

MTAS Interface to CSCF (ISC, Ma, Pw)

INTERWORK DESCR

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1 Document History

Rev	Date	Sign	Comment
A	2013-04-09	EJZSBAR	Initial revision for MTAS 13B based on 1/155 19-CRA 119 2104 (rev. J).
B	2013-09-25	EDMHORV	Added Japanese Charging Added Supported header to OPTIONS Added Resource-Priority header
C	2013-10-01	ELSZFST	Added X-GENERIC-NUM header
D	2014-05-05	EXUEFLI	Updates regarding PCV header support for Japanese Charging Update P-com.TimeZone header format Updated the description of the no-fork directive in Request-Disposition header to make it more generic since it is used by several services, where Gateway Model is one of them. Chapter 4.4 updated.
E	2014-11-10	ETXEHG	Updates regarding ST AS Added ST AS port Added support for P-Profile-Key header
F	2015-04-13	EXXGDDI	Update P-Served-User header by NCC when AS Chaining is enabled and O-CSI is applicable.
G	2015-10-19	EDARSAC	Updated with information on tel-URI and phone-context in new section 5.4 Added reference for mapping Accept-Contact and Contact headers in chapter 4.4
H	2016-04-15	ETXEJD /EXXGDDI	Chapter 4.3.2.2 Updated with response code 482 Loop Detected due to the Communication Diversion Loop Detection function in MMTEL AS. Chapter 5.2.5, update P-com.TimeZone header for originating case from INVITE to ACK/PRACK
J	2016-05-23	XMILMAT	Reference section updated, CBA links generalized to MOM.
K	2016-11-21	EYAYFEN	Added 5.2.14, 5.2.15 P-Asserted-

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L	2017-02-28	EEVASAN	Added Generic port in chapter 2.5 Chapter 4.2 updated with OOD SUBSCRIBE handling. Chapter 4.5.24 added
M	2017-05-30	EZHAOSO	Change chapter 4.4 to add MTAS usage part for Refer-To header.
N	2017-06-28	ESAUSIK	Chapter 4.3: Updated 305 Use Proxy SIP Response for Re-balancing Feature Chapter 4.4: Updated SIP Headers for Contact header for 305 Use Proxy for Re-balancing Feature
S	2017-11-07	ETHAMEE	Chapter 4.4: Updated for new Ericsson Proprietary Header P-Ericsson-Original- Contact Header. Chapter 5.2.16: Updated of P-- Ericsson-Original-Contact Header description.
T	2017-12-05	ESAUSIK	Chapter 4.3: Updated 500 Internal Server Error SIP Response for Re- INVITE Chapter 4.4: Updated SIP Headers for 500 Server Internal Error
U	2018-03-13	EEVASAN	Chapter 4.3: Updated 500 Internal Server Error SIP Response for Re- INVITE ACK crossing Chapter 4.4: Updated SIP Headers for 500 Server Internal Error Chapter 4.4: Updated P-Visited- Network-Id description Chapter 2.5 updated with information about generic SIP port
V	2018-05-25	EEVASAN	Chapter 4.4: Route header updated with information about initialSelection parameter. Chapter 4.4 updated and 5.2.17 added for P-Ericsson.Invocation-History header.

2 Scope and Purpose

2.1 Scope

This document defines how MTAS uses the ISC, Pw and Ma interfaces.

2.2 Interface Entities

Within the IMS architecture, the interface between MTAS and the S-CSCF is the ISC. The ISC interface is implemented using the SIP protocol.

The MTAS also uses the SIP protocol on the Ma interface with the I-CSCF and the Pw interface to the Presence Server.

This document describes how MTAS uses the SIP protocol. The description of the SIP protocol within this document includes

- Supported SIP Methods
- Supported SIP Responses
- Supported SIP Headers
- Supported SIP Bodies

Any routers or proxies that may be installed between MTAS and the CSCF are excluded from the scope of this document.

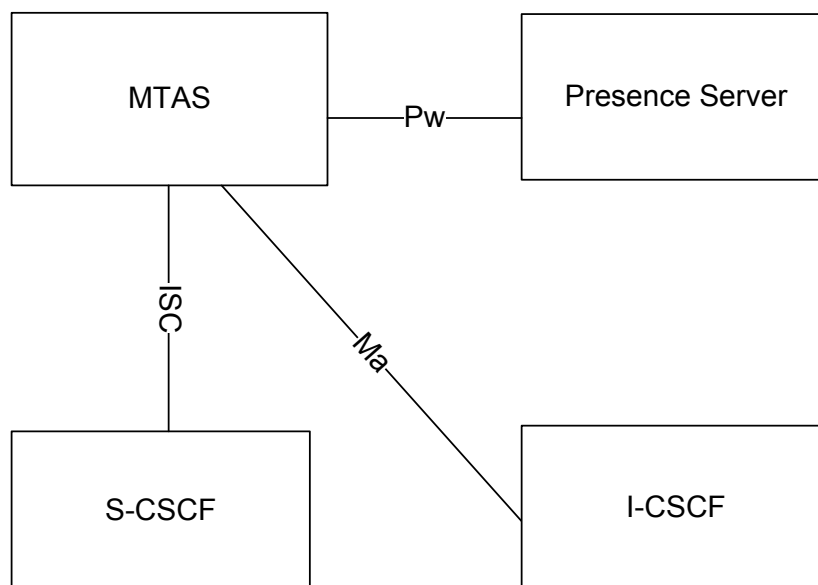


Figure 1 – MTAS connectivity

2.3 Interface Role

2.3.1 ISC Interface

SIP methods combine to support multi-media telephony communication and the associated multi-media services as well as support of communication with PBXs and services serving the PBXs.

In the context of the ISC interface, the role of MTAS can be

- User Agent Client
- User Agent Server
- Back-to-Back User Agent

The user agent role of the MTAS differs depending upon the transaction and service being invoked.

In general end-to-end call usage the MTAS acts as the B2BUA. In this role the MTAS will appear as a concatenated User Agent Server and User Agent Client as defined in RFC3261 [4].

The MTAS acts a User Agent Server when it rejects a SIP Request for whatever reason.

The MTAS act as a User Agent Client when it generates a SIP Request.

The MTAS will appear as a pair of back to back User Agent Clients when performing third party call control and originates sessions towards two separate clients.

2.3.2 Ma Interface

The MTAS receives requests on the Ma interface for Public Service Identities (PSI) or wildcarded PSIs (W-PSI).

The MTAS may send SIP Requests on the Ma interface directly to the I-CSCF when the served users S-CSCF is unknown.

From an MTAS perspective the Ma interface is functionally equivalent to the ISC interface, with the following additional functionality:

When MTAS act as a User Agent Client on the Ma interface, it tries to contact the next element in the result of DNS SRV and/or A/AAAA lookups when a dialog creating request has encountered transport failure or timeout.

2.3.3 Pw Interface

The Pw interface is defined for the Presence Service as being between a Watcher and the Presence Server. MTAS acts as a Watcher of served user presence. Presence related initial requests are sent on the ISC or Ma interfaces to the S-CSCF.

From an MTAS perspective the Pw interface is functionally equivalent to the ISC interface.

2.4 Services

Table 1: Offered Services

Offered Service	Description
User Agent Server	Acts as a UAS as defined in RFC 3261, [4], and as such receives SIP Requests and provides Responses.
User Agent Client	Acts as a UAC as defined in RFC 3261, [4], and as such generates SIP Requests and receives Responses.
Back to Back User Agent	The B2BUA service allows the concatenation of UAC and UAS services under control of the application to provide an end to end session capability.

Table 2: Used Services

Used Service	Description
SIP Proxy	The SIP proxy roles for S-CSCF or I-CSCF as defined in [19] for passing of SIP Requests and Responses.
User Agent	Remote UA for source and sink of SIP Requests or Responses for SIP methods.

2.5 Encapsulation and Addressing

SIP Requests and Responses are carried over TCP or UDP. On network layer either IPv4 or IPv6 can be used. In SIP headers and bodies IPv4 and IPv6 addresses can be mixed.

When using IPv4 the address 0.0.0.0 has a special meaning in some scenarios. In IPv6 there are several alternatives. One option is to use the IPv6 zero address, ::0 or just ::, another option is to use a domain within the .invalid top level domain, for example this.is.invalid. The value of attribute mtasFunctionInvalidAddress is used for IPv6 in all such scenarios.

See [30] for details of how to configure the IP ports.

MTAS receives initial SIP Requests on either the generic SIP port or on one of the following dedicated ports, the values of which are configurable:

- Originating Port
- Originating Unregistered Port
- Terminating Port
- Terminating Unregistered Port
- SCC Originating Port
- SCC Originating Unregistered Port
- SCC Terminating Port
- SCC Terminating Unregistered Port
- ST AS Port
- Presence Subscription Port (Pw reference point)
- PSI Port (Ma reference point)

The generic SIP port can be used by MMTel AS, SCC AS and NW AS over the ISC interface and by PSI services over the Ma interface.
For more information about the generic SIP port see chapter 8.1 in [63].

3 Procedures

3.1 Overview

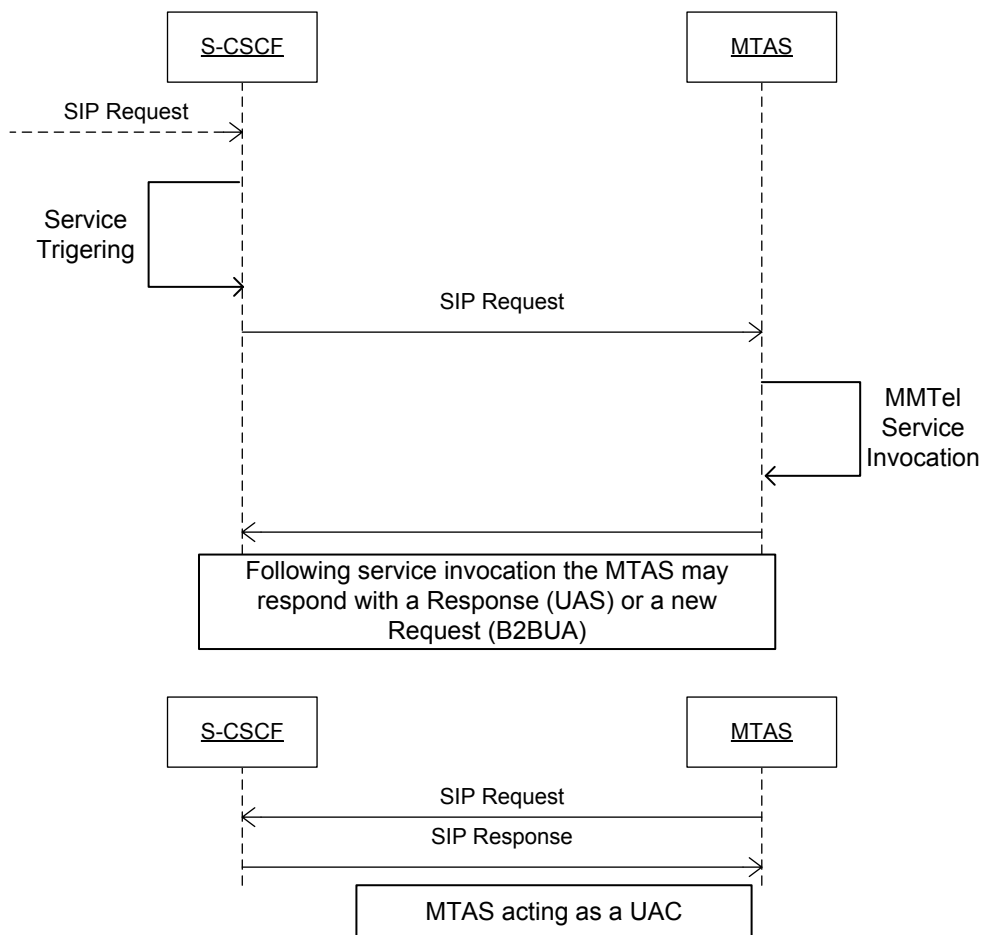


Figure 2 - Scope – ISC interface

MTAS send and receives SIP Requests and Responses on the ISC interface. The SIP Requests and Responses will include SIP headers appropriate to the specific Requests or Responses. The SIP Requests and Responses may also include bodies when required.

