Enriched Calling Technical Specification

Version 2.0

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Contents

1 Introduction

1.1 Overview

This document describes the functional architecture and technical realisation of Enriched Calling Services.

- The enriched calling concept focuses in evolving the current call experience in several key aspects:
- Pre-call experience: A user can “compose” information (by including a subject, location, picture, etc.) prior to placing the call such that the other side is able to see the composed pre-call information while receiving the incoming call.
- In-call experience: A user can share content during a call: chat, files (or group of files like presentations), location, background audio, video.
- Post-call experience: Similar to pre-call experience, a user can “compose” additional information when a call is rejected or unanswered, for the other side to view.
- Enriched Call Logs: A user is able to see call log with enriched information e.g. information shared during pre-call, and post-call.

All these features are to be provided in conjunction with Telco telephony services, i.e. traditional circuit switched (CS) call and Multimedia Telephony via LTE and EPC-integrated Wi-Fi.

Regarding the call enrichment features, they are to be provided creating new Enriched Calling services and also using the RCS services and extensions (chat, group chat, file transfer, geolocation push, video share and via the Conversational Video service (IR.94), companion service to Multimedia Telephony).

Enriched calling is based on RCS. Therefore, all RCS features will be supported in parallel with the new Enriched calling features. The method to identify an incoming message as an Enriched calling information is to use dedicated ICSI tags.

Multidevice use cases are currently not covered in this document.

1.2 Terms and Abbreviations

Term	Description
AMR	Adaptive Multi-Rate
A-Party	The party that initiates a communication event e.g. creates and sends a chat message or File Transfer, or initiates a call to the B-Party.
B-Party	The party that receives or is intended to receive a communication event e.g. Chat Message, File Transfer or call from the A-Party.
Call Composer	A view on the device that allows the A-Party to enrich outgoing calls with pre-call content before placing the call.
Call Composer Session	A call composer session is a logical session between the originating and the terminating party to exchange enriched call content. A call composer session represents all XML fragments, including one or more information elements (e.g. an optional field “call importance” and the mandatory field “call composer ID”) created between the time of A-Party starting call composer and until the Enriched Call is answered and (in case a picture was included in

Term	Description
	call composer) the picture delivery notification is received by the A-Party.
CLIP	Call Line Identification Presentation
CLIR	Call Line Identification Restriction
Contact	A communication partner represented by a unique communication identifier e.g. MSISDN or SIP URI. Note: Not to be confused with a device 'contact', saved to the address book.
CS	Circuit Switch
DTD	Document Type Definition
Enabled	A contact who is known to have Enriched Calling functionality, but may or may not be currently available (online or offline).
EnCall	Enriched Calling
Enriched Calling / Enriched Content	Functionality described in this document which allows the user to enhance the standard ('plain') voice call experience.
Enriched Calling service	An Enriched Calling service represent a service who uses Enriched Calling mechanism (e.g. Enriched Calling ICSI tags, Service Description, Session management, SDP and MSRP) as defined in this document.
Enriched Calling Session	An Enriched Calling session is a SIP session with MSRP media session each identified by a specific ICSI, used by the following services defined in this document: Call Composer, Post Call, Shared Sketch and Shared Maps.
EPC	Evolved Packet Core
IMDN	Instant Messaging Disposition Notification
LTE	Long Term Evolution
MSRP	Message Session Relay Protocol
Native RCS Device	A device with a pre-installed RCS client (as opposed to a downloaded RCS client).
Offline	A contact who is Enriched Calling enabled but is NOT currently available.
Online	A contact who is Enriched Calling enabled and is currently available.
Plain Voice Call	A voice call with no enriched content.
SDP	Session Description Protocol
TON	Type Of Number
UX	User Experience
XML	Extensible Markup Language

1.3 Document Cross-References

Ref	Document Number	Title
1	[3GPP TS 24.008]	3GPP TS 24.008 Release 12, 3rd Generation Partnership Project, Mobile radio interface Layer 3 specification; Core network protocols http://www.3gpp.org

Ref	Document Number	Title
2	[3GPP TS 24.229]	3GPP TS 24.229 Release 10, 3rd Generation Partnership Project; IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) http://www.3gpp.org
3	[3GPP TS 26.141]	3GPP TS 26.141 Release 10, 3rd Generation Partnership Project; IP Multimedia System (IMS) Messaging and Presence; Media formats and codecs http://www.3gpp.org
4	[PRD-IR.51]	IMS Profile for Voice, Video and SMS over Wi-Fi Version 2.1 13 August 2015 http://www.gsma.com/
5	[PRD-IR.92]	IMS Profile for Voice and SMS Version 9.0 08 April 2015 http://www.gsma.com/
6	[PRD-IR.94]	GSMA PRD IR.94 - "IMS Profile for Conversational Video Service" Version 8.0 17 November 2014 http://www.gsma.com/
7	[PRD-RCC.07]	GSMA PRD RCC.07 - "Rich Communication Suite 6.0 Advanced Communications Services and Client Specification" Version 7.0 21 March 2016 http://www.gsma.com/
8	[RFC3261]	SIP (Session Initiation Protocol) IETF RFC http://tools.ietf.org/html/rfc3261
9	[RFC3264]	An Offer/Answer Model Session Description Protocol IETF RFC http://tools.ietf.org/html/rfc3264
10	[RFC3326]	The Reason Header Field for the Session Initiation Protocol (SIP) IETF RFC http://tools.ietf.org/html/rfc3326
11	[RFC4028]	The Session Timers in the Session Initiation Protocol (SIP) IETF RFC http://tools.ietf.org/html/rfc4028
12	[RFC4122]	The Universally Unique IDentifier (UUID) URN Namespace IETF RFC http://tools.ietf.org/html/rfc4122
13	[RFC4867]	RTP Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs IETF RFC http://tools.ietf.org/html/rfc4867
14	[RFC4975]	The Message Session Relay Protocol (MSRP) IETF RFC http://tools.ietf.org/html/rfc4975

2 Service realisation

2.1 Client configuration

HTTP(s) based client configuration as defined in section 2.3.3.2 of [PRD-RCC.07] will be used to manage the Enriched Calling configuration parameters of the client.
Note, the Management Object extension for Enriched Calling is compatible to earlier versions of the HTTP(s) based client configuration.

NOTE: One device can only act as one client (with only one registration). It is not intended to have a stand-alone Enriched Calling client in parallel to another RCS messaging client.

2.1.1 Void

2.1.2 Additional configuration parameters

The following configuration parameters shall be used in the Service Provider Client Configuration to control Enriched Calling services:

Configuration parameter	Description	Enriched Calling usage
COMPOSER AUTH	This parameter controls the Call Composer service. If set to **0** the Call Composer service is disabled. If set to **1** the Call Composer service is enabled.	Mandatory Parameter
SHARED MAP AUTH	This parameter controls the Shared Map service. If set to **0** the Shared Map service is disabled. If set to **1** the Shared Map service is enabled.	Mandatory parameter
SHARED SKETCH AUTH	This parameter controls the Shared Sketch service. If set to **0** the Shared Sketch service is disabled. If set to **1** the Shared Sketch service is enabled.	Mandatory Parameter
POST CALL AUTH	This parameter controls the Post Call service. If set to **0** the Post Call service is disabled. If set to **1** the Post Call service is enabled.	Mandatory Parameter
CALL COMPOSER TIMER IDLE	This parameter controls the time in seconds after which an idle Call Composer Enriched Calling session shall be terminated by the client. Default value: 180 seconds	Optional Parameter

Table 1: Enriched Calling Related Configuration Parameters

In case the network does not support Enriched Calling and thus does not provide any of the Enriched Calling related parameters defined above, the client shall disable the Call Composer, Post Call, Shared Maps and Shared Sketch services.

2.1.2.1 RCS Management Tree Additions

The Enriched Calling configuration parameters extend the RCS Management Tree as follows.

Node: /<x>/Services/composerAuth

Leaf node that represents the authorization for the user to use the Enriched Calling Call Composer service

It shall be instantiated if the network supports any Enriched Calling Service.

Status	Occurrence	Format	Min. Access Types
Required	ZeroOrOne	bool	Get, Replace

Table 2: Services MO sub tree addition parameters (composerAuth)

- Values: 0, 1

  0, Indicates that the Enriched Calling Call Composer service is disabled
  The SIP REGISTER shall not include the ICSI pertaining to the Enriched Calling Call Composer in its Contact header. Also, they are not included as part of capability discovery.
  No Enriched Calling Call Composer entry points will be displayed.
  1, Indicates that the Enriched Calling Call Composer service is enabled
  The SIP REGISTER shall include the ICSI pertaining to Enriched Calling Call Composer if used by installed applications in its Contact header. Also, they are included as part of capability discovery. Enriched Calling Call Composer entry points will be displayed based on the capability discovery.

- Post-reconfiguration actions: If the re-configuration transits from "Call Composer service is disabled" to "Call Composer service is enabled" the client shall re-register in IMS to include the ICSI pertaining to the Enriched Calling Call Composer. The Enriched Calling Call Composer entry points will be displayed based on the capability discovery. If the re-configuration transits from "Call Composer service is enabled" to "Call Composer service is disabled" the client shall re-register in IMS to remove the ICSI pertaining to Enriched Calling Call Composer. The Enriched Calling Call Composer entry points will be no longer displayed.
- Associated HTTP XML parameter ID: “composerAuth”.

Node: /<x>/Services/sharedMapAuth

Leaf node that represents the authorisation for the user to use the Enriched Calling Shared Map service during a call.

It shall be instantiated if the network supports any Enriched Calling Service.

