS is able to handle calls in pre-alerting state, see above. Implementation is according to 3GPP R12.



4.58 Subscriber Credit Notification

Feature Identity:	FAJ 131 0561/1 R1A, Rev. B
Feature Type:	Optional in 11B (FD) to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

4.58.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

FAJ 131 0499/1 R4A Online Charging

Other node impacts and dependencies

Not Applicable

Terminal impacts and dependencies

Not Specified

4.58.2 Summary

The Subscriber Credit Notification allows the operator to monitor credit state and play credit balance announcements to online charging (prepaid) users.

Price Object INF 901 5045/40

4.58.3 Benefits

End-user

Prepaid users will be able to get credit balance announcements in different phases of an session.



Operator

Operator can offer a prepaid service using MTAS/MMTel. Calls to customer care center will be reduced when prepaid subscribers automatically get credit balance announcements.

4.58.4 Description

Subscriber Credit Notification 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).

4.58.4.1 Standards

3GPP TS 32.260 v7.6.0



4.59 Terminating Access Domain Selection

Feature Identity:	FAJ 131 0570/1 R4A, Rev. A
Feature Type:	Optional in 14B (FD) to 16B (FD)
Technology:	GSM, WCDMA, LTE, IMS

4.59.1 Attention

Commercial attention

Requires SCC AS Base

Dependencies

The following feature is always required:

- MTAS: FAJ1310567 - Service Centralization and Continuity Application Server

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

FAJ 131 0567 SCC-AS

Other node impacts and dependencies

Not Applicable

Terminal impacts and dependencies

Not Specified

4.59.2 Summary

The Terminating Access Domain Selection (TADS) function is executed in order to select the terminating access domain for the VoLTE UE (CS or PS). The access domain is selected based on registered contacts data, on access type for ongoing VoLTE session and most recent VoLTE session, and on the TADS information (if the current location access network supports IMS Voice over PS) obtained from HSS.



Price Object INF 901 5045/47.

4.59.3 Benefits

End-user

This is an operator feature.

Operator

The TADS function is a key component in certain VoLTE scenarios and facilitates migration from existing CS network to PS (VoLTE).

4.59.4 Description

MTAS provides a standard compliant (3GPP R10) implementation for Terminating Access Domain Selection (TADS). Both I2 and Mg interfaces are supported. The Service Domain selection is part of IMS Centralized Services (ICS) which provides one service engine handling multiple accesses (CS and PS). Having only one service engine makes it easier to handle service interactions and provide a consistent end user experience irrespective of used access.

On a terminating call setup received by SCC AS, the T-ADS procedure takes the following knowledge/data into consideration when deciding where to terminate the call:

- *IMS registration of the VoLTE UE from LTE/PS*
- *IMS registration by the MSC on behalf of the UE*
- *Ongoing VoLTE session access domain, PS or CS*
- *Recently terminated VoLTE session access domain, PS or CS*
- *IMS voice over PS support for the most recently used PS access network retrieved from HSS*

The T-ADS procedure may result in to:

- *terminate over LTE PS*



- *terminate over CS*
- *terminate over LTE PS & CS*
- *reject call*

SCC-AS is also able to include a configurable routing number (rn) parameter to the Request URI in the INVITE request when breaking out call to CS. This functionality is implemented in order for SCC-AS to support similar functionality as BCS Mobility. SCC-AS is replacing BCS Mobility going forward.

When T-ADS is performed for a VoLTE UE located in the CS network, SCC AS request the CSRN from HSS over Sh. When HSS receives the CSRN request, it request the HLR to provide MSRN. The received MSRN shall be returned as the CSRN to the SCC AS.

SCC ASI use the received CSRN in the INVITE in order to reach the terminal.

If T-ADS route towards eMSC (using I2 contact) but fails, the call can breakout using ICS Mg procedures

4.59.4.1 Standards

3GPP TS 24.292

4.59.5 Enhancement

New functionality in 14B includes the following.

CM Control of function "IMS Retry to CS when LTE users not reachable".



4.60 Text Chat

Feature Identity:	FAJ 131 0167/1 R1A, Rev. B
Feature Type:	Optional in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

4.60.1 Attention

Commercial attention

Not applicable

Dependencies

No internal technical dependencies have been defined for this Feature

4.60.2 Summary

MTAS supports 2-Party real-time text communication within MMTel.