The MTAS may act as an User Agent Client (UAC), a User Agent Server (UAS) or as a back to-back user agent (B2BUA) for the handling of messages.

When the MTAS receives a SIP Request or Response the MTAS may:

- Pass the received SIP Request or Response on the ISC interface, possibly with modified headers or header values

- Send a different SIP Request or Response on the ISC interface
- Not propagate (consume) the SIP Request or Response.
- Add, remove or modify bodies included in a SIP Request or Response which is passed back to the ISC interface.

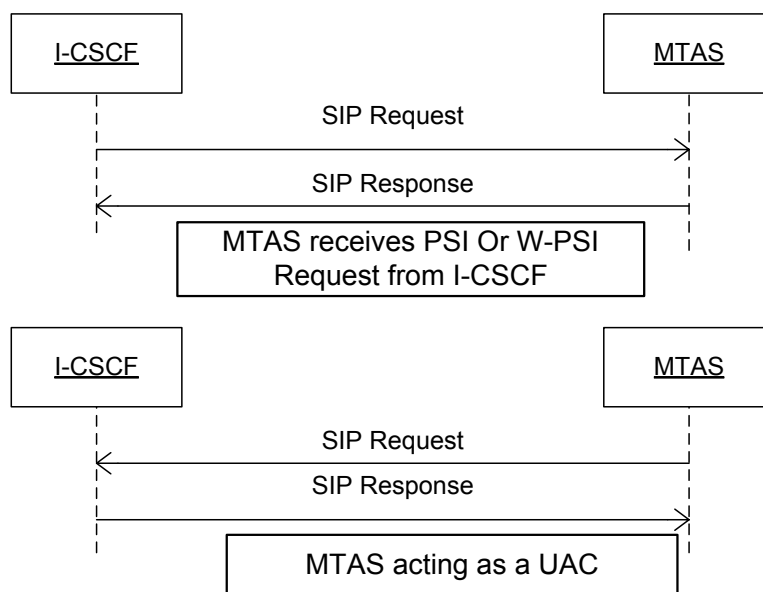


Figure 3 - Scope – Ma interface

MTAS sends and receives SIP Requests and Responses on the Ma interface. The SIP Requests and Responses will include SIP headers appropriate to the specific Requests or Responses. The SIP Requests and Responses may also include bodies when required.

MTAS acts as a User Agent Server (UAS) when receiving SIP Requests on the MA interface. MTAS acts as an User Agent Client (UAC) when sending SIP Requests on the Ma interface.

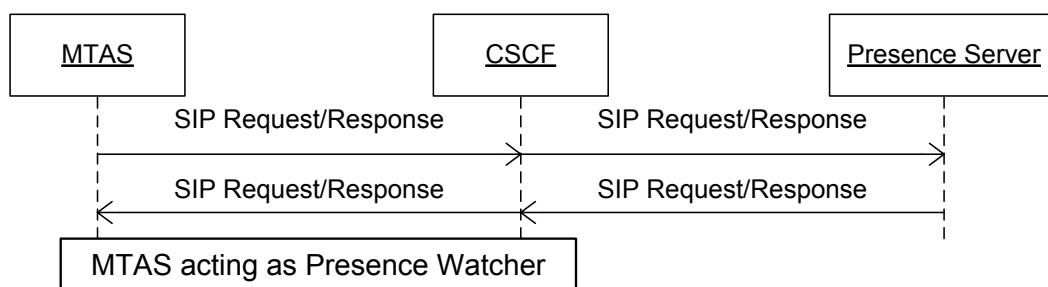


Figure 4 - Scope Pw Interface

When MTAS acts as a Presence Watcher the MTAS will send a SIP Request to the CSCF which will route the Request to the Presence server. Subsequent SIP Requests and Responses are also routed via the CSCF.

3.2 Lower Level Procedures

N/A

3.3 Back-To-Back User Agent

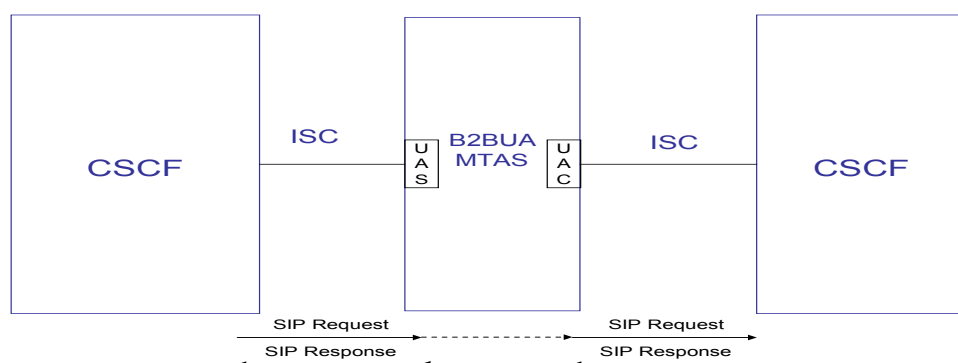


Figure 5 – MTAS B2BUA Context

The Back-to-Back User Agent functionality is formed by the concatenation of a User Agent Server and a User Agent Client for the same end-to-end session.. The UAS will receive a SIP Request which will be handled by the MTAS application logic with the result that an equivalent SIP Request may be sent from the UAC part of the MTAS. When a SIP Response is received by the UAC part of the MTAS an equivalent Response may be sent from the UAS part of the MTAS.

3.4 User Agent Client

The User Agent Client service on the MTAS generates SIP Requests and receives SIP Responses as defined in RFC 3261, [4].

3.5 User Agent Server

The User Agent Server service on the MTAS received SIP Requests and generates SIP Responses as defined in RFC 3261, [4].

4 Information Model

4.1 General

This section describes the supported SIP methods, the status codes generated by the MTAS platform and the services deployed on the MTAS platform and the SIP headers and bodies used by the MTAS and deployed services.

MTAS operates in the following modes:

- B2BUA – Back to Back User Agent
- UAC – User Agent Client
- UAS – User Agent Server

MTAS generates and acts upon the headers and bodies described in sections 4.4 and 4.5.

Table 3 shows the handing of methods, headers and bodies when acting in each mode.

Table 3 - Method, Header and Body handling in MTAS

Mode	Methods	Headers	Bodies
B2BUA	Passes received SIP Requests in new dialog. Only Requests in section 4.2 are supported.	Headers shown in section 4.4 may be added, removed or passed with modified values between received and sent Requests and Responses. Other headers are passed transparently	MTAS will only add or remove or act upon bodies described in section 4.5. Other bodies received will be passed transparently in the sent SIP Request or Response.
UAC	Only Requests included in section 4.2 will be sent.	Only headers included in section 4.4 are included in Requests sent from MTAS or are used in received Responses.	Only bodies described in section 4.5 and marked as “Receive only” or “Generates and receives” will be sent from the MTAS.

Mode	Methods	Headers	Bodies
UAS	Only Requests included in section 4.2 are supported. Responses shown in 4.3 may be sent in response.	Only headers included in section 4.4 are used by the MTAS when received in Requests or are sent in Responses.	Only bodies described in section 4.5 and marked as "Receives only" or "Generates and receives" will be used by the MTAS.

4.2 SIP Requests

The following table shows the SIP methods supported by MTAS. Details are included as to whether the method is only accepted and passed on by MTAS, whether MTAS generates the method, if the method may be used to create a new dialog and details of when the method is used. Methods which are shown as being received by the UAS and sent by the UAC for the same dialog type (N/E) may be considered to have been passed by the MTAS in B2BUA mode, see Section 3.3.

Table 4 - Supported SIP Methods

Method	Ref	UAS ¹	UAC ¹	Usage
INVITE	[4]	N/E	N/E	
ACK	[4]	E	E	
CANCEL	[4]	N/E ²	N/E ²	
BYE	[4]	E	E	
PRACK	[6]	E	E	
UPDATE	[3]	E	E	
REFER	[11]	E	E	

Method	Ref	UAS ¹	UAC ¹	Usage
NOTIFY ³	[7] , See also 5.1.1, [53], [61]	E	N/E	<p>NOTIFY is sent and received as part of the SUBSCRIBE dialog for Communication Completion or Conference (both ad-hoc and scheduled type) and an indication of status for a REFER request.</p> <p>NOTIFY is sent outside of established dialog for Dial Tone Management indication, for reporting User Location Info, and for Emergency Call start/stop indication.</p> <p>NOTIFY is received as part of the SUBSCRIBE dialog for SCC AS to obtain the served user's Registration Information.</p> <p>NOTIFY sent inside SUBSCRIBE dialog with event package 'dialog' is used for dialog event notifications.</p>
REGISTER	[4]	N		
SUBSCRIBE	[7], [53], [61]	N/E	N/E	<p>SUBSCRIBE is sent and received as part of Communication Completion service and sent to the Presence Server to obtain served user Presence status.</p> <p>SUBSCRIBE accepted by the Conference MTAS (both for ad-hoc and scheduled conferences) on an existing dialog to notify participants of the status and status changes of conference participants.</p> <p>SUBSCRIBE request to conference events received outside a dialog or within an early dialog is rejected by the Conference MTAS with "403 Forbidden" response.</p> <p>SUBSCRIBE is sent as part of the SCC AS and ST AS services to obtain the served user's Registration Information from the registrar (S-CSCF).</p> <p>SUBSCRIBE received outside a dialog with event package 'dialog' used as part of dialog event notifications.</p>
MESSAGE	[25]		S	<p>MESSAGE is sent as an indication to the served user by Communication Diversion service.</p>

Method	Ref	UAS ¹	UAC ¹	Usage
OPTIONS	[4]	S/E	E	<p>May be sent outside a dialog to one of the configured addresses defined in MTAS, as a ping to the MTAS, which will respond without a SDP body, ref [44]. May contain a 'Supported' header.</p> <p>May be sent outside a dialog to an address not defined in MTAS. MTAS will answer with 501 Not Implemented response.</p> <p>Maybe sent in an early dialog to MTAS which will respond with 200 OK without SDP.</p> <p>When MTAS has one well defined session between incoming and outgoing side, MTAS is transparent to OPTIONS request and response.</p> <p>OPTIONS may be sent in an existing conference dialog to the MTAS Conference server which will respond with details of Conference server capabilities.</p> <p>OPTIONS maybe sent in an established dialog of a 3rd party call. MTAS will respond with 200 OK with SDP.</p> <p>OPTIONS sent in an established incoming dialog when no corresponding outgoing, established dialog exists will be answered by MTAS with 200 OK without SDP.</p>
INFO	[26]	E	E	
PUBLISH	[15]	E	E	PUBLISH is sent and received as part of the Communication Completion service.

Notes:

- 1 UAS and UAC indicate if the method is supported when received by MTAS from another node or is sent from the MTAS respectively. The table indicates if the SIP method may be sent or received on; an existing dialog – E, as a new dialog creating request – N, or as a standalone transaction – S.
- 2 CANCEL is only sent for an INVITE transaction but may be sent for an INVITE for which there is no existing dialog, or for a 're-INVITE' sent on an existing dialog.
- 3 NOTIFY is a dialog creating response to a SUBSCRIBE Request. Where E is included for a NOTIFY Request the NOTIFY acts as the dialog creation message.