Status	Occurrence	Format	Min. Access Types
Required	ZeroOrOne	bool	Get, Replace

Table 3: Services MO sub tree addition parameters (sharedMapAuth)

- Values: 0, 1

0, Indicates that the Enriched Calling Shared Map service is disabled
The SIP REGISTER shall not include the ICSI pertaining to the Enriched Calling Shared Map in its Contact header. Also, they are not included as part of capability discovery. No Shared Map entry points will be displayed during the call.
1, Indicates that the Enriched Calling Shared Map service is enabled
The SIP REGISTER shall include the ICSI pertaining to the Enriched Calling Shared Map if used by installed applications in its Contact header. Also, they are included as part of capability discovery. Shared Map entry points will be displayed during the call based on the capability discovery.

- Post-reconfiguration actions: If the re-configuration transits from "Shared Map service is disabled" to "Shared Map service is enabled" the client shall re-register in IMS to include the ICSI pertaining to Enriched Calling Shared Map. If the re-configuration transits from "Shared Map service is enabled" to "Shared Map service is disabled" the client shall re-register in IMS to remove the ICSI pertaining to Enriched Calling Shared Map.
- Associated HTTP XML parameter ID: “sharedMapAuth”

Node: /<x>/Services/sharedSketchAuth

Leaf node that represents the authorisation for the user to use the Enriched Calling Shared Sketch service during a call.

It shall be instantiated if the network supports any Enriched Calling Service.

Status	Occurrence	Format	Min. Access Types
Required	ZeroOrOne	bool	Get, Replace

Table 4: Services MO sub tree addition parameters (sharedSketchAuth)

- Values: 0, 1

0, Indicates that the Enriched Calling Shared Sketch service is disabled
The SIP REGISTER shall not include the ICSI pertaining to the Enriched Calling Shared Sketch in its Contact header. Also, they are not included as part of capability discovery. No Shared Sketch entry points will be displayed during the call.
1, Indicates that the Enriched Calling Shared Sketch service is enabled
The SIP REGISTER shall include the ICSI pertaining to the Enriched Calling Shared Sketch if used by installed applications in its Contact header. Also, they are included as part of capability discovery. Shared Sketch entry points will be displayed during the call based on the capability discovery.

- Post-reconfiguration actions: If the re-configuration transits from "Shared Sketch is disabled" to "Shared Sketch is enabled" the client shall re-register in IMS to include the ICSI pertaining to Enriched Calling Shared Sketch. If the re-configuration transits from "Shared Sketch is enabled" to "Shared Sketch is disabled" the client shall re-register in IMS to remove the ICSI pertaining to Enriched Calling Shared Sketch.
- Associated HTTP XML parameter ID: “sharedSketchAuth”.

Node: /<x>/Services/postcallAuth

Leaf node that represents the authorisation for the user to use the Enriched Calling Post Call service.

It shall be instantiated if the network supports any Enriched Calling Service.

Status	Occurrence	Format	Min. Access Types
Required	ZeroOrOne	bool	Get, Replace

Table 5: Services MO sub tree addition parameters (postcallAuth)

- Values: 0, 1

  0, Indicates that the Enriched Calling Post Call service is disabled
  The SIP REGISTER shall not include the ICSI pertaining to the Enriched Calling Post Call service in its Contact header. Also, they are not included as part of capability discovery.
  No Enriched Calling Post Call entry points will be displayed.
  1, Indicates that Post Call service is enabled
  The SIP REGISTER shall include the ICSI pertaining to Enriched Calling Post Call if used by installed applications, in its Contact header. Also, they are included as part of capability discovery. Enriched Calling Post Call entry points will be displayed based on the capability discovery.

- Post-reconfiguration actions: If the re-configuration transits from "Post Call is disabled" to "Post Call is enabled" the client shall re-register in IMS to include the ICSI pertaining to Enriched Calling Post Call. If the re-configuration transits from "Post Call is enabled" to "Post Call is disabled" the client shall re-register in IMS to remove the ICSI pertaining to Enriched Calling Post Call.
- Associated HTTP XML parameter ID: “postCallAuth”.

Node: /<x>/Other/callComposerTimerIdle

This parameter controls the time in seconds during which an Enriched Calling Call Composer session is allowed to be idle before it is closed.

Status	Occurrence	Format	Min. Access Types
Required	ZeroOrOne	Int	Get

Table 6: Services MO sub tree addition parameters (callComposerTimerIdle)

- Values: < Timer value in seconds>

  When set to 0, there shall be no timeout.
  The default value is 180 (seconds).

- Optional parameter
  (It is mandatory and becomes relevant only if composerAuth is set to 1)

- Post-reconfiguration actions: There are no additional actions apart from application of the new value from the time of re-configuration onwards.
- Associated HTTP XML parameter ID: “callComposerTimerIdle”

As indicated, all parameters apart from the callComposerTimerIdle parameter shall be included in the Services MO defined in [PRD-RCC.07].

The callComposerTimerIdle parameter is added to the Other MO tree defined in [PRD-RCC.07].

2.2 Capability discovery

The service is implemented using the standard RCS contact and capability discovery service as per section 2.6.1 of [PRD-RCC.07].

In a capability exchange the client shall include the following groups of capabilities:

- RCS service capabilities and
- The capabilities associated to each of the Enriched Calling services supported by the client implementation as they are based on existing standards.

NOTE: The Enriched Calling ICSI tags and service tuples associated to each of the Enriched Calling services are described in the sections 2.3, 2.4, 2.5, 2.9.7 and 2.9.8.

The following Enriched Calling services described in section 2.9.1 – 2.9.6 do not use Enriched Call ICSI tags or service tuples:

- Live Video (Conversational Video or Video Share)
- Share any file (file transfer via http)
- Share text during a call (1 to 1 chat)
- Location share (Geolocation PUSH during a call)

The triggers for Capability discovery described in section 2.6 of [PRD-RCC.07] apply. In particular the capability discovery shall take place:

- When the initial capability discovery takes places.
  (The information has to be cached by the client. Initial discovery is done when the Enriched Calling client is started for the first time.) Initial capability discovery will verify the Enriched Calling services that may be used.
- When accessing the details of a contact in the address book (call composer)
- After establishing voice call to obtain the capabilities for content sharing during a call (A and B-Party)When in a call, if the capabilities of one of the participants changes, that client shall initiate the process
- After an unexpected termination of a service (e.g. video share session drops unexpectedly)

NOTE: It is expected the client does not check the capabilities more often than mentioned in the RCS specification.

In addition to the triggers defined in [PRD-RCC.07], Enriched Calling defines the following additional triggers:

- While typing numbers manually:
  - After minimum N digits (up to OEM implementation, 6 digits recommended) AND After M seconds (up to OEM implementation, 2 seconds recommended) since last typed number and before the next number is typed.

Support for Enriched calling services will be indicated through Capability discovery via dedicated ICSIs.

For Enriched calling services ICSI tags and service tuples definitions see sections 2.4.1, 2.5.1, 2.9.7 and 2.9.8 of this document.

2.3 Enriched Calling Session Service procedures

To support the Enriched Calling services discovery functionality presented in this document, it is necessary to extend the tag mechanism by adding ICSI tag and service tuple for each Enriched Calling Service.

It should be noted that the services described in sections 2.9.1 to 2.9.6 of this document do not use the Enriched Calling service definition, but instead use the service definition as defined in RCS:

- Live Video – Using Conversational Video or using Video Share as defined in RCS (IR.74 endorsement)
- Share any file – using file transfer via http as defined in RCS
- Exchanging messages – using 1 to 1 chat as defined in RCS
- Location push – using Geolocation PUSH during a call as defined in RCS

2.3.1 Initiating an RCS Enriched Calling Session

Handling at Initiating Nodes

The RCS Client shall send an initial SIP INVITE request according to the rules and procedures of [3GPP TS 24.229]. In this SIP INVITE request, the RCS Client:

- Shall include the address of the target RCS contact in the Request-URI;
- Shall include an Accept-Contact header field with the ICSI for the Enriched Calling service this session is related to (for Enriched Calling ICSI tag definition, see sections 2.3, 2.4, 2.5, 2.9.7 and 2.9.8 of this document) percent encoded as per [3GPP TS 24.229] section 7.2A.8.2 "*Coding of the ICSI*" in a g.3gpp.icsi-ref media feature tag with the *require* and *explicit* parameters;
- Shall set the P-Preferred-Service header field with the value of the Enriched Calling ICSI tags (for Enriched Calling ICSI tag definition, see sections 2.3, 2.4, 2.5, 2.9.7 and 2.9.8 of this document);
- Shall include a Contact header field with the ICSI for the Enriched Calling specific service this session is related to percent encoded as per [3GPP TS24.229] section 7.2A.8.2 "*Coding of the ICSI*" in a *g.3gpp.icsi-ref* media feature tag;

- Shall include the address of the originating RCS Client that has been authenticated as per section 2.5.3.2 of [PRD-RCC.07]and [3GPP TS 24.229];
- Shall include a User-Agent header field as specified in [RFC3261];
- Should include a Session-Expires header field with the refresher parameter set to "uac" according to the rules and procedures of [RFC4028];
- Shall include a MIME SDP body as an SDP offer as described in section 2.3.4 of this document.
- Shall send the SIP INVITE request according to the rules and procedures of [3GPP TS 24.229]

On receipt of the SIP 200 "OK" response to the initial SIP INVITE request the RCS Client shall handle the response according to the rules and procedures of [3GPP TS 24.229], with the following clarifications:

1. The RCS Client shall start a SIP session timer using the value received in the Session- Expires header field according to the rules and procedures of [RFC4028].
2. The RCS Client shall generate and send a SIP ACK request as an acknowledgement of the final response according to the rules and procedures of [RFC3261].
3. The RCS Client shall establish the Media Plane as per [3GPP TS 24.229].