Price Object INF 901 5045/9, Multimedia Communication

4.60.3 Benefits

End-user

The feature enables an end-user to chat with other party using real-time text communication which will enhance his/her communication experience.

Operator

This new type of communication generates additional revenue sources.

By offering attractive real-time text communication services, subscriber churn rate can be decreased.



4.60.4 Description

This feature considers set-up and clear down of 2-Party real-time text sessions. Variants of the basic 2-Party session involving support for 100rel (SIP Provisional Message Reliability), early media and SIP forking are supported.

A number of performance counters are provided by MTAS to evaluate the usage and quality of service, like:

- number of initiated sessions
 - number of media streams of type text
 - number of sessions that was not initiated due to internal exceptional events
 - number of sessions that was not initiated due to external exceptional events

4.60.4.1 Standards

3GPP TS 22.173 Stage 1, 3GPP TS 24.173 Stage 3, 3GPP TS 26.114, IETF RFC 3261, IETF RFC 3264



4.61 Video Communication

Feature Identity:	FAJ 131 0166/1 R1A, Rev. B
Feature Type:	Optional in 3.1 to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

4.61.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

MRS:

FAJ 121 2413/3 MRFP Basic

FAJ 121 2019/2 Video Basic

Terminal impacts and dependencies

Not Specified

4.61.2 Summary

MTAS supports the handling of 2-Party video communication within MMTel.

Price Object INF 901 5045/9, Multimedia Communication

4.61.3 Benefits

End-user



The feature enables an end-user to handle advanced, video enabled communication which will enhance his/her communication experience.

Operator

This new type of communication generates additional revenue sources.

By offering attractive video communication services, churn rate can be decreased.

4.61.4 Description

The video communication feature considers set-up and clear down of 2-Party video sessions.

Variants of the basic 2-Party session involving support for 100rel (SIP Provisional Message Reliability), early media and SIP forking are covered by the feature.

A number of performance counters are provided by MTAS to evaluate the usage and quality of service, like:

- number of initiated sessions
 - number of video type media streams
 - number of sessions that was not initiated due to internal exceptional events
 - number of sessions that was not initiated due to external exceptional events

4.61.4.1 Standards

3GPP TS 22.173 Stage 1, 3GPP TS 24.173 Stage 3, 3GPP TS 26.114, IETF RFC 3261, IETF RFC 3264



4.62 Video Fallback to Audio

Feature Identity:	FAJ 131 0549/1 R1A, Rev. B
Feature Type:	Optional in 11A to 16B (FD)
Technology:	Fixed Broadband, Wireless Broadband

4.62.1 Attention

Commercial attention

Not applicable

Dependencies

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

FAJ 131 0166/1 R1A Video Communication

Other node impacts and dependencies

Not Applicable

Terminal impacts and dependencies

Not Specified

4.62.2 Summary

Attempted video calls to devices or networks that do not support video will automatically fallback to an audio call instead.

Price Object INF 901 5045/37, Video Fallback to Audio

4.62.3 Benefits

End-user

The end user will benefit as more call attempts will succeed.

Operator



The operator will benefit from this feature as the number of successful call attempts increase.

4.62.4 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..

4.62.4.1 Standards

IETF RFC 3264



4.63 Wholesale

Feature Identity:	FAJ 131 0635/1 R2A, Rev. B
Feature Type:	Optional in 14B (FD) to 16B (FD)
Technology:	IMS

4.63.1 Attention

Commercial attention

Not applicable

Dependencies

The following node features are optional for IMS system level feature Wholesale/multi-tenancy. For more information, see IMS Feature Description.

Hardware impacts and dependencies

Not Applicable

Internal product impacts and dependencies

Not Applicable

Other node impacts and dependencies

EMA:

FAJ 901 929 DRC - Service Provider

CSCF:

FAJ 131 0613/1 R1A Charging support for Virtual Telephony Operator

SBG:

FAJ 131 0255 SLA performance monitoring and logging

FAJ 131 0582 Optimized BGF Selection

Terminal impacts and dependencies

Not Specified



4.63.2 Summary

Wholesale allows an operator to offer MMTel services to virtual operators. This is achieved by introducing an administrative level called Virtual Telecom Provide (VTP). Per VTP it is then possible to configure data to allow for differentiation between different VTPs.

In order to have a complete end to end solution for wholesale optional features in IMS core may also be needed.