4.3 SIP Responses

SIP responses received in B2BUA mode are usually passed transparently through the MTAS. This means that any valid SIP response code may be passed on the ISC interface. The error text included in a received reject response is passed through unchanged.

SIP responses fall into two major categories:

- Error Responses – generated by the MTAS node software to indicate errors in the received SIP message (section 4.3.1).
- Service Responses – generated by deployed services as responses to usage of specific deployed services (section 4.3.2).

The same SIP response may be used for both MTAS Node and deployed service responses, e.g. “500 Internal Server Error” may be generated by the platform or from a service where an error has occurred.

Responses are defined in RFC 3261 [4] except where specifically referenced.

4.3.1 Error Responses

Error responses to SIP methods are summarized in Table 5. These responses are generated by the platform due to errors in the Request received. The warning, see [4], is added to the error response as additional information about the node issuing the response.

Table 5 - Error responses generated by MTAS.

Status Code	Reason Phrase	MTAS could not process the request because...	Warning Code
400	Bad Request	The request could not be understood due to malformed syntax.	399
403	Forbidden	The request is not allowed outside a dialog.	399
405	Method Not Allowed	The request is not allowed.	399
408	Request Timeout	Other SIP servers down the signalling path did not respond in a timely fashion.	399
416	Unsupported URI Scheme	The Request URI is neither a tel URI nor an embedded tel SIP URI.	399
420	Bad Extension	The “199” option-tag is included in the Require and/or Proxy-Require header field of the dialog establishment request, e.g. INVITE.	399

Status Code	Reason Phrase	MTAS could not process the request because...	Warning Code
422	Session Interval Too Small	The session expiry interval suggested in the INVITE Request was smaller than the configured minimum value (RFC4028 [1])	399
480	Temporarily unavailable	The served user is temporarily unavailable.	399
481	Call/Transaction Does Not Exist	The request could not be identified as part of any known session.	399
487	Request Terminated	The call was cancelled	399
488	Not Acceptable Here	The UPDATE request is received on an early dialog that is terminated.	399
491	Request Pending	MTAS has another request pending.	399
500	Internal Server Error	May be received from external systems that did not perform as expected or due to internal processing error. Reject of SUBSCRIBE and re-SUBSCRIBE requests in shutting down. "Another request is being processed" when the Re-INVITE Delay/Retry Timer is running. Early reception of re-INVITE followed by ACK for 200OK(INVITE or re-INVITE).	399
501	Not Implemented	MTAS could not recognize the request method.	399
503	Service Unavailable	The service was disabled or system is overloaded.	399

Text associated with the Warning Code is dependent upon the reason the response was generated.

4.3.2 Service Responses

The following SIP responses may be provided by the services deployed on the MTAS platform. These responses are generated as a result of normal session handling. These responses are divided into 2 categories:

- Provisional Responses, 1xx series
- Final Responses – These may be either acceptance of the SIP method, 2xx series, or rejection of the SIP method, 3xx, 4xx, 5xx and 6xx series.

4.3.2.1 Provisional Responses

Provisional responses are only sent or received in response to an INVITE Request.

Table 6 - Provisional Responses

Status Code	Reason Phrase	Comment...
100	Trying	
180	Ringing	
181	Call is being forwarded	
182	Queued	
183	Session Progress	
199	Early Dialog Terminated	Like all the others, it may happen that 199 as the first provisional response is creating a new early dialog [45].

4.3.2.2 Final Responses

The warning, see [4], is added to the error response as additional information about the node issuing the response.

Table 7 - Final Responses

Status Code	Reason Phrase	Comment...	Warning Code
200	OK	Acceptance for all requests except REFER	
202	Accepted	Acceptance of REFER (RFC3515 [11]) and SUBSCRIBE	
301	Moved permanently		
302	Moved temporarily		
305	Use Proxy	Redirect users to target MTAS node (Re-Balancing by moving users between MTAS instances) vMTAS only.	
400	Bad Request		399
403	Forbidden		399
404	Not found		399
406	Not Acceptable		
408	Request timeout		399
410	Gone		399
416	Unsupported URI Scheme		399
420	Bad extension		
421	Extension required		
433	Anonymity Disallowed	(RFC5079 [41])	399

Status Code	Reason Phrase	Comment...	Warning Code
480	Temporarily unavailable		399
482	Loop Detected		399
486	Busy here		399
487	Request terminated		399
488	Not acceptable here		399
500	Internal server error	When the LRBT, NRBT, CAT, FCD or Ad hoc Conferencing Service is triggered, then an outgoing rejection on reINVITE will be accompanied with a Retry-After header, and if the Re-INVITE Delay/Retry Timer is running, a warning-header: "Another request is being processed" will also be sent out	399
502	Bad gateway		399
503	Service unavailable		399
603	Decline		399
606	Not acceptable		399

4.4 SIP Headers

The MTAS will transparently pass SIP headers except those listed in Table 8, when working in B2BUA mode. Headers shown in Table 8 may be modified by the MTAS.

When working as a UAS or UAC headers in received SIP requests or Responses not included in Table 8 will be ignored.

When sending a new SIP Request or Response, as a UAC or UAS, MTAS will include all headers specified as mandatory in the RFC where a SIP Method or header is defined, see Table 4 and Table 8. MTAS will include the headers included in Table 8 where applicable.

MTAS usage is only shown where the MTAS specifically inserts these values into the header in a method and the usage is not specifically defined in the referenced RFC.

Table 8 - SIP Headers

Header	Ref	MTAS Usage
Accept	[4] 20.1	MTAS will pass through received values and insert content types for bodies which the MTAS is prepared to receive.
Accept-Contact	[12] 9.2 See also [58] 2.3.3	MMTel specific feature tags may be added.

Header	Ref	MTAS Usage
Alert-Info	[4] 20.4	A communication waiting Alert-info header maybe added.
Allow	[4] 20.5	MTAS will update received contents to include SIP Methods used to support specific MMTel services.
Call-ID	[4] 20.8	MTAS creates new Call-Id values which are mapped between received and sent messages.
Call-Info	[4] 20.9	MTAS will remove the informational elements from the Call-Info header of the SIP messages that are sent by the served user if the informational element has the purpose=call-transfer;m=consultative parameters.
Contact	[4] 20.10, and 21.3.4 See also [58] 2.3.3	A new Contact is included in a sent initial Request. MTAS maps the Contact header in subsequent requests and responses between received and sent messages. For 305 Use Proxy responses, the requested resource MUST be accessed through the proxy given by the Contact field. The Contact field gives the URI of the proxy. The recipient is expected to repeat this single request via the proxy.
Content-Disposition	[4] 20.11 [21]	When this header is absent the default values for content disposition and handling for a body are implied.
Content-Length	[4] 20.14	
Content-Type	[4] 20.15	
Cseq	[4] 20.16	
Diversion	[18]	
Event	[15] 4, 6, See also 5.2.1 , [53]	
Expires	[4] 20.19, [15] 4, 5, 6	
Feature-Caps	[51] 6.2	If SRVCC for alerting calls is enabled, SCC AS will include the header in SIP INVITE or SIP 1xx/2xx responses where applicable.
From	[4] 20.20	From header may be modified between received and sent initial requests. MTAS generates a new from-tag for a sent initial request. The From header, including from-tag, is mapped between received and sent responses and subsequent requests.
History-info	[16] 4.1	Inserted to indicated call diversion. Existing entries maybe modified to include served user privacy settings.
Info-Package	[50] 7.2	SRVCC for alerting calls includes Info-Package in SIP INFO
Max-Forwards	[4] 20.22	
MIME-Version	[4] 20.24	
Min-Expires	[4] 20.23, [15] 5, 6	
Min-SE	[1] 5	
Organization	[4] 20.25	
Path	[47] [48]	
P-Access-Network-Info	[10] 4.4	
P-Asserted-Identity	[5] 9.1	
P-Asserted-Service	[60] 4.1	
P-Area-Info	See 5.2.2	Used by the Japanese Charging service for ICBS data.

Header	Ref	MTAS Usage
P-Charging-Function-Addresses	[10] 4.5	
P-Charging-Vector	[10] 4.6 See 5.2.4 [55]	In addition to usage in SIP specification [10], it is used by Japanese Charging service for ICBS and FCH data.
P-Com.TimeZone	See 5.2.3	Carries Time Zone and DST info.
P-Com.PrivateUserID	See 5.2.4	Indicates the IMS Private User ID (IMPI).
P-Com.User-Equipment-Info	See 5.2.5	Carries info on user equipment.
P-Early-Media	[17] 8	MTAS includes in provisional responses when an announcement is to be played prior to session establishment or rejection. MTAS will pass through a P-Early-Media received in a provisional response.
P-Ericsson-Original-Contact	See 5.2.16	When Multi Mobile subscriptions feature is enabled in MTAS, P-Ericsson-Original-Contact is used to identify the mobile subscription used in originating session scenarios.
P-Ericsson.Invocation-History	See 5.2.17	Specifies which Application servers the initial INVITE message has passed in case the generic SIP port used. Added by MMTel, SCC and NW AS in outgoing initial INVITE messages.
P-Preferred-Service	[60] 4.2	
P-Private-Network-Indication	[40]	
P-Profile-Key	[57]	Carries a wildcarded PSI (W-PSI). ST AS may (depending on connection mode) use the W-PSI to fetch the main PBX identity from HSS. When routing the call to the terminating PBX or to the terminating network the ST AS discard the P-Profile-key header.
P-Served-User	[46] [63] 8.1	<p>The attribute <code>mtasSipSupportPServedUserHeader</code> [30] defines if the P-Served-User header is supported. When supported, the served user is primarily determined from the received P-Served-User header.</p> <p>The P-Served-User header can be inserted by MTAS for Call Out of Blue sessions, or by Northbound Call Control (NCC) service, the service handling CAMEL interaction, when AS Chaining is enabled and O-CSI is applicable.</p> <p>In case the generic SIP port is used over the ISC interface the P-Served-User header must be supported, <code>mtasSipSupportPServedUserHeader</code> set to enabled.</p>
P-Visited-Network-ID	[10] 4.3	
Priority	[4] 20.26	
Privacy	[8] 4.2	Inserted or modified based upon served user privacy settings or when served user privacy is applied, prior to sending Request or Response to user.
Rack	[6] 7.2	
Reason	[9]	Q.850 cause values in the Reason header in a SIP Response may be used by the MTAS to determine subsequent session handling actions.
Record-Route	[4] 20.30	
Recv-Info	[50] 7.3	SRVCC for alerting calls will send/receive Recv-Info header containing the applicable <code>g.3gpp.state-and-event</code> package name.
Referred-By	[14] 3	