On receipt of a SIP error response to the initial SIP INVITE request the RCS Client shall handle the response according to the rules and procedures of [3GPP TS 24.229], with the following clarifications:

1. The RCS Client may indicate to the user that the session could not be established;
2. The RCS Client shall generate and send a SIP ACK request as an acknowledgement of the final response according to the rules and procedures of [RFC3261].

Handling at Intermediate Nodes

Intermediate nodes (e.g. access gateways, application servers) may stay in the media path depending on Service Provider policy.

Handling at Terminating Nodes

On receipt of the SIP INVITE request the RCS Client shall check if the Enriched Calling services as indicated by the ICSI in the Accept-Contact header are running on the device:

1. If Enriched Calling services are not running on the device, the RCS Client shall respond with a SIP 403 Forbidden error with a Warning header set to “*Unsupported Service*”.
2. If Enriched Calling services are running on the device, the RCS Client:

   a) Shall respond with a SIP 200 OK, with a valid SDP offer (as per section 2.3.4 of this document) if the session is accepted and shall start a SIP session timer and take on the role of "uas" according to the rules and procedure of [RFC4028], and establish the Media Plane as per [3GPP TS 24.229], or
   b) Shall respond with a SIP 603 Decline if the session is not accepted by the user.

If the RCS Client is already involved in an Enriched Calling session with the same contact and the same service tag (i.e. same ICSI), it shall terminate the ongoing session as per section 2.3.2 of this document before accepting the new one.

The Multi-device scenario is left open to the MNO implementation decision.

2.3.2 Terminating an RCS Enriched Calling Session

To close an Enriched Calling session due to an explicit closing request, the RCS Client:

1. Shall generate a SIP BYE request according to the rules and procedures of [3GPP TS 24.229], with the Reason Header field as defined in [RFC3326] with the protocol-value set to SIP, the protocol-cause set to *200* (e.g., *SIP;cause=200*);
2. Shall send the SIP BYE request according to the rules and procedures of [3GPP TS24.229];
3. Shall release all Media Plane resources corresponding to the Enriched Calling Session being closed.

For the Call Composer session a client shall close an Enriched Calling session when it has been idle for longer than the value configured for the COMPOSER SESSION TIMER configuration parameter defined in section 2.1.2 of this document. In this case, the RCS Client:

1. Shall generate a SIP BYE request according to the rules and procedures of [3GPP TS 24.229], with the Reason Header field as defined in [RFC3326] has a protocol-value set to SIP and a protocol-cause set to *200*;
2. Shall send the SIP BYE request according to the rules and procedures of [3GPP TS24.229];
3. Shall release all Media Plane resources corresponding to the Enriched Calling session being closed when a final response to that BYE request is received.

For the Enriched Calling Post-call procedures the same behaviour applies, except that there is no explicit session timer parameter.

When receiving a SIP BYE request, the client shall

1. Shall generate a SIP 200 OK response according to the rules and procedures of [3GPP TS 24.229];
2. Shall send the SIP 200 OK response according to the rules and procedures of [3GPP TS24.229];
3. Shall release all Media Plane resources corresponding to the Enriched Calling session being closed.

NOTE: When the Enriched Calling session wants to send further traffic to the other client after the session has been closed, a new session shall be started as described in section 2.3.2 of this document.

2.3.3 SDP Contents

SDP Contents when Initiating a Session

An initiating entity (e.g. an RCS Client) shall populate the SDP of an Enriched Calling session invitation request to match the media streams that are requested by the pertaining Enriched Calling service. Therefore the initiating entity shall include in the SIP INVITE request a MIME SDP body as an SDP offer according to the rules and procedures of [3GPP TS 24.229].

The SDP offer shall contain media descriptions matching the requested Media Streams according to the following clarifications:

When including an offer for a Media Stream using MSRP, the initiating entity shall include a media description according to the rules and procedures of [RFC4975], with the following constraints:

- The *a=accept-types* shall only include:
    - for Call Composer the *application/vnd.gsma.encall+xml* and *message/cpim* MIME types
    - for a Post Call Note or Post Call Audio message, the *application/vnd.gsma.encall+xml* and *application/vnd.gsma.rcs-ft-http+xml* MIME types.
- for Call Composer, the a=accept-wrapped-types attribute shall only include the *message/imdn+xml* and *application/vnd.gsma.rcs-ft-http+xml* MIME types.
- for a Post Call Note or Post Call Audio Message, no a=accept-wrapped-types attribute shall be set

For Shared Sketch and Shared Map refer to section 2.9.9.4.

SDP Handling at Intermediate Nodes

Intermediate nodes shall include the contents of the SDP they received in the SDP they send out, in accordance with the rules and procedures of [3GPP TS 24.229] and [RFC3264]. Specific attributes in the SDP may be modified for the following reason:

To modify IP-address and port information to insert the intermediate entity in the media path of the session.

All modifications shall be done according to the rules and procedures of [RFC3264] and the respective Media Stream standards (i.e. [RFC4975] for MSRP-based media description and [RFC3264].

SDP Handling at Terminating Nodes

A terminating entity (e.g. an RCS Client) shall process an incoming SDP and accept, modify or reject the Media Streams requested in the incoming SDP as defined by [3GPP TS 24.229] and [RFC3264]. The terminating entity shall handle the media descriptions according to the following clarifications:

Media descriptions for a Media Stream for messages, using MSRP, shall be handled and responded to according to the rules and procedures of [RFC4975], with the following constraints:

- The *a=accept-types* shall only include:
  - for Call Composer the *application/vnd.gsma.encall+xml* and *message/cpim* MIME types
  - for a Post Call Note or Post Call Audio message, the *application/vnd.gsma.encall+xml* and *application/vnd.gsma.rcs-ft-http+xml* MIME types.
- For Call Composer, the *a=accept-wrapped-types* attribute shall only include *message/imdn+xml* and *application/vnd.gsma.rcs-ft-http+xml* MIME types.
- For a Post Call Note or Post Call Audio Message, no *a=accept-wrapped-types* attribute shall be set.

For Shared Sketch and Shared Map refer to section 2.9.9.4.

2.3.4 MSRP Session Handling

A client sending a MSRP SEND request SHALL request MSRP Failure Reports and SHALL NOT request MSRP Success Reports.

When no response is received to an MSRP SEND, the rules and procedures of [RFC4975] are followed with the following clarification:

The client not receiving an MSRP SEND response should set the *cause=503* along with an optional protocol-text (e.g. *SIP;cause=503;text="Service Unavailable"*) in the SIP BYE request it generates. The client should indicate to the user that an error occurred when sending the message in the MSRP SEND.

2.3.5 NNI and IOT considerations

No specific guidelines apply other than what is already defined in section 2.12 of [PRD-RCC.07].

2.4 Call composer

2.4.1 Call composer procedures

The Call Composer will use the following elements and is defined as a separate service:

Element	Value / Description
Value carried in an Accept-Contact or Contact header	+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.gsma.callcomposer"
Value carried in a P-Preferred-Service or P-Asserted-Service header	urn:urn-7:3gpp-service.ims.icsi.gsma.callcomposer
Service Tuple	org.3gpp.urn:urn-7:3gpp-service.ims.icsi.gsma.callcomposer Version: 1.0

<table>
<tr><td colspan="2">Service realisation type</td><td>Enriched Calling service</td></tr>
<tr><td rowspan="3">Enriched Calling service*</td><td>Media type</td><td>MSRP</td></tr>
<tr><td>Auto accept</td><td>Yes</td></tr>
<tr><td>Store & Forward</td><td>Not required</td></tr>
</table>

Table 7: Enriched Calling Composer service realisation summary

2.4.2 Picture transfer procedures

The Call Composer is implemented using the Enriched Calling service procedures described in section 2.3 of this document.

If a picture is sent as part of a call composer session, the picture is first uploaded to the HTTP content server using File Transfer upload procedures defined in steps 1, 2, 3 and 4 of section 3.5.4.8.3.1 of [PRD-RCC.07].

If the file upload cannot be completed the client MAY apply the Upload Resume procedure defined in section 3.5.4.8.3.1.1 of [PRD-RCC.07]. The file upload should start as soon as the picture is selected by the user and the B-Party has Enriched Call Capability.

The format of the picture should be compliant with section 4.2 and 4.3 of [3GPP TS 26.141].

In case the picture file size exceeds a recommended value of 80kB or in case its resolution would not give an advantage for displaying on a mobile screen, the picture should be resized before it is uploaded.

The A-Party may upload subsequent different pictures in case they decide to change the call composer picture to be displayed. If this change happens while the previous picture is still being uploaded, the A party shall cancel the ongoing upload.

2.4.3 Session management

If the client is authorized to offer the Call Composer service according to the definition of the COMPOSER AUTH parameter defined in Table 1 and if the capability discovery is successful for a recipient address (B-Party supports Call Composer service) and the user enters a call composer screen associated with this recipient address, the A-Party client shall initiate an Enriched Calling session to the recipient address using procedures described in section 2.3 and the ICSI defined in section 2.4.1.

A client receiving an invitation for a call composer session shall auto accept the session in order to establish the MSRP session.

The A-Party shall only close the Enriched Calling session if the user has initiated the call set-up to the recipient and all of the following conditions apply:

- the pre-call data has been received by the B-Party, i.e. a 200 OK MSRP response is received by the client
- the call set-up has been cancelled or the call is answered
- the A-Party has received the displayed notification for the latest picture sent to the B-Party if a pre-call picture has been set.