Price Object INF 901 5045/59, Wholesale.

4.63.3 Benefits

End User

N/A. This is an operator service

Operator

An operator is able to significantly increase revenue and traffic by supporting virtual operators in his market.

4.63.4 Description

Wholesale is the process by which an organization owns and operates a telephony network, and sells capacity to other operators, which do not own and operate their own networks. Both the organization that owns and operates the network (the Operating Telephony Provider (OTP)) and their wholesale customers (Virtual Telephony Providers (VTP)) can sell services to end users.

The MTAS supports Wholesale by allowing the OTP to specify a set of VTPs, allowing each VTP to configure almost every aspect of almost every MTAS service for their users, and determining which VTP a user belongs to.

The OTP remains in control of which services are provided by the MTAS, because the OTP administrative state attribute associated with each service takes precedence over the corresponding VTP administrative state attribute.

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.

Each VTP can choose, for each service, to use the OTP values for the attributes rather than configure the service itself.



Each VTP is associated with a set of domains. A served user is deemed to be a member of a VTP if the host part of its primary PUI (the first entry in the user's IRS) is one of the domains associated with that VTP.

Any user not associated with any VTP is treated as a user of the OTP, and the OTP attribute values apply.

The following features in MTAS are supported for wholesale scenarios:

- 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
- Anonymous Communication Rejection
- 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
- Self Administration via SSC
- Flexible Communication Distribution
- Call return
- Hotline
- Address Policing
- Short Number Dialing
- Number Translation service
- Multi Subscriber Number
- Advice of Charge
- Closed User Group
- Dial Tone Management
- Communication Completion on Not Logged-in
- Identity Presentation - CNIP

4.63.5 Enhancement

Enhancement in MTAS 13A FD1:

- More supplementary services supported



4.64 WiFi Calling

Feature Identity: FAJ 131 0791/1 R2A, Rev. A

Feature Type: Optional in 16B (FD)

Technology: LTE, IMS, WLAN

4.64.1 Attention

Commercial attention

Not applicable

Dependencies

Enriched with Multi Device, Multimedia, dependant on TADS.

4.64.2 Summary

Support for terminals with WiFi calling

4.64.3 Benefits

- › Improve user satisfaction with enhanced voice and video calling coverage
- › Possibility to address new revenue potential - depending on the operator's price model
- › Cost-efficient deployment and fast-time-to-market
 - Can be introduced using IMS and EPC without requirements on further investments in Radio Access Network
- › One network for all services - Wi-Fi calling and VoLTE. Same end-user experience over LTE and WiFi with the same services implemented

4.64.4 Description

For networks supporting voice over Wi-Fi access (EPC integrated) when device registered over Wi-Fi, SCC AS avoid the TADS query and attempt to deliver the call to device on Wi-Fi access.

If the call is rejected due to caller preference not found, then SCC AS will try to deliver to VoLTE device on PS.



A specific charging AVP is used to indicate WiFi calling.

4.64.5 Enhancement

Enhancement in MTAS 16B:

New T-ADS timers for Wi-Fi, separate from the legacy VoLTE termination timers, for provisional response (180 and 183).



4.65 WiFi Calling MMTel

Feature Identity: FAJ 901 0060/1 R2A, Rev. A

Feature Type: Optional in 16B (FD)

Technology: LTE, IMS, WLAN

4.65.1 Attention

Commercial attention

Dependant on SI and solution activities.

Dependencies

The feature depend on that the P- Access Network Information SIP header contains both an LTE portion and a Wi-Fi portion. This could potentially be achieved by P-CSCF SMM script e g based on ANI from device chipsets or separate location nodes/functions that some operators have in their network.

4.65.2 Summary

Enable location services based on mobile nw information to work for WiFi Calling, while still charging based on Wi-Fi vall delivery.

4.65.3 Benefits

Location services enabled for WiFi Calling.

4.65.4 Description

Location Services such as local number format dialling, local service numbers and location based service number redirect (e g £TAXI) is supported for WiFi Calling.

The location for registered devices is taken from the LTE/3G/2G portion of the P-ANI and the charging from the Wi-Fi portion.



4.65.5 Enhancement

Enhancement in MTAS 16B:

Announcement for failed short code dialling due to Wi-Fi access when cell information is missing in the PANI and short code dialling is not supported for fixed/Wi-Fi access type attachment.