Header	Ref	MTAS Usage
Refer-To	[11] 3	MTAS can use IMPU or Replace parameter in Refer-To header to move active session to existing conference.
Replaces	[13] 6.1	
Reply-To	[4] 20.31	
Request-Disposition	[12] 9.1	Support of the "no-fork" fork-directive.
Require	[4] 20.32	
Resource-priority	[55]	MTAS understands ets and wps namespaces of the Resource-priority header.
Retry-After	[4] 20.33	Receiving: Handled by MMTel AS services LRBT, NRBT, CAT, FCD or Ad hoc Conferencing Service if it is received in 500 Internal Server Error in response to re-INVITE requests. Sending: Added to a 500 Internal Server Error in response to re-INVITE request crossing with ACK (200 OK). Contains an integer value indicating the Re-INVITE retry interim time.
Route	[4] 20.34 [63] 8.1 [64] [65]	In case the generic SIP port is used over the ISC interface the top Route header in initial requests must contain the route parameter 'as=' specifying the AS name. The SCC AS now handles the 'initialSelection' URL parameter in the top Route header for INVITE or REGISTER messages and applies SRVCC registration procedure for registered SC UE including SIP MESSAGE towards ATCF
Rseq	[6] 7.1	
Server	[4] 20.35	MTAS can pass through the received value or by configuration insert a Server header in all SIP Responses, with information about the MTAS version.
Session-Expires	[1] 4	
Session-Id	[34]	Used for 3PTY service to identify sessions to be used in 3PTY. MTAS adds the Session-Id header if it is not present in the initial INVITE Request and in Responses.
Subject	[4] 20.36	
Subscription-State	[7] 8.2.3	
Supported	[4] 20.37	
Target-Dialog	[49]	SRVCC Release-10 introduces the Target-Dialog header in SCC AS. Target-Dialog header field is used in INVITE(ATU-STI) request coming from ATCF in order to identify the old PS dialog to be transferred.
To	[4] 20.39	To header may be modified between received and sent initial requests. MTAS generates a new to-tag for each dialog created from an initial request. The To header, including to-tag, is mapped between received and sent responses and subsequent requests in a dialog.
Unsupported	[4] 20.40	
User-Agent	[4] 20.41	MTAS can pass through the received value or by configuration insert a User-Agent header in all SIP Requests, with information about the MTAS version.
Via	[4] 20.42	MTAS support [59] RFC 7339 SIP Overload Control reporting role by adding oc parameters (oc/oc-algo/oc-validity/oc-seq) in topmost Via header.
Warning	[4] 20.43	Service specific Warning headers maybe included in reject responses.
X-AUT	See 5.2.6	Used by the Japanese Charging service for ICBS data.
X-Carrier-Info	See 5.2.7	Used by the Japanese Charging service for ICBS data.
X-CHGDelay	See 5.2.8	Carriers FCH data but it is only transported transparently through MTAS and the Japanese Charging service.
X-CHGInfo	See 5.2.9	Used by the Japanese Charging service for FCH data.

Header	Ref	MTAS Usage
X-GENERIC_NUM	See 5.2.11	Can be used on the diverted leg for indicating a BCD call
X-VMS-Request	See 5.2.10	Used in conjunction with Voice Mail Service.

4.5 Bodies

This section specifies the message bodies that are used by MTAS.

Any other message bodies that are received are passed on transparently when the MTAS node is working in B2BUA mode, and are ignored when the MTAS node is working in UAS or UAC mode.

4.5.1 Properties

All documents in the body of a SIP method /response can have the following properties

Table 9 - Generic Properties of SIP Documents

Property	Purpose/ Possible values
Content Type ³ (MIME subtype)	This will map to either a MIME type or MIME sub-type and will uniquely identify the document type
Direction	Possible Values <ul style="list-style-type: none"> • Generates only • Receives only • Generates & Receives • Forward
Content Disposition ^{1,3}	Possible values <ul style="list-style-type: none"> • Render • Session • Info-package
Handling (by the recipient) ²	Possible Values <ul style="list-style-type: none"> • Optional • Required (default)
Character Set Encoding	As UTF-8 is backwards compatible with ASCII, it is the usual form of encoding.
Version	Possible values <ul style="list-style-type: none"> • Versioning not supported • All versions • <i>List defined subset</i>

Notes

1: The default Content-disposition should be defined for each specific body type. If no default is defined for a body then 'render' is used as the default. The default Content-disposition is used when no Content-disposition header is included for the body in the SIP message.

2: The default value for the handling parameter is 'required'. The default value is used if there is no Content-disposition header is included for the body or if there is no handling parameter included in the Content-disposition header.

3. Content-type is always included in the headers section of a SIP Request which includes a body part. Content-disposition is optional. If the Content-type is 'multipart/mixed' then separate Content-type and optional Content-disposition headers are included in the body section of the message for each separate body included, see section 4.5.2.2.

4.5.1.1 Content-Type

Each document type supported over this interface is defined to have a unique MIME type (or a MIME sub-type) which equates to the Content-Type.

Content-Type is a header field that defines the body. See section 20.15 of [4] for further details.

XML Content Types were defined in RFC 3023 [20].

Each XML Content-Type should be registered with IANA in accordance with RFC 3688 [28].

4.5.1.2 Version

Table 10 provides properties additional to those shown in Table 9 which are applicable to xml bodies only.

Table 10 – Additional properties that apply to XML only

Property	Possible or typical values	
Handling of schema version	Document Header	Possible Values <ul style="list-style-type: none"> • Versioning not supported • Version always present • If absent, assume version [version no.]
	Content-Type (as m-params)	Possible Values <ul style="list-style-type: none"> • Versioning not supported • Version always present • If absent, assume version [version no.]

Individual document types and their properties are defined in the following sub-sections.

4.5.2 MIME

The MIME mechanism is used to carry documents in the body of SIP methods. Separate behavior is defined for a SIP body carrying one document and for a SIP body carrying multiple documents

4.5.2.1 Single document body

This is the simple case with a single content-type definition.

An example is shown below

```
Content-Type: application/vnd.etsi.aoc+xml; sv="2"
Content-Disposition: render
Content-Length: 246
<CRLF>
<Body goes here>
```

4.5.2.2 Multi-part bodies

This case requires

- i. An overall content-type definition of “multipart/mixed” to indicate the body contains multiple documents of different types. (RFCs 2045 & 2046 [42] [43])
- ii. A content-type definition per document
- iii. An optional content-disposition definition per document
- iv. A boundary which is a unique string that marks the beginning and end of each content-type definition.

An example is shown below

```
Content-Type: multipart/mixed; boundary="959595"
Content-Length: 896
--959595
Content-Type: application/vnd.3gpp.cw+xml;sv="1"
Content-Disposition: render
<CRLF>
<Body goes here>
--959595
Content-Type: application/vnd.etsi.aoc+xml; sv="2"
Content-Disposition: render
<CRLF>
<Body goes here>
--959595--
```

4.5.3 SDP

An extension to the body of a SIP method with SDP [2] has the following attributes:

Table 11 - SDP specific properties

Session Description Protocol		
Content Type (MIME subtype)		application/sdp
Direction		Generates and receives
Content Disposition	Generates	session
	Receives	If not included default disposition of 'session' used
Handling	Generates	Required
	Receives	If not included default handling of 'required' used
Character Set Encoding		UTF-8
Version		versions not supported

4.5.3.1 Content

The syntax of the SDP body is as defined in RFC 4566, [2].

4.5.4 Advice-of-Charge to the User

Table 12 - AOC Specific Properties

Advice-of-Charge	
Content Type (MIME subtype)	application/vnd.etsi.aoc+xml
Direction	Generates only
Content Disposition	Render
Handling (by the recipient)	not included default value implied (required)
Character Set Encoding	UTF-8
Version	1.0 (default) or 2

Handling of schema version	Document Header	Version always present
	Content-Type (as m-params)	Version always present

4.5.4.1 Schema

The XML schema for version 1.0 is defined in 3GPP TS 24.647 [22].

The XML schema for version 2 is defined in TS 183 043 [29].

The version of the body sent from the MTAS will depend upon the sv (schemaversion) parameter for the AOC service in the received Accept header.

4.5.5 Communication Waiting

Table 13 - Communication Waiting specific properties

Communication Waiting		
Content Type (MIME subtype)	application/vnd.3gpp.cw+xml	
Direction	Generates only	
Content Disposition	Included: value – render	
Handling (by the recipient)	Included: value – optional	
Character Set Encoding	UTF-8	
Version	Versioning not supported	
Handling of schema version	Document Header	Versioning not supported
	Content-Type (as m-params)	Versioning not supported

4.5.5.1 Schema

The XML schema is defined in 3GPP TS 24.615 [24].

4.5.6 Dial Tone Management Notification

Table 14 - Dial Tone Management specific properties

DTM		
Content Type (MIME subtype) (Note 1)	text/xml or application/simservs+xml	
Direction	Generates only	
Content Disposition	not included default value implied (render)	
Handling	not included default value implied (required)	
Character Set Encoding	application/xml <i>Note: This definition maps to UTF-8</i>	
Version	Versioning not supported	
Handling of schema version	Document Header	Versioning not supported
	Content-Type (as m-params)	Versioning not supported

Note 1: The MTAS may be configured to send DTM notification with Content Type of text/xml or application/simservs+xml.

4.5.6.1 Schema

MTAS implemented the dial-tone-management global element as defined in the schema as specified Appendix A of TS 183 043 [29].

4.5.7 Conference State - NOTIFY

An extension to the body of a SIP NOTIFY method by the Conference service (both ad-hoc and scheduled type) has the following attributes:

Table 15 - Conference specific properties

Conference State - NOTIFY		
Content Type (MIME subtype)		application/conference-info+xml
Direction	User MTAS	Forwards
	Conference MTAS	Generates only
Content Disposition		not included default value implied (render)
Handling		not included default value implied (required)
Character Set Encoding		UTF-8
Version		Versioning not supported
Handling of schema version	Document Header	Versioning not supported
	Content-Type (as m-params)	Versioning not supported

4.5.7.1 Schema

The XML schema is defined in RFC 4575 [31].

There is an exception in case of Scheduled Conference, where the 'entity' attribute of 'conference-info' element contains the XCON identity of the conference room without namespace nomination instead of the SIP URI of the conference session as defined in RFC 4575.

4.5.8 Real-time Transfer of Tariff Information

Table 16 - Properties of RTTI document

RTTI		
Content Type (MIME subtype)	application/vnd.etsi.sci+xml	
Direction	Receives only is the usual mode but will forward any document of this type if the SIP session has charging not active	
Content Disposition	not defined default value implied (render)	
Handling	not included default value implied (required)	
Character Set Encoding	application/xml <i>Note: This definition maps to UTF-8</i>	
Version	Not Supported	
Handling of schema version	Document Header	Versioning not supported
	Content-Type (as m-params)	Versioning not supported

4.5.8.1 Schema

The XML schema is defined in Annex C of TS 29.658 [23].

4.5.9 URI List – Conference/3PTY creation

Table 17 - Properties of URI list document

URI list		
Content Type (MIME subtype)	application/resource-lists+xml	
Direction	Receives only	
Content Disposition	recipient-list	
Handling	not included default value implied (required)	
Character Set Encoding	application/xml <i>Note: This definition maps to UTF-8</i>	
Version	Not Supported	
Handling of schema version	Document Header	Versioning not supported
	Content-Type (as m-params)	Versioning not supported

4.5.9.1 Schema

The URI list body uses two schemas:

- resource-lists which is defined in RFC 4826 [35]
- copy-control which is defined in RFC 5364 [36]

4.5.9.2 Service Level Validation

- Multiple URI list bodies are not allowed.
- The maximum number of entries in the URI list is 31.
(The 3PTY service accepts only 2 entries).
- The URI in each list entry may contain header parameters like Call-Id, From, To and/or Session-id etc.
- The 'Session-id' header parameter is required by the 3PTY service.

4.5.10 Communication Diversion – Presence Request

Table 18 - Communication Diversion Presence Request specific properties

CDIV – Presence Request		
Content Type (MIME subtype)	application/simple-filter+xml	
Direction	Generates only	
Content Disposition	not defined default value implied (render)	
Handling	not included default value implied (required)	
Character Set Encoding	application/xml <i>Note: This definition maps to UTF-8</i>	
Version	Not Supported	
Handling of schema version	Document Header	Versioning not supported
	Content-Type (as m-params)	Versioning not supported

4.5.10.1 Schema

The schema is defined in RFC 4661 [38].