NOTE: It is still possible to toggle the call importance after the call has been initiated until it is answered or cancelled. Also, if the link for the pre-call picture is not available before the call is initiated, it can be sent afterwards. Other call composer elements cannot be changed after the call has been initiated.

If the first condition is never met, the A-Party will close the Enriched Calling session following the procedures described in section 2.3.4 of this document.

If the third condition is never met, the A-Party closes the session at the end of the call or eventually before in the specific case an in call services is launched.

The A-Party shall close the Enriched Calling session if there is no transfer of call composer elements in the call composer session and no attempt to set-up a call to the recipient for the time defined in the configuration parameter CALL COMPOSER TIMER IDLE of Table 1.

The A-Party client shall establish a new session after time-out when a new call composer element needs to be sent using procedures described in section 2.3 and the ICSI defined in section 2.4.1.

In accordance with the receiver procedures in section 2.4.5 an A-Party client shall send a call composer picture again if the user commences a call composer session and a call composer picture has been sent more than 60 minutes ago. Resending of the call composer picture shall follow the procedures described in step 1 of section 2.4.4 of this document.

2.4.4 Call Composer elements

Using the MSRP session established for the Enriched Calling Call Composer, the service employs a XML-based protocol to convey the information to the receiver. Two different elements are used:

1. When a pre-call picture is set: after the upload procedure described in section 2.4.2 (of this document) is successful, the A-Party shall then send a XML message to the receiver in the call composer session:

```
<?xml version="1.0" encoding="UTF-8"?>
<file xmlns="urn:gsma:params:xml:ns:rcs:rcs:fthttp">
  <file-info type="file" >
  <file-size>[file size in bytes]</file-size>
  <file-name>[original file name]</file-name>
  <content-type>[MIME-type for file]</content-type>
  <data url="[HTTP URL for the file]" until="[validity of the file]"/>
</file-info>
<file>
```

Table 8:XML message based on HTTP content server response (see the File Transfer procedure in section 3.5.4.8.3 of [PRD-RCC.07])

This XML body shall be carried in a CPIM body to support delivery and displayed notification (see section 2.4.6). The CPIM content-type header field shall be *application/vnd.gsma.rcs-ft-http+xml*. The MSRP SEND content-type header field shall be message/cpim.

2. The content elements of the call composer, which is sent when the call button is pressed on the A side:

```
<?xml version="1.0" encoding="UTF-8"?>
<rcsenvelope xmlns="urn:gsma:params:xml:ns:rcs:rcs:calldata">
        <rcscalldata>
          <subject>this is the subject</subject>
          <importance>[0/1]</importance>
          <location>[coordinates]</location>
          <composerid>[Random generated number to identify the session]</composerid>
          <picture_url>[HTTP URL for the file] </picture_url >
        </rcscalldata>
</rcsenvelope>
```

Table 9: Call composer service; Complete XML sample

Elements under rcscalldata are optional except composerid which is mandatory. The picture_URL is received from the file upload procedure. The client shall ignore any unrecognized child element of rcscalldata (this is to secure future service extensibility).
The elements under rcscalldata are described below:

- <subject>: The maximum length of the subject field should be 60 characters.
- <importance>: The values are “1” for an important call; “0” for a standard call.
- <location>: The location shall be a geo URL, e.g. geo:37.786971,-122.399671
- <picture_url>: The URL of the picture file on the FT content server to be displayed.
- <composerid>: Random generated number to identify the session. The maximum length should be 10 characters. The composerid is a mandatory field.

In case a user is composing a call and has already selected and sent a picture, but then decides not to show it to the receiver, no picture element will be sent in the XML to indicate to the B-Party to not display the picture which was already downloaded earlier.

In case information has to be sent after the call button has been pressed the following XML is used. This is valid for call importance and in case a picture upload is not finished at the time the call button is pressed.

```
<?xml version="1.0" encoding="UTF-8"?>
<rcsenvelope xmlns="urn:gsma:params:xml:ns:rcs:rcs:calldata">
    <rcscalldata >
        <composerid>[composerid already assigned to this session]</composerid>
        <importance>[0/1]</importance>
        <picture_url>[HTTP URL for the file]</picture_url>
    </rcscalldata>
</rcsenvelope>
```


Table 10: complete XML example after the call button has been pressed

The content of the XML can either include the importance field, or the picture URL or both. If the element is listed, it replaces any value that may have been included in the previous XML sent for this call. The composerid is always mandatory. The ID shall be set to the same value than other call composer data which has been sent previously for the same call composer session.

The MSRP SEND content-type header field shall be *application/vnd.gsma.encall+xml*.

2.4.5 Receiver procedures

After the call composer MSRP session is established, the receiving party of the call composer session needs to wait until the file information is received, which should be downloaded immediately, and/or the call composer information is received. The receiving party shall display all available information when the next incoming call, provided the call comes from the same user who sent the call composer information. The call composer session and the incoming call are correlated by matching the calling party identification information from the SIP/Call Composer session and the incoming CS or Multimedia Telephony call.

The originator calling party identity of the telephony call and the call composer session may be provided in various formats both in the home networks and when roaming. The receiving client shall therefore apply the following matching mechanism to determine whether an incoming telephony call corresponds to received call composer information:

1. If both the originator identity of the telephony call and the call composer session are phone numbers in an international format, the client shall compare all digits of the provided numbers to determine whether they match.
2. If any of the originator identities is not in an international format, the client shall apply an enhanced matching mechanism between the originator identity from the telephony call and the call composer session, e.g. by comparing the 7 digits starting from the end of the number. It is left to the client implementation to apply an even more advanced matching algorithm to decrease the probability of false matches.

The client shall consider an originator identity to be in international format if:

1. for a CS call, the Type Of Number (TON) of the Calling Party BCD Number is set to “international” as defined in [3GPP TS 24.008].
2. for a multimedia telephony call, the P-Asserted-Identity of the SIP INVITE request contains either:

- a tel URI starting with a “+” without phone-context i.e. a global number or
- SIP URI with user=phone parameter without a phone-context in the user part starting with a “+”

3. for a Call Composer session, the P-Asserted-Identity of the SIP INVITE request for call composer session contains either:

   - a tel URI starting with a “+” without phone-context i.e. a global number or
   - SIP URI with user=phone parameter without a phone-context in the user part starting with a “+”

Examples:

- The originator identity of the telephony call: *+447123456789* (display string for an international format number)
- The originator identity of the call composer session: *+447123456789*

⇨ Matching result: Successful

when the applied enhanced matching algorithm is based on the 7 digits starting from the end of the number:

- The originator identity of the telephony call: *006447123456789* (non-international format)
- The originator identity of the call composer session: *+447123456789*

⇨ Matching result: Successful

In case a picture has been sent as part of Call Composer, the receiver side will download the call composer picture file automatically as soon as it has received the link in the call composer picture XML. For download of the call composer picture the procedures in sections 3.5.4.8.3.2 and 3.5.4.8.6.4 of [PRD-RCC.07] apply.

The call composer information shall be automatically accepted by the receiving side. This also includes the picture, which is limited in size.

If the client does not receive calling party identification with the next received call (e.g. due to calling line restriction services), then the call composer information shall not be displayed.

The client displays the call composer information when the Call Composer XML is received AND it matches the corresponding incoming CS or Multimedia Telephony call (as described above).

If call composer information is received by the client but no matching call was received yet, it SHALL discard the call composer information after 30 seconds.

If a call composer picture is received, but no call was received yet, the picture and the picture URL shall be stored by the client for at least 60 minutes.

If the client has stored the picture but not the corresponding composer information, the call composer picture shall only be displayed if the picture URL is received within a subsequent call composer information associated with a call from the originator.

In the case when the B-Party answers the call before the call composer information arrives, the call composer information will be displayed during the call.

2.4.6 Delivery and display notifications

When a Call Composer session is set up, the pre-call picture file information is transported in the MSRP session. The MSRP SEND request carrying the file information contains a request to receive an Instant Messaging Disposition Notification (IMDN) ‘delivery’ notification, and a request to receive an IMDN ‘display’ notification. A client should therefore always include “positive-delivery, display” in the value for the CPIM/IMDN Disposition-Notification header field. The value of “negative-delivery” is not used in Call Composer sessions.

The receiving devices must generate an MSRP SEND request containing the IMDN delivered status when the pre-call picture file information is delivered and another MSRP SEND request containing the IMDN displayed status when the picture is downloaded.

When the B-Party has delivered / displayed notification to be sent while the MSRP session has been torn down, the B-Party sends it through SIP Message using the call composer ICSI (similar to what is done for FToHTTP).

For more details see also Annex A.

2.4.7 Expected Call forwarding behaviour

The receiver side may have one of the call forwarding services activated (e.g. call forward unconditional, call forward on busy) that may interact with call composer service. The interactions will be handled in the following way:

- In case of call forward towards another number (e.g. A-Party makes a call towards B-Party that has activated call forward if not answered towards C-Party), call composer information will never be displayed on C-Party.
- Call composer information will be displayed on the receiver side if there is an incoming call from the same user that led the receiver party into the “ringing” state. If due to call forward the call was not presented on the receiver side, call composer information shall not be displayed.
- Call composer information will be stored in the call logs as soon as the information has been displayed on the receiver side when the call has been ringing (e.g. call forward if not answered).
- Call composer information will NOT be stored in the call logs if the information was not displayed (e.g. call forward unconditional in the network).

2.4.8 Legacy support for call composer

Call composer will not have any legacy support. Call composer can only be used if both sides support enriched calling.

2.5 Post-call Service

If a call is unanswered, the user can choose to either leave a note (reason) or a voice message to send to the receiving side.