MTAS specifically requests responses with person and activities as defined in RFC 4480 [39].

4.5.11 Communication Diversion – Presence Response

Table 19 - Communication Diversion Presence Response specific properties

CDIV Presence Response		
Content Type (MIME subtype)	application/pidf+xml	
Direction	Receives only	
Content Disposition	If not included default value implied (render)	
Handling	If not included default value implied (required)	
Character Set Encoding	application/xml <i>Note: This definition maps to UTF-8</i>	
Version	Not Supported	
Handling of schema version	Document Header	Versioning not supported
	Content-Type (as m-params)	Versioning not supported

4.5.11.1 Schema

The schema is defined in RFC 3863 [37].

4.5.12 Communication Diversion – MESSAGE

Table 20 - Communication Diversion Message specific properties

CDIV Message	
Content Type (MIME subtype)	text/plain
Direction	Generates only
Content Disposition	not included default value implied (render)
Handling	not included default value implied (required)
Character Set Encoding	UTF-8
Version	Versioning not supported

4.5.12.1 Content

The body included in the MESSAGE Request is a plain text message configured by the network operator.

The syntax is as defined in RFCs 2045 & 2046 , [42] [43].

4.5.13 NOTIFY Body

Table 21 - NOTIFY specific properties

NOTIFY Body		
Content Type (MIME subtype)		message/sipfrag
Direction		Generates and receives
Content Disposition	Generated	not included default value implied (render)
	Received	If not included default disposition of 'render' used
Handling	Generated	not included default value implied
	Received	If not included default handling of 'required' used
Character Set Encoding		UTF-8

Version	Default
---------	---------

4.5.13.1 Content

The body contains a SIP fragment giving details of the NOTIFY; e.g. SIP 2.0/100 Trying.

The syntax of the sipfrag body is as defined in RFC 3420, [27].

4.5.14 Communication Completion – NOTIFY

Table 22 - Communication Completion specific properties

CC	
Content Type (MIME subtype)	application/call-completion
Direction	Generates and receives
Content Disposition	not included default value implied (render)
Handling	not included default value implied (required)
Character Set Encoding	UTF-8
Version	Versioning not supported

4.5.14.1 Content

- The formal syntax is provided in [33].

4.5.15 Communication Completion – PUBLISH

Table 23 - Communication Completion specific properties

CC	
Content Type (MIME subtype)	application/pdf+xml
Direction	Generates and receives
Content Disposition	not included default value implied (render)
Handling	not included default value implied (required)
Character Set Encoding	UTF-8
Version	Versioning not supported

4.5.15.1 Schema

The schema is defined in RFC 3863 [37].

4.5.16 SRVCC – MESSAGE

Table 24 - Srvcc Message specific properties

Srvcc Message	
Content Type (MIME subtype)	text/plain
Direction	Generates only
Content Disposition	not included default value implied (render)
Handling	not included default value implied (required)
Character Set Encoding	UTF-8
Version	Versioning not supported

4.5.16.1 Content

SCC AS sends SIP MESSAGE to ATCF on receipt of third party registration. The MESSAGE request contains the ATCF-Management-URI in Request-URI and message body of “application/vnd.3gpp.SRVCC-info+xml” type. If 3rd Party REGISTER does not include Path header or there is no ATCF-Path-URI in Path header then MESSAGE request IS NOT sent. The XML Schema of “vnd.3gpp.SRVCC-info+xml” is defined in 3GPP TS 24.237 [48].

4.5.17 Third party registration with multipart message/sip body

Table 25 – REGISTER specific properties

NOTIFY Body	
Content Type (MIME subtype)	message/sip
Direction	Receives
Character Set Encoding	UTF-8
Version	Default

4.5.17.1 Content

The body of third party registration is a multipart body containing the message/sip REGISTER request from UE and 200 OK response. The inclusion of the register request and response is configured in the iFC for an AS.

This is required for SCC AS, ST AS and for MMTel AS when standalone and FCD service is intended to be used to distribute call to ICS user's devices. In both cases the registered contacts information available in 200 OK register response is needed.

If the node receiving a 3rd party registration is a SSC AS then the third party register may include a Path header [47]. This Path header is used to determine if the SRVCC Release-9 or Release-10 procedure will be applied on the contact.

The syntax of the message/sip body is as defined in RFC 3261, [4].

4.5.18 SRVCC Alerting - INFO

Table 26 – state-and-event-info specific properties

INFO body	
Content Type (MIME subtype)	Application/vnd.3gpp.stat-and-event-info+xml
Direction	Generates & Receives
Content Disposition	Info-package
Handling	not included, default value implied (required)
Character Set Encoding	UTF-8
Version	Versioning not supported

4.5.18.1 Content

SCC AS sends/receives SIP INFO on SRVCC handover for alerting call. The INFO request contains the Info-Package header and message body of "application/vnd.3gpp.state-and-event-info+xml" type. The XML Schema of "vnd.3gpp.state-and-event-info+xml" is defined in 3GPP TS 24.237 [48].

4.5.19 Flexible Communication Distribution – Presence Request

This is the same as for Communication Diversion. See 4.5.10.

4.5.20 Flexible Communication Distribution – Presence Response

This is the same as for Communication Diversion. See 4.5.11.

4.5.21 Flexible Communication Distribution – MESSAGE

This is the same as for Communication Diversion. See 4.5.12.

4.5.22 Closed User Group – INVITE

Table 27 - CUG Specific Properties

CUG indication		
Content Type (MIME subtype)	application/vnd.etsi.cug+xml	
Direction	Forwards	
Content Disposition	If not included default disposition of 'render' used	
Handling (by the recipient)	If not included default handling of 'required' used	
Character Set Encoding	UTF-8	
Version	Versioning not supported	
Handling of schema version	Document Header	Versioning not supported
	Content-Type (as m-params)	Versioning not supported

4.5.22.1 Content

This body defines the Closed User Group Interlock Code of the subscriber. The schema to be used is defined in [52].

4.5.23 Registration Event Response - NOTIFY

Table 28 – SCC AS/ST AS Registration Event Response specific properties

Registration Event Response		
Content Type (MIME subtype)	application/reginfo+xml	
Direction	Receives only	
Content Disposition	If not included default value implied (render)	
Handling	If not included default value implied (required)	
Character Set Encoding	application/xml <i>Note: This definition maps to UTF-8</i>	
Version	Not Supported	
Handling of schema version	Document Header	Versioning not supported
	Content-Type (as m-params)	Versioning not supported

4.5.23.1 Schema

The XML schema of the “reg” event package, defined in RFC 3680 [53] is supported by the MMTEL AS, SCC AS and ST AS.

The MMTEL AS and SCC AS uses the Ericsson specific extension to the “reg” event package in order to support the MMTEL AS and SCC AS restart or failover scenarios.

The ATCF feature-caps-header pvni and impi extension of the schema is only supported by the Ericsson S-CSCF since the extension is not standardized but Ericsson specific.

The SCC AS uses the ATCF feature-caps-header and impi extension in order to support the rel10 SR-VCC handover after a SCC AS restart or failover.

The namespaces of the Ericsson specific reg event xml extensions are:

com:ericsson:schema:xml:ims:feature-caps-header-regevent-extension

com:ericsson:schema:xml:ims:impi-regevent-extension

com:ericsson:schema:xml:ims:pvni-regevent-extension

com:ericsson:schema:xml:ims:pani-regevent-extension

The namespace reference is included in all of the Ericsson specific reg event xml extension's xmlns attribute.

The ATCF management information is stored in the "feature-caps-header" element in the notification xml. The value of this element is equal with the Feature-caps header value in a 3rd Party Registration according to [54].

The "feature-caps-header" string is escaped except the " (quote) sign.

The "impi" element is introduced to transfer the IMPI of a contact within the reg event xml.

If the "feature-caps-header" element is included for a contact, the "impi" element has to be included also.

The P-Visited-Network-ID information is stored in the "pvni" element in the notification xml extension. The value of this element is equal with the P-Visited-Network-ID header value in a 3rd Party Registration according to [48]

4.5.24 Dialog State - NOTIFY

An extension to the body of a SIP NOTIFY method by the Dialog Event Notifier service has the following attributes:

Table 29 – Dialog State specific properties

Dialog State - NOTIFY	
Content Type (MIME subtype)	application/dialog-info+xml
Direction	Towards served user
Content Disposition	not included default value implied (render)
Handling	not included default value implied (required)
Character Set Encoding	UTF-8
Version	1.0
Handling of schema version	Versioning not supported

4.5.24.1 Schema

The XML schema is defined in RFC 4235 [62].

5 Formal Syntax or Schema

5.1 Requests and Responses

The syntax for SIP Requests and Responses as defined in RFC 3261, [4]. The syntax of individual Requests types are as defined in the documents referenced in the 'Ref' column of Table 3.

Vendor specific extensions are described below.

5.1.1 NOTIFY Request

5.1.1.1 User Location Indication

The CSCF may send user location info to MTAS in an unsolicited out-of-dialog SIP NOTIFY with the P-Access-Network-Info (PANI) header set accordingly. In order to make it possible for MTAS to correlate the NOTIFY to an active session, in the same NOTIFY the Charging Info Extension to SIP EVENT Package is used (see 5.2.1.1).

5.1.1.2 Emergency Call Start/Stop Indication

The CSCF may send an unsolicited out-of-dialog NOTIFY message using the emergency call extension to SIP EVENT package (see 5.2.1.2) to indicate the start or stop of an emergency call.

5.2 Headers

The syntax of SIP headers is as defined in RFC 3261 [4], with the syntax of specific headers as defined in the documents referenced in the 'Ref' column of Table 8.

Additional vendor specific extensions are described below.

5.2.1 Event Header

5.2.1.1 Charging Info Extension

The Charging Info extension to the SIP Event package is used in relation with the unsolicited out-of-dialog SIP NOTIFY sent by the CSCF to pass on access network info to MTAS (see 5.1.1.1).

Example:

Event: charging-info; icid=<value>; call-id=<value>; from-tag=<value>; to-tag=<value>

MTAS will use the value of the icid parameter to find the corresponding call session.

5.2.1.2 Emergency Call Extension

The Emergency Call extension to SIP EVENT package is used by the CSCF to inform MTAS through an unsolicited SIP NOTIFY (see 5.1.1.2) that a subscriber has started or ended an emergency call.

Example 1:

Event: emergencyCall; start

Example 2:

5.2.2 Event: emergencyCall; stopP-Area-Info Header

The P-Area-Info header is used by MTAS to transfer Originating Charge Area (OCA) in SIP INVITE messages from the originating network to the terminating network and also to transfer Terminating Charge Area (TCA) in SIP responses from the terminating network to the originating network. A Charge Area (CA), is a market-specific 5-digit code indicating geographical location of the subscriber; it can either be originating or terminating CA. The header presented as plain text.

Syntax is based on ABNF grammar:

P-Area-Info = "P-Area-Info" HCOLON ca [SEMI area-info][SEMI area-id][SEMI area-type]

ca = "ca" EQUAL (token)

area-info = "area-info" EQUAL (quoted-string)

area-id = "area-id" EQUAL (token)

area-type = "area-type" EQUAL "ca"/"ma"

Only CA is mandatory, area-info and area-id shall not be set by MTAS. When area-type is not set, 'ca' is assumed and there is no need to set it.

Example:

P-Area-Info: ca=32000

5.2.3 P-Charging-Vector

The syntax for the P-Charging-Vector header is described in RFC3455 [10] as follows:

P-Charging-Vector = "P-Charging-Vector" HCOLON icid-value*(SEMI charge-params)
charge-params = icid-gen-addr / orig-ioi / term-ioi / generic-param

Example:

P-Charging-Vector: icid-value=1234567890;icid-generated-at=192.0.0.1;orig-ioi=cscfAloi.com;ttc-charging-params="cari=iecind-0,cat-olec,code-2051;cai=98800;auc=fixed_1-2;fci=nii-nat,oa-isdn"

5.2.4 P-Charging-Vector.ttc-charging-params

ttc-charging-params is an extension (generic-param) used by MTAS to transfer Interconnection Charge Billing System (ICBS) and Flexible Charging (FCH) parameters, which are defined by the Japanese standardization group Telecommunication Technology Committee (TTC). More information regarding this can be obtained from the reference [55].

ICBS parameters consist of "cari", "cai", "auc", "fci", "bci", and "contractor", while FCH parameters consist of "cid", "ci", and "cit" in below grammar. MTAS does not support two instances for "cari", "cai", "fci", "bci", and "contractor" parameters although it doesn't violate the syntax below. On the other hand, "auc", "cid", "cit", and "ci" parameters can be handled with two instances.

The syntax is based on ABNF grammar:

ttc-charging-params = "ttc-charging-params" EQUAL LDQUOTE
 (ttc_icbs_parameter / ttc_charging_parameter) *(SEMI ttc_icbs_parameter /
 ttc_charging_parameter) RDQUOTE

ttc_icbs_parameter = cari / cai / (auc [SEMI auc])

cari = "cari" EQUAL iec_ind 1*(COMMA carrier_category_and_info)

iec_ind = "iecind" "-" ("0" / "1" / "2" / "3")

carrier_category_and_info = carrier_category 1*(COMMA carrier_info)

carrier_category = "cat" "-" ("scpc" / "olec" / "tlec" / "ciec" / "iec")

Carrier_Info = poi_hierachy / poi_cai / cid_code

poi_hierachy = "poi_hi" "-" entry_poi_hierarchy "-" exit_poi_hierarchy

entry_poi_hierarchy = "0" / "1" / "2"

exit_poi_hierarchy = "0" / "1" / "2"

poi_cai = "poi_cai" "-" 5DIGIT

cid_code = "code" "-" 4DIGIT

cai = "cai" EQUAL 5DIGIT

auc = "auc" EQUAL (type_1_auc_for_fixed / type_1_auc_for_mobile /
type_2_auc_for_mobile)

type_1_auc_for_fixed = "fixed_1-1" / "fixed_1-2"

type_1_auc_for_mobile = "mobile_1-1" / "mobile_1-2" / "mobile_1-3" /
"mobile_1-4" / "mobile_1-5"

type_2_auc_for_mobile = "mobile_2-1" / "mobile_2-2" / "mobile_2-3" /
"mobile_2-4" / "mobile_2-5" / "mobile_2-6" / "mobile_2-7" / "mobile_2-8" /
"mobile_2-9"

ttc_charging_parameter = (cid [SEMI cid]) / cit / ci / fci / bci / contractor

cid = "cid" EQUAL ("charge_rate" / "tca")

cit = "cit" EQUAL ("advanced" / "charge_rate")

ci = "ci" EQUAL (charge_rate_for_ordinary_subscriber /
charge_rate_for_payphone / charge_rate_for_pink_phone /
charge_rate_for_non_flex)

charge_rate_for_ordinary_subscriber = utp COMMA cric COMMA iu COMMA dcr
COMMA ecr COMMA ncr COMMA scr

utp = "utp" "-" ("hundred" / "ten" / "no_ind")

cric = "cric" "-" ("payphone" / "ordinary" / "no_flex")

iu = "iu" "-" 2DIGIT

dcr = "dcr" "-" 3DIGIT

ecr = "ecr" "-" 3DIGIT

ncr = "ncr" "-" 3DIGIT

scr = "scr" "-" 3DIGIT

charge_rate_for_payphone = utp COMMA cric COMMA iu COMMA dcr COMMA
ecr COMMA ncr COMMA scr

utp = "utp" "-" ("hundred" / ten" / "no_ind")

cric = "cric" "-" ("payphone" / "ordinary" / "no_flex")

iu = "iu" "-" 2DIGIT

dcr = "dcr" "-" 3DIGIT

ecr = "ecr" "-" 3DIGIT

ncr = "ncr" "-" 3DIGIT

scr = "scr" "-" 3DIGIT

charge_rate_for_pink_phone = charge_rate_for_ordinary_subscriber COMMA
charge_rate_for_payphone

charge_rate_for_non_flex = utp COMMA cric

utp = "utp" "-" ("hundred" / ten" / "no_ind")

cric = "cric" "-" ("payphone" / "ordinary" / "no_flex")

fci = "fci" EQUAL nii COMMA oa

nii = "nii" "-" ("nat" / "int")

oa = "oa" "-" ("non_isdn" / "isdn")

bci = "bci" EQUAL chi COMMA cdpci COMMA ta

chi = "chi" "-" ("no_ind" / "no_charge" / "charge" / "spare")

cdpci = "cdpci" "-" ("no_ind" / "ordinary" / "payphone" /
"spare")

ta = "ta" "-" ("non_isdn" / "isdn")

contractor="contractor" EQUAL telephone-url

Example:

ttc-charging-params="cari=iecind-0,cat-tlec,code-0902;cai=97800;auc=mobile_1-
1;auc=mobile_2-8;bci=chi-no_ind,cdpci-no_ind,ta-
isdn;contractor="+8613810018638"

5.2.5 P-com.TimeZone Header

The P-com.TimeZone header is included by CSCF in SIP messages sent during call establishment based on the time zone info received via the Rx interface. The header is applicable for both SIP requests and responses. For example it can be sent in ACK/PRACK on originating side and in 200 OK on terminating side.

Syntax:

P-com.TimeZone = "P-com.TimeZone" HCOLON timezone

timezone = quoted-string

The P-com.TimeZone header indicates the time zone where the MS/UE currently resides.

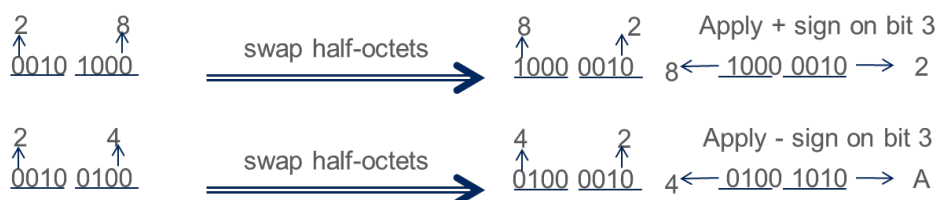
Format of the header:

Bit	7	6	5	4	3	2	1	0
Octet 1	Time Zone							
Octet 2	Spare						Daylight saving time	

The Time Zone octet indicates the difference, expressed in quarters of an hour (i.e. 15 minutes), between the local time and GMT. Range of value is from GMT-12:00 until GMT+13:00.

Time zone field is *semi-octet represented*. Semi-octet means 1 octet has two half octets and each half-octet represents a decimal digit (BCD, Binary-Coded Decimal).

In the first of the two semi octets, the first bit (bit 3 of the time zone octet) represents the algebraic sign of this difference (0: positive, 1: negative). For example '82'H means GMT+07:00 and '4a'H means GMT-06:00.



The two LSB of second octet encodes the daylight saving time:

Bits		Meaning
0	0	No adjustment for Daylight Saving Time
0	1	+1 hour adjustment for Daylight Saving Time
1	0	+2 hours adjustment for Daylight Saving Time
1	1	Reserved

Bits marked 'Spare' should be set to '0' by the sending node.

5.2.6 P-com.PrivateUserID Header

The P-com.PrivateUserID header is included by CSCF in SIP messages sent during call establishment and it contains the IMS Private User ID (IMPI) of the subscriber. The header is present only if the user is registered. The header is applicable for both SIP requests and responses. For example it can be sent in INVITE on originating side and in 200 OK on terminating side.

Example:

P-com.PrivateUserID: 16309790000@vzims.com

5.2.7 P-com.User-Equipment-Info Header

The P-com.User-Equipment-Info (UEI) header is included by CSCF in SIP messages sent during call establishment based on user equipment info received via the Rx interface. The header is applicable for both SIP requests and responses. For example it can be sent in INVITE on originating side and in 200 OK on terminating side.

Syntax:

P-com.User-Equipment-Info = "P-com.User-Equipment-Info" HCOLON uei-type
";" uei-value

uei-type = "uei-type" EQUAL [0 - 3]

uei-value = "uei-value" EQUAL quote string

The possible enumerations for the equipment type (uei-type) are as follows:

uei-type	code
IMEISV	0
MAC	1
EUI64	2
MODIFIED_EUI64	3

Example:

P-com.User-Equipment-Info: uei-type=1; uei-value="4488951064550101"

5.2.8 X-AUT Header

The X-AUT header is used by MTAS to transfer Originating Additional User Category (OAUC) in SIP INVITE messages from the originating network to the terminating network and also to transfer Terminating Additional User Category (TAUC) in SIP responses from terminating network to originating network. The AUC specifies the user or mobile service characteristics in more detail.

The X-AUT header holds either the originating or terminating Additional User Category (AUC) and each header contains two categories, for example 'Cellular telephone service' and 'IMT-2000'. The value of the header is interpreted bit-wise and encoded in hexadecimal.

Syntax is based on ABNF grammar:

X-AUT = "X-AUT" HCOLON 1*(2HEXDIG)

Structure of the header content:Possible values are as follows:

8 (byte bit)	7	6	5	4	3	2	1	Octet number
Type of additional user/service information (1)								1st
Additional user/service information (1)								2nd
.								
..								
Type of additional user/service information (n)								2n-1
Additional user/service information (n)								2n

Values for type of additional user/service information:

- 00000000 Spare
- 00000001-10000000 Reserved for network specific use

10000001-11111010	Spare
11111011	Type 3 of additional mobile service information
11111100	Type 2 of additional mobile service information
11111101	Type 1 of additional mobile service information
11111110	Type 1 of additional fixed user information
11111111	Spare

Values for type 1 of additional fixed user information:

00000000	Spare
00000001	Train payphone
00000010	Pink (non NTT payphone)
00000011-11111111	Spare

Values for type 1 of additional mobile service information:

00000000	Spare
00000001	Cellular telephone service
00000010	Maritime telephone service
00000011	Airplane telephone service
00000100	Paging service
00000101	PHS service
00000110-11111111	Spare

Values for type 2 of additional mobile service information:

00000000	Spare
00000001	HiCap method (analog)
00000010	N JTACS
00000011	PDC 800 MHz
00000100	PDC 1500 MHz
00000101	N STAR satellite
00000110	cdmaOne 800 MHz

00000111	Iridium satellite
00001000	IMT 2000
00001001	PHS (fixed network dependent)
00001010-11111111	Spare

Values for type 3 of additional mobile service information:

00000000- 11111111 Reserved for network specific use (eg. for Charging Plan)

Example:

X-AUT: fd01fc06

According to Japanese ISUP inter-carrier interface **Error! Reference source not found.**, the above example means:

11111101 = Type 1 of additional mobile service information (fd)

00000001 = Cellular telephone service (01)

11111100 = Type 2 of additional mobile service information (fc)

00000110 = cdmaOne 800 MHz (06)

5.2.9

X-Carrier-Info Header

The X-Carrier-Info header is used by MTAS to transfer carrier information of different categories between the originating and terminating networks. Originating Local Exchange Carrier (OLEC) is the carrier from where the call originates. OLEC information is transferred from the originating network to the terminating network in SIP INVITE. OLEC information contains carrier identification code and possibly POI (Point Of Interfaces)-hierarchy.