This service is implemented using the Enriched Calling mechanism as described in sections 2.1, 2.1.2, 2.2 and 2.3 (of this document) and it uses a similar method as for Enriched Calling call composer as described in section 2.4 of this document.

- If a voice message is sent as part of the post-call services, the voice message associated with the post-call service is transmitted using the file upload procedure in section 2.5.2 and download procedure described in section 2.5.5 of this document.
- For the transmission of information received from the content server and for the "note" information, the Enriched Calling mechanism as described in chapter 2.3 shall be employed.

2.5.1 Post-call procedures

The Post-call service will use the following elements and is defined as a separate service:

Element		Value / Description
Value carried in an Accept-Contact or Contact header		+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.Ims.icsi.gsma.callunanswered"
Value carried in a P-Preferred-Service or P-Asserted-Service header		urn:urn-7:3gpp-service.ims.icsi.gsma.callunanswered
Service Tuple		org.3gpp.urn:urn-7:3gpp-service.ims.icsi.gsma.callunanswered Version: 1.0
Service realisation type		Enriched Calling service
Enriched Calling service*	Media type	MSRP
	Auto accept	Yes
	Store & Forward	Not required

Table 11: Post-call service realisation

2.5.2 File upload for audio message

If an audio message is sent as part of a Post-call session, the file is first uploaded to the HTTP content server using Audio Messaging sender procedures (file upload) as described in section 3.11.4.2.2 of [PRD-RCC.07].

2.5.3 Session Management

If the client is authorized to offer the Post-call service according to the definition of the POST CALL AUTH parameter defined in Table 1 and the capability discovery is successful and, therefore the service is available to the user, the media session establishment, as described in section 2.3 using the Post Call ICSI shall commence when the user confirms to send a Post-Call note or an audio message.

The MSRP session is closed by the A-Party when the sending of the note or audio message has finished. In the normal case that means A-Party receives the MSRP 200 OK response.

In case the MSRP session was terminated due to an error message or reception of a SIP BYE message with the Reason header field containing the 503 (service unavailable) response code, the offline procedure will be triggered and the information will be sent using operator messaging.

2.5.4 Post-call elements

Service media specification for leaving a Post-call Note

Using the MSRP session, the service employs a XML-based protocol to convey the information to the receiver. The complete XML is presented below:

```
<?xml version="1.0" encoding="UTF-8"?>
<rcsenvelope xmlns="urn:gsma:params:xml:ns:rcs:rcs:calldata">
        <rcscalldata>
              <note>this is the note</note>
        </rcscalldata>
</rcsenvelope>
```

Table 12: Post-call Note service

The elements under rcscalldata are described below:

- <note> :

  Text to specify the reason for the missed call. This is similar to the subject in the call composer. The maximum length of the note should be 60 characters.
  This XML body shall be transported in the MSRP SEND message with content-type *application/vnd.gsma.encall+xml*.

Service media specification for leaving a voice message

After the upload procedure as described in section 2.4.2 was successful, the A-Party shall send a XML message to the receiver in the post call session:

```
<?xml version="1.0" encoding="UTF-8"?>
<file xmlns="urn:gsma:params:xml:ns:rcs:rcs:fthttp">
   xmlns:am=" urn:gsma:params:xml:ns:rcs:rcs:rram">
        <file-info type="file" file-disposition="[file-disposition]">
              <file-size>[file size in bytes]</file-size>
              <file-name>[original file name]</file-name>
              <content-type>[MIME-type for file]</content-type>
              <am:playing-length>[duration of the rram]</am:playing-length>
              <data url ="[HTTP URL for the file]" until="[validity of the file]"/>
        </file-info>
<file>
```

Table 13: Example of Audio Message Transfer using File Transfer via HTTP)

See section 3.11.4.2.2 of [PRD-RCC.07]

This XML body shall be transported in the MSRP SEND message with content-type application/vnd.gsma.rcs-ft-http+xml (IMDN not used).

The client shall encode the voice message using the Adaptive Multi-Rate (AMR) codec.

The voice message shall be formatted in the file format defined in RFC4867 as described in section 3.11.4.1 of [PRD-RCC.07].

The audio message shall not exceed a duration of 10 minutes and the resulting file size should not exceed the maximum file size as defined in FT MAX SIZE (as defined in Annex A of [PRD-RCC.07])

2.5.5 Receiver procedures

The receiver side will download the audio message automatically as soon as it has received the XML document containing the links.
For unanswered call components auto accept of the content should be used according to the network defined configuration and limits (usually auto accept is enabled up to a certain file size. Above this file size the user has to accept the download first).

The Post-call note information associated to an unanswered call shall be stored and associated to the last missed call in the call log from the calling line identity. The client shall do the matching of the received Post-call note to the earlier missed call based on the calling party identity provided for the missed call and the originator identity provided for the Post-Call SIP session in the same way as the matching for the incoming call to Call Composer information described in section 2.4.5.

There is one edge case that is not covered in this version of the specification. Assuming the A-Party makes a call towards the B-Party that has activated unconditional call forward towards the C-Party and the A-Party cancels the call before the C-Party answers. The “Post-call note” service will be proposed to the A party. Nevertheless the B-Party in that case does not have any call log entry for that call and will have to discard the received information.

2.5.6 Call forwarding behaviour

The receiver side may have one of the call forwarding services activated (e.g. call forward unconditional, call forward on busy) that may interact with the add reason service. The Interactions will be handled in the following way:

- The information associated to an unanswered call will only be stored in the call logs if there is an associated entry in the call log for this information. If due to call forward the call is never presented on the receiver side, the information will be discarded on B-Party side as if the call has never been placed by the A-Party side.

2.5.7 Legacy and offline support for Post-call Services

In case the B-Party is offline or not an Enriched Call user the Post-call Services should be offered on A-Party side by the client, using standard operator messaging instead.

2.6 XML Schema

The following is the complete XML Schema supporting the XML structures as described in Call Composer and Post Call service.

```xml
<?xml version="1.0" encoding="UTF-8"?>
<!--
  Enriched Calling document
    version - 2.1
    date    - 19 May 2015

FILE INFORMATION

   GSMA OIG Enriched Calling 1.0: Technical Specification Document
    File: OIG-SUP-XSD_Enriched_Calling-V2_1-20150519.xsd
    Type: Text - Schema Description
-->
<xs:schema
    targetNamespace="urn:gsma:params:xml:ns:rcs:rcs:calldata"
    xmlns="urn:gsma:params:xml:ns:rcs:rcs:calldata"
    xmlns:xs="http://www.w3.org/2001/XMLSchema"
    elementFormDefault="qualified" attributeFormDefault="unqualified">

    <!-- This import brings in the XML language attribute xml:lang -->
    <xs:import namespace="http://www.w3.org/XML/1998/namespace"
         schemaLocation="http://www.w3.org/2001/xml.xsd"/>

  <!-- The root "rcsenvelope" element -->
   <xs:element name="rcsenvelope">
     <xs:complexType>
       <xs:sequence>
         <xs:element name="rcscalldata" type="reasontype" maxOccurs="1"/>
         <xs:any minOccurs="0" maxOccurs="unbounded" processContents="lax"/>
       </xs:sequence>
     </xs:complexType>
   </xs:element>

  <!-- The definition of type "reasontype" is as below -->
   <xs:complexType name="reasontype">
     <xs:sequence>
       <xs:element name="subject" type="xs:string" minOccurs="0" maxOccurs="1"/>
       <xs:element name="importance" type="xs:boolean" minOccurs="0" maxOccurs="1"/>
       <xs:element name="location" type="xs:string" minOccurs="0" maxOccurs="1"/>
       <xs:element name="composerid" type="xs:string" minOccurs="0" maxOccurs="1"/>
       <xs:element name="picture_url" type="xs:string" minOccurs="0" maxOccurs="1"/>
       <xs:element name="note" type="xs:string" minOccurs="0" maxOccurs="1"/>
        <xs:any minOccurs="0" maxOccurs="unbounded" processContents="lax"/>
```



```
    </xs:sequence>
  </xs:complexType>
</xs:schema>
```

Table 14: Complete XML Schema supporting for the EnCall Call Composer and Post call service

2.7 Enriched call logs

This feature is implemented by the client and does not require any interaction with the network.

The client shall store on the A-Party and B-Party side:

- Call composer information in the event of an incoming call, if available
- Unanswered call information in case a call is not answered
- Information shared during a call: messages, files, shared maps, shared sketches and locations

All information will be mapped to existing call log entries. It is not intended to generate any new call log entry.

The client shall associate the call record (call log) with the data mentioned above based on the procedure defined in 2.4.5 and 2.5.5 of this document.

Since Enriched Calling makes use of operator messaging chat and file transfer procedures, any chat or file transfer during a call will be available to the user after the call in the messaging threads.

As the content shall also be made accessible from the call logs, the selection of the content is left to implementation choice. For instance:

The devices apply a filter using the call start and ending date to filter the messages attached to that specific call: any message with time stamp is included in the call time frame shall be selected for the call entries.

NOTE: Any message deleted from the messaging thread will no longer be accessible from the call log.

2.8 Video call

A direct entry point to a video call is out of scope.

2.9 In call sharing

In call sharing consists of independent features which will their own respective tags to identify the feature.

The following in call sharing features are possible:

- Live Video – Conversation Video or Video Share (as defined in RCC.07)
- Share any file (file transfer via http)
- Exchanging messages (1 to 1 chat)
- Location push (Geolocation PUSH during a call)

2.9.1 Video Share

Video Share shall be supported by the client according to [PRD-RCC.07], especially section 3.6.4.1.

Next to the mandatory codecs, it is recommended to support additional video formats providing different levels of quality and to use them in an adaptive fashion depending both on the terminal status and the network conditions/coverage.

As specified [RFC3264], formats must be listed in order of preference in the SDP media description. As such, additional codecs providing better quality than these mandatory ones should be listed in the SDP before the mandatory codecs.

Tag to identify video share during a call (as defined in [PRD-RCC.07]):

Element	Value / Description
Service extension tag (IARI)	+g.3gpp.cs-voice

Table 15:Video Share SIP OPTIONS tag

The client configuration parameter to allow Video Share is PROVIDE VS as defined [PRD-RCC.07].

When used in SIP OPTIONS exchanges these Video Share capabilities can only be sent during an active call and are included only if the exchange takes place between the users in the active call.

For the definitions for the use of Video Share to provide the Live Video service please refer to section 2.9.12.

2.9.2 Share any file during a call

In order to be compliant with the Image Share service, Image Share shall be supported by the client on the receiving side.

The Enriched Calling feature “share any file” will have a dedicated entry point during a call. The difference compared to the image share feature is:

- Share any file behaviour is similar than FT via http and continues even after the ongoing call has ended.
- Share any file will use the File Transfer via http method which is described in section 3.5.4.8 of [PRD-RCC.07]. This method will be used, even if file transfer takes place during a call.

If an incoming file transfer arrives from a user who is not in the call, the notification will not be displayed in the call screen.

Files exchanged during a call between the parties currently in a call shall be included in the messaging thread and shall also be included in the call logs for the corresponding contact.

2.9.3 Exchanging messages during a call

The feature to exchange messages during a call will be technically implemented by using the 1 to1 chat feature as described in [PRD-RCC.07].

If an incoming chat message arrives from a user who is not in the call, the notification will not be displayed in the call screen.

Messages exchanged during a call between the parties currently in a call shall be included in the messaging thread and shall also be included in the call logs for the corresponding contact.

2.9.4 Location push during a call

The feature to share the location during a call will be technically implemented by using the Geolocation push feature as described in chapter 3.10.4.1.1 of [PRD-RCC.07].

Tag to identify Geolocation services as defined in [PRD-RCC.07]:

Element	Value / Description
Service extension tag (IARI)	+g.3gpp.iari-ref="urn%3Aurn-7%3A3gpp-application.ims.iari.rcs.geopush"

Table 16: SIP OPTIONS tag for Geolocation push service

Location information shared during a call between the parties currently in a call shall also be included in the call logs for the corresponding contact.

Required network parameters:

The network parameter to enable share location during a call is

<parm name="geolocPushAuth" value="1" />

2.9.5 In call Receiver procedures

The information associated to an ongoing call shall be displayed during the call.

For in call sharing components auto accept of the content should be used according to the network defined configuration and limits (usually auto accept is enabled up to a certain file size. Above this file size the user has to accept the download first).

2.9.6 Legacy and offline support for In call Sharing

Legacy and offline support for in call sharing for files, location and messages will be offered as part of the standard messaging functionality. All content which is shared during a call with a legacy or offline user will be sent as a normal message. The B-Party will receive it via operator messaging.

2.9.7 Shared Map

Shared Map is an application that lets two users draw, share markers, view each other positions on a "shared" map.

2.9.7.1 Service realisation information

RCS Service		Tag
Value carried in an Accept-Contact or Contact header		+g.3gpp.icsi-ref=”urn%3Aurn-7%3A3gpp-service.ims.icsi.gsma.sharedmap”
Value carried in a P-Preferred-Service or P-Asserted-Service header		urn:urn-7:3gpp-service.ims.icsi.gsma.sharedmap
Service Tuple		org.3gpp.urn:urn-7:3gpp-service.ims.icsi.gsma.sharedmap Version: 1.0
Service realisation type		Enriched Calling service
Enriched Calling service*	Media type	MSRP
	Auto accept	No
	Store & Forward	No

Table 17: Shared Map service realisation information

2.9.7.2 XML Protocol

Shared Map uses an XML protocol to transmit data through a MSRP channel.

2.9.7.2.1 Document Type Definition (DTD)

This DTD contains the elements used by Shared Map.

```
<!DOCTYPE PROTOCOL[
    <!ELEMENT actions
      ((version|drawing|bounds|undo|marker|user|remove|anyelement)+,
      (anyelement)*)>
    <!ELEMENT close (anyelement?)>
    <!ELEMENT version (anyelement?)>
    <!ELEMENT drawing (points, anyelement?)>
    <!ELEMENT remove (id, anyelement?)>
    <!ELEMENT undo (anyelement?)>
    <!ELEMENT bounds (points, anyelement?)>
    <!ELEMENT marker (title?, snippet?, point, id, anyelement?)>
    <!ELEMENT user (point, anyelement?)>
    <!ELEMENT points (#PCDATA)>
    <!ELEMENT point (#PCDATA)>
    <!ELEMENT title (#PCDATA)>
    <!ELEMENT snippet (#PCDATA)>
    <!ELEMENT id (#PCDATA)>
    <!ELEMENT anyelement ANY>
    <!ATTLIST actions seq CDATA #REQUIRED>
    <!ATTLIST version id CDATA #REQUIRED>
    <!ATTLIST drawing color CDATA #REQUIRED>
    <!ATTLIST drawing width CDATA #REQUIRED>
    <!ATTLIST drawing erase CDATA “false”>
    <!ATTLIST points encoding CDATA “Base64”>
    <!ATTLIST point encoding CDATA “Base64”>
]>
```

2.9.8 Shared Sketch

Shared Sketch is an application that lets two users draw, add background images, change background colour on a “shared” canvas.

2.9.8.1 Service realisation information

RCS Service		Tag
Value carried in an Accept-Contact or Contact header		+g.3gpp.icsi-ref=”urn%3Aurn-7%3A3gpp-service.ims.icsi.gsma.sharedsketch”
Value carried in a P-Preferred-Service or P-Asserted-Service header		urn:urn-7:3gpp-service.ims.icsi.gsma.sharedsketch
Service Tuple		org.3gpp.urn:urn-7:3gpp-service.ims.icsi.gsma.sharedsketch Version: 1.0
Service realisation type		Enriched Calling service
Enriched Calling service*	Media type	MSRP
	Auto accept	No
	Store & Forward	No

Table 18: Shared Sketch service realisation information

2.9.8.2 Canvas area

The coordinate system in Shared Sketch describes a surface with a 1:1 aspect ratio. This allows different screen resolutions to have the same view of the canvas being shared.

2.9.8.3 XML Protocol

Shared Sketch uses an XML protocol to transmit data through a MSRP channel.

2.9.8.3.1 Document Type Definition (DTD)

This DTD contains the elements used by Shared Sketch.

```
<!DOCTYPE PROTOCOL[
    <!ELEMENT actions
      (version|drawing|undo|image|background_color|anyelement)+,
      (anyelement)*>
    <!ELEMENT version (anyelement?)>
    <!ELEMENT close (anyelement?)>
    <!ELEMENT drawing (points, anyelement?)>
    <!ELEMENT undo (anyelement?)>
    <!ELEMENT image (#CDATA)>
    <!ELEMENT points (#PCDATA)>
    <!ELEMENT background_color (anyelement?)>
    <!ELEMENT anyelement ANY>
    <!ATTLIST actions seq CDATA #REQUIRED>
    <!ATTLIST version id CDATA #REQUIRED>
    <!ATTLIST background_color color CDATA #REQUIRED>
    <!ATTLIST drawing color CDATA #REQUIRED>
    <!ATTLIST drawing width CDATA #REQUIRED>
    <!ATTLIST drawing erase CDATA “false”>
```

```
    <!ATTLIST points encoding CDATA "Base64">
    <!ATTLIST image encoding CDATA "Base64">
]>
```

2.9.9 Shared Map / Shared Sketch session management

2.9.9.1 Media Session Management

If Shared Map or Shared Sketch

- is enabled via SHARED MAP AUTH or SHARED SKETCH AUTH parameters respectively as defined in section 2.1.2
- and the capability discovery is successful for the active call based on the Service Identification defined in section 2.9.7.1 and 2.9.8.1 respectively

and therefore the service is available to the user, the client of the user initiating a Shared Map or Shared Sketch session shall establish a media session as described in section 2.3. When establishing the media session the Shared Map or Shared Sketch ICSI shall be used in the accept-contact and P-Preferred-Services headers depending on the type of session (see sections 2.9.7.1 and 2.9.8.1).

If a client receives an invitation for a Shared Map or Shared Sketch session and:

- it is not involved in a call then the client shall reject the session with a SIP 486 Busy Here response.
- it is involved in an active mobile terminated call and the calling line identity of the active call has been restricted then the client shall reject the media session with a SIP 486 Busy Here response.
- the media session matches the call but the user rejects the invitation to the Shared Map or Shared Sketch then the client shall reject the media session as defined for terminating nodes in section 2.3.1. On reception of a SIP error response the client of the initiating party shall act as defined for initiating nodes in section 2.3.1.
- the media session matches the call and the user accepts the invitation to the Shared Map or Shared Sketch this is indicated by a SIP 200 OK response. The clients shall act as defined in section 2.3.1.

The media session matches the call if the phone number in the P-Asserted-Identity header of the media session matches the identity of the other party in the active call as follows:

- for a mobile originated call the connected party number if available, otherwise the called party number.
- for a mobile terminated call the calling line identity.

Media sessions are used by the clients to manage a Shared Map or Shared Sketch session as defined in section 2.9.9.2.

The termination of the Shared Map or Shared Sketch session is initiated by the clients as defined in section 2.9.9.2. The media session shall be terminated by the client with the procedure defined in section 2.3.2.