Terminating Local Exchange Carrier (TLEC) is the carrier from where the call terminates. TLEC information is transferred from the terminating network to the originating network in SIP responses. TLEC information contains carrier identification code and possibly POI-hierarchy.

Chosen Inter Exchange Carrier (CIEC) is the carrier which is chosen by the calling subscriber or the OLEC to be a transit carrier. CIEC information consists of carrier identification code and possibly POI-CA (Charge Area) information and POI-hierarchy. CIEC information is transferred from the chosen carrier to the terminating carrier and it can also be transferred from the chosen carrier in backward direction (from terminating network to originating network) if Inter Exchange Carrier (IEC) indicator indicates so. The IEC indicator is also included in this header.

IEC is a transit carrier which is interconnected but is not a chosen carrier. IEC information consists of carrier identification code and possibly POI-CA information and POI-hierarchy. IEC information is transferred from the IEC in forward (originating network to terminating network) or in backward direction, or both if the IEC indicator mentioned above indicates so.

Service Control Point Carrier (SCPC) is the carrier which operates the Service Control Point (SCP). SCPC information consists of carrier identification code. SCPC information can be transferred in forward or in backward direction, or both.

CIEC, IEC, or SCPC, information or other network-specific carrier categories are not generated by MTAS but they can be received and processed by MTAS. POI-hierarchy and POI-CA are not determined and set by MTAS only the carrier identification code. The value of the header is interpreted bit-wise and encoded in hexadecimal like.

Syntax is based on ABNF grammar:

X-Carrier-Info = "X-Carrier-Info" HCOLON 1*(2HEXDIG)

The table below shows the maximum number of different carrier information that X-Carrier-Info header can hold:

Carrier category	Maximum number of occurrences in the X-Carrier-Info header.
Original SCP Carrier Information	1
Terminal SCP Carrier Information	1
SCP Carrier (SCPC) Information	1
Originating Carrier (OLEC)	1
Terminating Carrier (TLEC)	1
Chosen Inter-Exchange Carrier (CIEC) Information	1
Transit Carrier (IEC) Information	6

Maximum size of this header is 146 bytes.

Structure of the header content:

Structure of the header content								
8 (bit)	7	6	5	4	3	2	1	Octet number
spare						IEC Indicator		1st
Category of carrier 1								2nd
Length of carrier 1								3rd
Carrier 1 information / octet 1								4th

.	
Carrier 1 information / octet n	3+n
.	
Category of carrier x	
Length of carrier x information	
Carrier x information / octet 1	
.	
Carrier x information / octet n	

Different IEC indicator values:

00	No transfer
01	Transfer in forward direction
10	Transfer in backward direction
11	Transfer in both directions

Different Category of carriers:

00000000	Spare
00000001 to 100000000	Reserved for network specific use
10000001 to 11111001	Spare
11111010	SCPC (Service Control Point Carrier)
11111011	OLEC (Originating Local Exchange Carrier)
11111100	TLEC (Terminating Local Exchange Carrier)

11111101 CIEC (Chosen Inter|Exchange Carrier)

11111110 IEC (Inter|Exchange Carrier)

11111111 Spare

Structure of Carrier x information:

8 (bit)	7	6	5	4	3	2	1	Octet number
Type of carrier x information 1								1st
Length of carrier x information 1								2nd
Carrier x information 1 / octet 1								3rd
.								
.								
.								
Carrier x information 1 / octet n								2 + n
.								
.								
.								
Type of carrier x information y								
Length of carrier x information y								
Carrier x information y / octet 1								
.								
.								
.								
Carrier x information y / octet n								

Types of carrier x information 1:

00000000 Spare

00000001 to 100000000 Reserved for network specific use

10000001 to 11111011 Spare

11111100	POI Hierarchy Information
11111101	POI/CA Information (Charge Area)
11111110	Carrier Identification (ID) code
11111111	Spare

Structure of Carrier identification (ID) code:

8 (bit)	7	6	5	4	3	2	1	Octet number
Odd/even	Spare							1st
2 nd ID code digit				1 st ID code digit				2nd
Filler (if necessary)				n th ID code digit Currently length of ID code is fixed to 4 digits.				m

Odd/Even:

0	Even number of ID code digits
1	Odd number of ID code digits

Address signals (ID code digits):

0000	Digit 0
0001	Digit 1
0010	Digit 2
0011	Digit 3
0100	Digit 4
0101	Digit 5
0110	Digit 6

0111	Digit 7
1000	Digit 8
1001	Digit 9
1010 to 1111	Spare

Example:

X-Carrier-Info: 02fb05fe03000560

The first part **02fb05fe03000560** of the above example means:

1st octet: Spare (0) and IEC -> 02 (transfer in backward direction)

2nd octet: Category of carrier 1 -> fb (OLEC)

3rd octet: Length of carrier 1 information -> 05 (5 octets)

The second part 02fb05**fe03000560** of the above example means:

4th octet: Type of carrier x information 1 -> fe (Carrier Identification (ID) Code)

5th octet: Length of carrier x information 1 -> 03 (3 octet)

The third part 02fb05fe03**000560** of the above example means:

Since carrier x information is Carrier Identification code.

6th octet: odd/even + spare -> 00 (even)

7th to 8th octet: ID code digits (4 digits) -> 0560

5.2.10 X-CHGDelay Header

The X-CHGDelay header is transported transparently through MTAS to pass the Charge Information Delay (CID) from the terminating Circuit Switch (CS) network to the originating IMS network. CID indicates that Charge Information (CI) or Terminating Charge Area (TCA) information, or both will be sent in backward direction in a later response message. The value of the header is interpreted bit-wise and encoded in hexadecimal. The content of this header is not generated by MTAS.

Syntax is based on ABNF grammar:

X-CHGDelay = "X-CHGDelay" HCOLON 2HEXDIG

Type of delayed charging information:

00000000	Spare
00000001 to 10000000	Reserved for network specific use
10000001 to 11111100	Spare
11111101	Charge rate transfer
11111110	Terminating charge area information
11111111	Spare

Example:

X-CHGDelay: fe

fe (terminating charge area information)

5.2.11 X-CHGInfo Header

The X-CHGInfo header is sent to MTAS to transfer Charge Information Type (CIT) and Charge Information (CI) from the terminating CS network to the originating IMS network. The value of the header is interpreted bit-wise and encoded in hexadecimal. The content of this header is not generated by MTAS.

The CIT is set to 'Charge rate transfer' in conjunction with CI when used for Flexible Charging (FCH) and set to 'Advanced charge rate transfer' when used by Telephone directory service (TDS) provided by the Public Switched Telephone Network (PSTN) network in Japan.

CI consists of Units per Time Period, Charge Rate Information Category, Initial Units, Daytime Charge Rate, Evening time Charge Rate, Night time Charge Rate and Spare Charge Rate. Syntax is based on ABNF grammar:

X-CHGInfo = "X-CHGInfo" HCOLON x-ci-kind x-ci-data [SEMI x-ci-record]

x-ci-kind = HEXDIG HEXDIG

x-ci-data = *(HEXDIG HEXDIG)

x-ci-record = "record"

CIT is represented by x-ci-kind, CI is represented by x-ci-data and CDR record is represented by x-ci-record.

Structure of header content:

:8 (byte bit)	7	6	5	4	3	2	1	Octet number
Charge information type (CIT) - (x-ci-kind)								1st
Charge Information (CI) - (x-ci-data)								2nd
								...
								2n

Values for CIT:

00000000 KDDI/public telephone for international direct distance dialling

00000001 International system/public telephone for international direct distance dialling

00000011 Advanced charge rate transfer (received when connected to Telephone directory service (TDS) or NTT104)

11111110 Charge rate transfer (flexible charging (FCH) rate transfer)

Structure of Charge Information Type (x-ci-kind):

8 (byte)	7	6	5	4	3	2	1	Octet number
Charge Information Type								1st

Charge Information Type:

00000000 Spare

00000001 Reserved

00000010 Reserved

00000011 Advanced Charge Rate Transfer (TDS service provided by NTT)

00000100 to 10000000 Reserved for network specific use

10000000 to 11111101 Spare

11111110 Charge rate transfer (flexible charging)

11111111 Spare **Structure of Charge Information (type 1) (x-ci-data):**

The coding of CI depends on the value of CIT.

Structure of Charge Information for CIT 'Charge rate transfer':

If CIT's value is "11111110 (Charging rate transfer)", then CI consists of Units per Time Period, Charge Rate Information Category, Initial Units, Daytime Charge Rate, Evening time Charge Rate, Night time Charge Rate and Spare Charge Rate. The structure of CI for this type is as follows:

8 (bit)	7	6	5	4	3	2	1	Octet number
Units per Time Period								1 st
Extens ion	Charging Rate Information Category							2 nd
Charging Rate Information Length								3 rd
(Initial Units) Expresses batch registration unit as IA5								M: 4 th
(10M + N) (Octets M,N)								N: 5 th
Expresses first charging interval (seconds) as IA5 ((100A + 10B +C)/2) seconds/unit charge (Octets A, B, C)								A: 6 th B: 7 th C: 8 th
Expresses second charging interval (seconds) as IA5 ((100D + 10E +F)/2) seconds/unit charge (Octets D, E, F)								D: 9 th E: 10 th F: 11 th
Expresses third charging interval (seconds) as IA5 ((100G + 10H +I)/2) seconds/unit charge (Octets G, H, I)								G: 12 th H: 13 th I: 14 th
Expresses fourth charging interval (seconds) as IA5 ((100J + 10K +L)/2) seconds/unit charge (Octets J, K, L)								J: 15 th K 16 th L: 18 th

Octets M to L are added to the CI parameter only if Charging Rate Information Category is 'ordinary'.

Units per Time Period (UTP)

00000000

Spare

00000001 to 10000000

Reserved for network specific use 1

10000001 to 11111011	Spare
11111100	Unit charge: 100 yen
11111101	Unit charge: 10 yen
11111110	No indication
11111111	Spare

Extension Indicator:

0	Octet continues through next octets (octets 3 to 17 or 19 to 33)
1	Last octet

Charge Rate Information Length:

Can be 5, 8, 11 or 14

Charge Rate Information Category (CRIC1 and CRIC2):

00000000	Spare
00000001 to 10000000	Reserved for network specific use
10000001 to 11111011	Spare
11111100	Public (payphone)
11111101	Ordinary
11111110	No flexible charge rate information
11111111	Spare

A certain specific CRIC value can only occur once in the CI parameter.

Initial Units (IU)

Value range 0 - 15 encoded as ASCII (IA5) coded in 2 octets M and N. The units are calculated using the formula $(10M + N)$.

Daytime charge rate (DCR) -> first charging interval

Consists of the Time Period (TP) field with the vValue range '1 - 999' encoded as ASCII (IA5) coded in three octets A, B and C. The timer period is calculated using the formula $(100A + 10B + C)/2$. For every period indicated by this interval, the call charge is incremented with the amount indicated by the UTP field. The unit of the TP field is 0.5 s and for example value 120 stands for 60 s (the call charge will be incremented every minute with the amount indicated by the UTP field for example 10 yen).

Evening time charge rate (ECR) -> second charge interval

Same coding as for DCR.

Night time charge rate (NCR) -> third charging interval

Same coding as for DCR.

Spare charge rate (SCR) -> fourth charging interval

Same coding as for DCR.

Note: The maximum length (33 octets) of the CI parameter occurs, when the charge rate information is given for both the 'ordinary' and 'public' category at the same time for two CRIC values. The length 17 octets, if the charge rate information is given either for 'ordinary' or 'public' category.