If an error occur on the media session or the media session is closed while the Shared Sketch or Shared Map session remains active, the client should reconnect the media session as follows:

- If a client needs to transmit a Shared Map or Shared Sketch action it shall first terminate the failed media session if it is not yet terminated with the procedure defined in sections 2.3.2 and 2.3.4 without notifying the user and then establish a new media session as described in section 2.3. If the establishment of the media session succeeds, the Shared Map or Shared Sketch session shall be continued. If the establishment of the media session fails the client may retry. If the session establishment continues to fail the client may cache the actions of the user to be able to continue the session at a later stage or to inform the user the Shared Map or Shared Sketch session is currently not available.
- If the other party client receives an INVITE from the same user that he is already in a Shared Map or Shared Sketch session with, it shall consider the new INVITE as a reconnect attempt, accept it and continue to send and receive Shared Map or Shared Sketch session data via the new media session.
- If race conditions apply it is possible that both clients tries reconnect the media session at the same time. This will lead to two open sessions, which shall be kept by the clients for the Shared Map or Shared Sketch session. Both sessions can be used for sending and receiving Shared Map or Shared Sketch session data. However a Shared Map or Shared Sketch action shall only by sent via one media session in this case. If the Shared Map or Shared Sketch session is terminated the clients shall terminate both media sessions with the procedure defined in section 2.3.2.

The media session status does not alter the state of the Shared Map or Shared Sketch session defined in section 2.9.9.2.

2.9.9.2 Shared Map / Shared Sketch Session Management

Media sessions as defined in section 2.9.9.1 are used by the clients for a Shared Map or Shared Sketch session as follows.

A user can only be engaged with a specific contact at a given time in one Shared Map or Shared Sketch respectively, e.g. it is not possible to share more than one map or sketch with the same contact at the same time. To start a new Shared Map or Shared Sketch session the clients shall terminate first a possibly active Shared Map or Shared Sketch session and the underlying media session. It is possible though to have a Shared Map and a Shared Sketch with the same contact at a given time.

The client initiating the Shared Map or Shared Sketch shall open the Shared Map or Shared Sketch session by sending a <*version*> element with the highest supported value in the <*id*> attribute. On reception of the <*version*> element the other party client shall return a <*version*> element with the highest supported version in the <*id*> value. Both parties shall inspect the <*id*> attribute value received from the other party. Both clients shall then make use of the lowest version of the two values.

If the client initiating a Shared Map or Shared Sketch session by sending a <*version*> element does not receive a <*version*> element it shall send a <*close*> element and continue the termination of the Shared Map or Shared Sketch session as defined below. The client should inform the user that a Shared Map or Shared Sketch is currently not available.

If the client initiating a Shared Map or Shared Sketch session by sending a <*version*> element does receive a <*close*> element it shall send a <*close*> element and continue the termination of the Shared Map or Shared Sketch session as defined below. To cope with cases where the clients were out of sync due to connection issues, the initiating client shall automatically retry to establish the session. If this retry fails again it shall inform the user that the Shared Map or Shared Sketch session failed.

If the <*version*> element handshake succeeds the Shared Map or Shared Sketch session shall be considered as active and can be used to exchange service specific elements as defined in section 2.9.10.

If the call used for sharing is terminated or the user closes the Shared Map or Shared Sketch session or the user puts the call used for sharing on hold, the client handling the request shall first send all pending Shared Map or Shared Sketch service elements and then send the <*close*> element.

A client receiving the <*close*> element shall first send all pending Shared Map or Shared Sketch service elements and then return a <*close*> element to the initiator.

A client shall consider the Shared Map or Shared Sketch session as closed, if it has sent a <*close*> element and received a <*close*> element from the other party in any order and shall then initiate the termination of the media session as defined in section 2.9.9.1.

If the client initiating the termination of a Shared Map or Shared Sketch session by sending of a <*close*> element does not receive a <*close*> element it may retry to initiate session closure. If the session closure for the Shared Map or Shared Sketch session continues to fail it shall consider the session to be closed. The client shall then initiate the termination of the media session as defined in section 2.9.9.1.

If a client receives a Shared Map or Shared Sketch element other than <*version*> without having a Shared Map or Shared Sketch session active, it shall return a <*close*> element to the other party client and follow the procedure for Shared Map or Shared Sketch session termination defined above in the role of the initiating client.

If a client receives a <*version*> element with having a Shared Map or Shared Sketch session active, it shall return a <*close*> element to the other party client and follow the procedure for Shared Map or Shared Sketch session termination defined above in the role of the initiating client.

If a client receives a <*close*> element without having a Shared Map or Shared Sketch session active, it shall return a <*close*> element to the other party client and follow the procedure for Shared Map or Shared Sketch session termination defined above in the role of the other party client.

2.9.9.3 Timeout and Retry Handling

For the robustness of Shared Map and Shared Sketch implementations the client should implement a retry schema for the handing of media sessions as defined in section 2.9.9.1

and Shared Map and Shared Sketch session as defined in section 2.9.9.2 for cases where the transmission of a Shared Map or Shared Sketch element fails. A common retry schema may be based on the following concept:

- A request should be considered to be failed if the request could not be processed by the initiating client successfully after a timer "t" has expired. If a timeout happens the client should free up resources that are related to a failed request, e.g. cancel the failed request, if applicable.
- If for the failure of a request a retry is applicable, the client should retry by sending the same request again. The retry should follow the rules of timeout handling above. If the failure of the request persists the client should retry again, up to "n" times.
- If a request is processed successfully after retry the client shall continue with the normal operation as defined in section 2.9.9.2.
- If the last retry of the request fails the client should consider the transaction failed and continue with the error handling as defined in section 2.9.9.2.

The recommended values for retry handling are:

- t = 5 sec
- n = 2

If required, these values be may revised in RCS implementation guidelines.

2.9.9.4 Content type

The content type used for the Shared Map MSRP messages shall be *application/vnd.gsma.sharedmap+xml*.
This should be set accordingly in the SDP content as below.

- The *a=accept-types* shall only include the *application/vnd.gsma.sharedmap+xml* MIME type

The content type used for the Shared Sketch MSRP messages shall be application/vnd.gsma.sharedsketch+xml.
This should be set accordingly in the SDP content as below.

- The *a=accept-types* shall only include the *application/vnd.gsma.sharedsketch+xml* MIME type

2.9.10 Shared Map and Shared Sketch XML elements

2.9.10.1 Elements used by both Shared Map and Shared Sketch

2.9.10.1.1 <actions>

<*actions*> is the outermost element. It contains 1 or more elements of the other types.

Attributes:

- seq – A unique sequence number of the action.

The sender shall add a sequence number value for each <actions> it sends. The sender shall start for any new Shared Map or Shared Sketch session with sequence number 1 and increase it by 1 for each <actions> element it sends in this session. The receiver of the <actions> element shall check the value of the sequence number. If the sequence number has already been received by the client, then the <actions> element shall be ignored, otherwise it shall be processed.

Elements/PCDATA:

Can contain drawing, background_color, image, bounds, undo, marker, user or remove.

2.9.10.1.2 <version>

This tag is used to tell the other party what protocol version you want to use. This allows new features to be added but still maintaining backwards compatibility with older versions. It shall be sent directly after the session has been established by both parties. The lowest version shall be used by both parties, and the UI updated to remove unsupported features. For detailed procedures for Shared Sketch/Map session establishment refer to section 2.9.9.2.

If the only the basic features are supported the version should be set to 1.

Attributes:

- *id* – The protocol version suggested to be used by the sender. It shall contain an integer value. In this version of the specification only the value "1" is used.

Elements/PCDATA:

None

2.9.10.1.3 <close>

This tag is used to tell the other party that the session has ended. Any actions received after this shall be discarded. For detailed procedures for Shared Sketch/Map session closure refer to section 2.9.9.2.

Attributes:

None

Elements/PCDATA:

None

2.9.10.1.4 <drawing>

This tag describes a drawing.

Attributes

- *Width*: provides the width of the line used for the drawing (in dp) with float precision.

  When used for Shared Sketch, the value provides the width of the line normalized to the canvas width. It is represented as decimal number with "." as decimal mark.

When used in Shared Map, the attribute provides the width of the line in density independent pixels (dp) as a decimal number with "." as decimal mark. The receiver shall render the drawing in the current presentation of the map using the dp value provided in this attribute. Since the line width is not normalized to the map presentation of the clients, it is up to the client implementation how to scale the line width if the scale of the map changes.

- *Color*: describes the colour of the drawing in ARGB in hexadecimal presentation with a fixed length of 8 characters, i.e. with a value set for all channels (alpha, red, green, blue).
- *Erase*: a Boolean value that indicates whether this is an erase or a drawing.

  If the value is set to false it is interpreted as a drawing.
  If the value is set to true it is interpreted as an erase.
  If attribute is not present it is interpreted as being a drawing.

Elements/PCDATA:

- *Points*: A drawing must contain one and only one points tag.

2.9.10.1.5 <undo>

This tag represents an undo action. This notifies the application that the latest draw action from the sender shall be discarded.

Multiple UNDOs shall be supported by the clients.

If the client need to redo an action, it shall send the draw action again.

Attributes:

None

Elements/PCDATA:

None

2.9.10.1.6 <points>

The drawing is formed by a number of points as derived from the user input, e.g. by tapping on the canvas, moving and finally lifting his finger. A single point is represented by its x and y coordinate values based on the coordinate system derived from the outer tag.

The coordinate system in Shared Sketch describes a surface with a 1:1 aspect ratio and all coordinates are normalized.

In Shared Map the coordinates are latitude and longitude of the geographic coordinate system. A negative value represents the southern or western hemisphere respectively.