The minimum length (2 octets) of the CI parameter occurs, when CRIC field holds the value 'No flexible charge rate information'. In this case the UTP field holds the value 'No indication' and the Extension indicator is coded '1'. This setting is used e.g. when flexible charging information is not transferred bit TCA is sent in CHG.

Structure of Charge Information for CIT 'Advanced charge rate transfer':

Structure of Charge Information (type 2) (x-ci-data):

The type 2 information is sent in backward direction from a charging point to indicate the actual charged amount (pulse) to be incremented to the subscriber's accumulator. Note that the presented structure content of Charge Information (CI) parameter is valid only, if the Charge Information Type (CIT) parameters holds the value 'Advanced Charge Rate Transfer' (TDS service).

Tariff Advanced Charge Rate Transfer (TARCT) service								
8 (bit)	7	6	5	4	3	2	1	Octet number
Extension	Reserved				Signal Element Type			1st
Extension	Reserved			Activation ID				1a
Extension	Operation Class		Operation Type					1b
Extension	Charging Party Type		Tariff Collecting Method					1c
Tariff Rate Presentation								2nd

Signal Element Type:

000 Reserved

001	Reserved
010	Start (Executes operation to be executed)
011 to 11	Reserved

Activation ID:

0000	Preliminary value
------	-------------------

Operation Class:

00	Class 1 (no report)
01 to 11	Reserved

Operation Type:

00000 to 00101	Reserved
00110	Immediate charging indications (notifies charging timing).

Charging rate information may also be included.

00111 to 11111	Reserved
----------------	----------

Charging Party Type:

000	Caller charging
-----	-----------------

Others Reserved

Tariff Collecting Method:

0000	Subscriber billing – normal
------	-----------------------------

Others Reserved

Tariff Rate Presentation

00000000	Reserved
00000001	Reserved
00000010	No fare/rate information
00000011 to 11111111	Reserved

Example for CI (when CIT is 11111110 (Charging rate transfer)):Example CI Type 1:

X-CHGInfo:

fe**fd**7d0e3030303238303330303332303332fe**fc**7d0e3132333435363738393031323334

The first octet

fe**fd**7d0e3030303238303330303332303332fe**fc**7d0e3132333435363738393031323334 of the above example means:

1st octet: Charge Information Type -> fe (Charge rate transfer)

The second octet

fe**fd**7d0e3030303238303330303332303332fe**fc**7d0e3132333435363738393031323334 of the above example means:

2nd octet: Units per Time Period -> fd (100 yen)

The third and fourth octets

fe**fd**7d0e3030303238303330303332303332fe**fc**7d0e3132333435363738393031323334 of the above example means:

3rd octet: Extension + Charging Rate Information Category -> 7d (0 + Ordinary)

4th octet: Charging Rate Information length -> 0e (14 octets, 2+3x4)

The last part

fe**fd**7d0e**3030303238303330303332303332**fe**fc**7d0e**3132333435363738393031323334** of the above example means:

Remaining octets Charging rate information content:

IU = 3030 (M=0, N = 0) -> 0 yen = 10(0)+0

DCR = 303238 (A = 0, B = 2, C = 8) -> 14 secs for 10yen (100(0)+20+8)/2

ECR = 303330 (D = 0, E = 3, F = 0) -> 15 secs for 10yen (100(0)+30+0)/2

NCR = 303332 (G = 0, H = 3, I = 2) -> 16 secs for 10yen (100(0)+30+2)/2

SCR = 303332 (J = 0, K = 3, L = 2) -> 16 secs for 10yen (100(0)+30+2)/2
For IU value 1 -> 3132 (M=30H, N = 31H)

DCR for TP value 120 (60 minutes) -> 3132 (A = 31H, B = 32H, C = 30H)

ECR, NCR and SCR similar as for DCR.

Example for CI (when CIT is 00000011 (advanced charge rate transfer)) Example CI Type 2:

X-CHGInfo:030220068002

The first part **030220068002** of the above example means:

1st octet: Charge Information Type -> 03 (Advanced Charge Rate Transfer)

The second part 03**02**20068002 of the above example means:

2nd octet: Extension (0) + Reserved (0000) + Signal Element Type (010) -> 02 (Start, executes operation to be executed)

The third part 0302**20**068002 of the above example means:

3rd octet: Extension (0) + Reserved (010) + Activation ID (0000) ->20 (Preliminary value).

The fourth part 030220**06**8002 of the above example means:

4th octet: Extension (0) + Operation Class (00) + Operation Type (00110) -> 06 (Class 1 (no report) and Immediate charging indication (notifies charging timing)

The fifth part 03022006**80**02 of the above example means:

5th octet: Extension (0) + Charging Party Type (000) + Tariff Collecting Method (0000) -> 80 (Caller charging and Subscriber billing (normal))

The sixth part 0302200680**02** of the above example means:

6th octet: Tariff Rate Presentation (00000010) -> 02 (No fare/rate information)

5.2.12 X-VMS-Request Header

The X-VMS-Request header is used in conjunction with Voice Mail Service and is included in initial SIP INVITE by CSCF.

Syntax:

X-VMS-Request = "X-VMS-Request" HCOLON
VMS-Billing-Code SEMI VMS-Call-Type

VMS-Billing-Code = quoted-string

VMS-Call-Type = "text" EQUAL [LDQUOT]
("fax-print" |
"personal-redirect" |
"transfer-ivr" |
"outcall-pager" |
"outcall-notify" |
"call-sender" |
"wakeup" |
token) [RDQUOT]

Example:

X-VMS-Request: "12"; text="fax-print"

5.2.13 X-GENERIC-NUM Header

The X-GENERIC-NUM header is used to determine the call as a BCD call in diversion cases.

Syntax:

X-GENERIC-NUM = "X-GENERIC-NUM" HCOLON X-GENERIC-NUM-VALUE

*(COMMA X-GENERIC-NUM-VALUE)

X-GENERIC-NUM-VALUE = *(HEXDIG)

Example:

X-GENERIC-NUM: 0683101332040000, 0683101332040001

X-GENERIC-NUM: 0683101332040000

Note: there can be multiple "generic numbers" up to eight X-GENERIC-NUM headers.

The X-GENERIC-NUM value must contain even number of characters.

5.2.14 P-Asserted-Service Header

The P-Asserted-Service header field is used among trusted SIP entities (typically intermediaries) to carry the service information of the user sending a SIP message.

It carries information that is derived service identification.

Example:

P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel

5.2.15 P-Preferred-Service Header

The P-Preferred-Service header field is used by a user agent sending the SIP request to provide a hint to a trusted proxy of the preferred service that the user wishes to be used for the P-Asserted-Service field value that the trusted element will insert.

Example:

P- Preferred-Service: urn:urn-7:3gpp-service.ims.icsi.mmtel

5.2.16 **P-Ericsson-Original-Contact Header**

The P-Ericsson-Original-Contact Header is used in conjunction with Multi Mobile subscriptions feature.

The P-Ericsson-Original-Contact Header is added by Originating SCC-AS in outgoing initial INVITE to preserve the original Contact Header.

Originating MMTel AS uses the P-Ericsson-Original-Contact Header of the incoming initial INVITE to identify the mobile subscription of session. After mobile subscription identification, the originating MMTel AS removes this header from the outgoing initial INVITE.

Syntax:

Same as SIP Contact Header. Refer [4]

5.2.17 **P-Ericsson.Invocation-History**

The P-Ericsson.Invocation-History header is added in outgoing initial INVITE messages by application servers (MMTel, SCC and NW AS), when using the generic SIP port. The header provides information of which application servers an initial INVITE message has invoked. In case several application servers have been invoked for the INVITE session, each AS will add its own header with name, session case and registration state.

Example:

P-Ericsson.Invocation-History: as=MMTelAS;sescase=orig;regstate=unreg

5.3 **Bodies**

The definition of the bodies for each body used by MTAS is as specified in section 4.5.

5.4 **Adaptations**

5.4.1 **Phone-context**

Telephone numbers used in a tel URI (or a corresponding SIP-URI) can be either global (starting with '+') or local (doesn't start with '+'). According to RFC3966 a local telephone must have a 'phone-context' parameter associated with it.

Examples:

Local tel-URI

```
tel:112;phone-context=foo.com  
sip:112;phone-context=foo.com@bar.com;user=phone
```

Global tel-URI `tel:+46-8-112`
 `sip:+46-8-112@foo.com;user=phone`

MTAS behavior described below.

- a MTAS accepts a SIP message with a tel-URI containing a local telephone number without the parameter 'phone-context'. It depends on the actual service and number normalization which phone-context will be used.
 MTAS may send out a SIP message with a tel-URI containing local telephone number without the parameter 'phone-context', if such has earlier been received.
- b MTAS accepts a SIP message with a tel-URI containing a global telephone number with the parameter 'phone-context'. The 'phone-context' parameter is in this case seen as regular parameter unrelated to the number. It depends on the actual service how it will be used.
 MTAS may send out a SIP message that contains a tel-URI with a global telephone number with 'phone-context' parameter if such has earlier been received.

6 Related Standards

The MTAS implementation of SIP is based on ref [19] and ref [4].

7 Terminology

7.1 Abbreviations

Abbreviation	Expansion
AOC	Advice of Charge
ASCII	American Standard Code for Information Interchange
ATCF	Access Transfer Control Function
AUC	Additional User Category
B2BUA	Back-to-Back User Agent
BCD	Business Call Direct
CA	Charge Area
CI	Charge Information
CID	Charge Information Delay
CIEC	Chosen Inter Exchange Carrier
CIT	Charge Information Type
CS	Circuit Switch

Abbreviation	Expansion
CSCF	Call/Session Control Function
CUG	Closed User Group
FCD	Flexible Communication Distribution
IANA	Internet Assigned Numbers Authority
ICS	IMS Centralized Services
IEC	Inter Exchange Carrier
IMS	IP Multimedia Subsystem
IP	Internet Protocol
ISC	IMS Service Control
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
MIME	Multipurpose Internet Mail Extensions
NCC	Northbound Call Control
OAUC	Originating Additional User Category
OCA	Originating Charge Area
OLEC	Originating Local Exchange Carrier
POI	Point of Interfaces
PSTN	Public Switched Telephone Network
RFC	Request For Comments
RTTI	Real-time Transfer of Tariff Information
PSI	Public Service Identity
S-CSCF	Serving CSCF
SCC	Service Centralization and Continuity
SCP	Service Control Point
SCPC	Service Control Point Carrier
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SRVCC	Single Radio Voice Call Continuity
ST	SIP Trunking
TAUC	Terminating Additional User Category
TCA	Terminating Charge Area
TLEC	Terminating Local Exchange Category

Abbreviation	Expansion
TCP	Transmission Control Protocol
UAC	User Agent Client
UAS	User Agent Server
UDP	User Datagram Protocol
UTF	UCS/Unicode Transformation Format
XML	Extended Markup Language

7.2 Definitions

Term	Meaning
Back-to-Back User Agent	When the same MTAS acts a UAS and as a UAC in an end-to-end session.
Conference MTAS	MTAS acting as the Conference Focus in an ad hoc or a scheduled conference.
Document	A block of non-SIP data that is carried in the body of a SIP body. <i>Note: This term was defined by this document.</i>
Interface	The designation of a connection between two nodes
Protocol	The implementation of an interface
User MTAS	MTAS acting for the served user as either an originating, terminating or diverting application server,
userinfo	The part of a SIP URI before the "@". Refer to reference [4].
XML Schema	A data definition that an XML instance document must conform to.

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