The tag encodes pairs of IEEE 754 double precision floating point numbers with x or latitude coordinates first. There should be at least 2 pairs. Big endian is used.

Attributes:

- *Encoding*: The transfer encoding used for points. The following values are defined:
  - Base64: base64 transfer encoding is used. Currently the only supported format

Elements/PCDATA:

Pairs of IEEE 754 double precision floating point numbers encoded with the encoding set in attributes.

The doubles are back to back with the x coordinate first.

2.9.10.2 Shared Map specific elements

2.9.10.2.1 <bounds>

This tag is used to notify the bounds of what the user is currently watching. The bounds indicate the diagonal of the map by means of the coordinates of the northeast and the southwest corners of the current map presentation.

Attributes:

- *Encoding*: The transfer encoding used for bounds. The following values are defined:
  - Base64: base64 transfer encoding is used. Currently the only supported format

Elements/PCDATA:

Two pair of IEEE 754 double precision floating point numbers encoded with the encoding set in attributes. The doubles are back to back with the x coordinate first.

2.9.10.2.2 <marker>

This tag is used to send a Map marker.

Attributes:

None

Elements/PCDATA:

- *title* (optional) – The title on the marker,
- *snippet* (optional) – The snippet on the marker,
- *point* (required) – The position of the marker,
- *id* (required) – The id of the marker.

2.9.10.2.3 <remove>

This tag is used to remove markers that has been sent

Attributes

None

Elements/PCDATA:

- *id* (required) – The id of the marker to remove

2.9.10.2.4 <user>

Describes where the sender is. Only used for shared map.

The client initiating a Shared Map session as defined in section 2.9.9.2 shall send the <*user*> element after the <*version*> handshake is completed. The receiving client shall use the location contained in the <*user*> element to display the shared map.

The receiving client shall send a <*user*> element to the sender if the user has allowed the share of his location within this shared map session. If the user has not allowed the share of his location or location information is not available the receiving client shall omit sending of a <*user*> element.

Attributes:

None

Elements/PCDATA:

- *Point* (required) – The position of the user

2.9.10.2.5 <title>

Describes the title or headline of a marker. Currently the first address line if the marker is manually placed or it might be the name of an establishment if the user has searched for the marker.

Attributes:

None

Elements/PCDATA:

The text to display as the title or headline of the marker.

2.9.10.2.6 <point>

This tag describes the location of the marker on the map by its latitude and longitude coordinates.

Attributes:

- *Encoding*: The transfer encoding used for point. The following values are defined:
  - Base64: base64 transfer encoding is used. Currently the only supported format

Elements/PCDATA:

Two IEEE 754 double precision floating point numbers encoded with the encoding set in attributes.

The doubles are back to back with the x coordinate first.

2.9.10.2.7 <snippet>

Contains the subtitle of a marker.

Attributes:

None

Elements/PCDATA:

Text to display as a subtitle when the marker is inspected more closely.

2.9.10.2.8 <id>

Contains text representation of a UUID as per [RFC4122]

Attributes:

None

Elements/PCDATA:

Must contain a 128 bit UUID in string representation as per [RFC4122]

2.9.10.3 Shared Sketch specific elements

2.9.10.3.1 <background_color>

This elements specifies the background colour of the canvas.

Attributes

- *Color*: describes the colour of the drawing in ARGB in hexadecimal presentation with a fixed length of 8 characters, i.e. with a value set for all channels (alpha, red, green, blue)

Elements/PCDATA:

None

2.9.10.3.2 <image>

Tag containing an image that will be set as background. Recommended size is less than 500 kB.

The image binary shall be encoded as jpeg or png. In absence of a *content-type* indication the receiving client needs to detect the type of encoding while rendering the image. It shall be included in the tag by use of base64 transfer encoding.

Both the sender and the receiver client shall arrange the image to "fit to screen", i.e. centred and the edge with the largest extent fitted to the size of the canvas.

Attributes:

- *Encoding*: The encoding used. Defaults to Base64. Currently the only supported format

Elements/PCDATA:

Base64 encoded image.

2.9.11 Shared Map and Shared Sketch Examples

2.9.11.1 Shared Map, sending a drawing

Input:

Point (lat, lng): 55.72689635634269,13.19581925868988

Point (lat, lng): 55.72689635634269,13.195833340287209

Point (lat, lng): 55.72688842645664,13.195840716362

Point (lat, lng): 55.726884272706144,13.195840716362

Point (lat, lng): 55.72687634281765,13.195847757160664

Point (lat, lng): 55.72686822412059,13.195847757160664

Output (what will be sent to the other party)


```
<?xml version='1.0' encoding='UTF-8' ?>

<actions>

<drawing color="FFFF00FF" width="0.006">

<points encoding="Base64">
QEvdCvCXCWZAKmRCbAAAAEBL3QrwlwlmQCpkRESAAABAS90KrhG8cEAqZEU8AAAAQEvdCos5n/R
AKmRFPAAAAEBL3QpItFGmQCpkRihAAABAS90KBJmMvUAqZEYoQAAA
</points>

</drawing>

</actions>
```


2.9.11.2 Shared Map, sending a Marker

Input:

Location of Marker: 55.90060839130326,12.698330841958523

Output:

```
<?xml version='1.0' encoding='UTF-8' ?>

<actions>

<marker>

<point encoding="Base64">QevzRyLBk0lAKWWLnsAAAA== </point>

<id>58818160-8b60-4ab9-8ae5-78feaed16343</id>

</marker>

</actions>
```


2.9.11.3 Shared Map, remove Marker


```
<?xml version='1.0' encoding='UTF-8' ?>

<actions>

<remove>

<id>58818160-8b60-4ab9-8ae5-78feaed16343</id>

</remove>

</actions>
```


2.9.11.4 Shared Sketch, drawing a line

2.9.12 Live Video

Live Video is an application that allows users during a voice call to add uni- or bi-directional video streaming.

2.9.12.1 Service realisation information

The Live Video service is realised via two different enablers:

- Conversational Video as per [PRD-IR.94]] via LTE or via EPC integrated Wi-Fi [PRD-IR.51], provided that:
  - both users are using the IMS Multimedia Telephony service and
  - both users are using a Conversational Video capable phone and
  - both users are authorized to use Conversational Video and
  - end-to-end availability of Conversational Video is indicated by capability exchange during call set-up or during the call.
- Video Share during a call as defined in [PRD-RCC.07] provided that
  - the conditions for Conversational Video service do not apply and
  - the clients are authorized to provide the Video Share service and
  - the Video Share service is available end-to-end.

RCS Service	Tag
Feature Tag value in Contact Header for IMS Multimedia Telephony Service	+g.3gpp.icsi-ref=”urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel” see
Feature Tag value in Contact Header for support of video as streaming media	video see [PRD-IR.94]
Feature Tag value in Contact Header for RCS Video Share Service	+g.3gpp.cs-voice see [PRD-RCC.07]

Table 19: Live Video Service identification summary

2.9.12.2 Service logic

In addition to the procedures described in the relevant specifications ([PRD-RCC.07], [PRD-IR.51], and [PRD-IR.94]) apply.

Annex A Call flow examples

A.1 Example 1: Call Composer: Subject and picture with updates, B can download all information before call starts ringing.

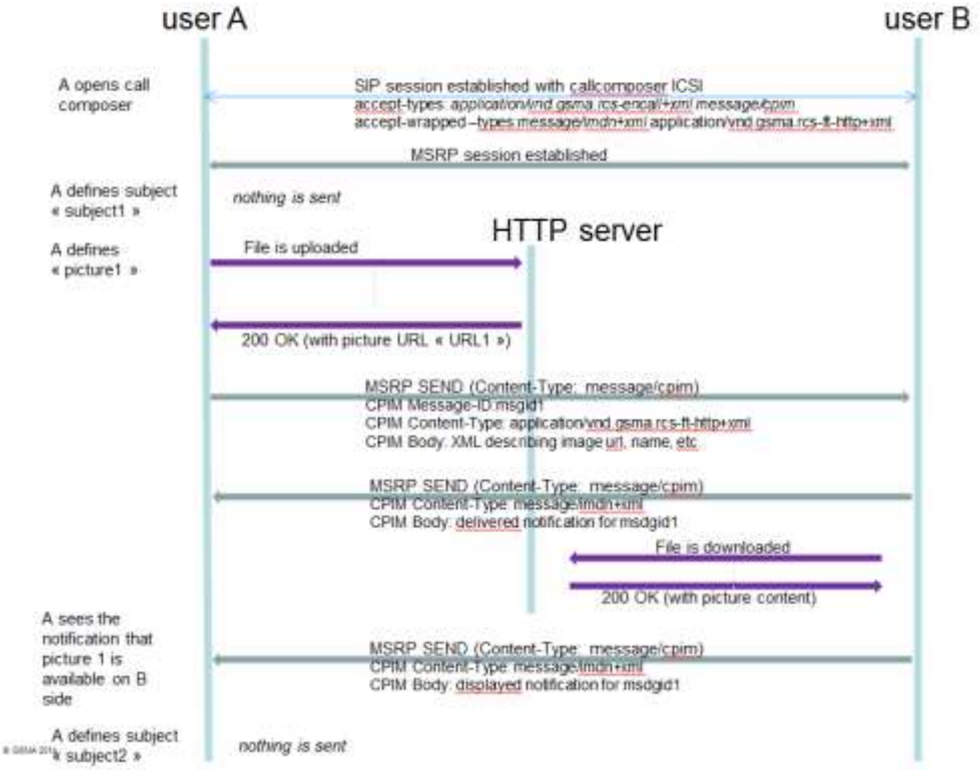
User A does the following action (in this order)

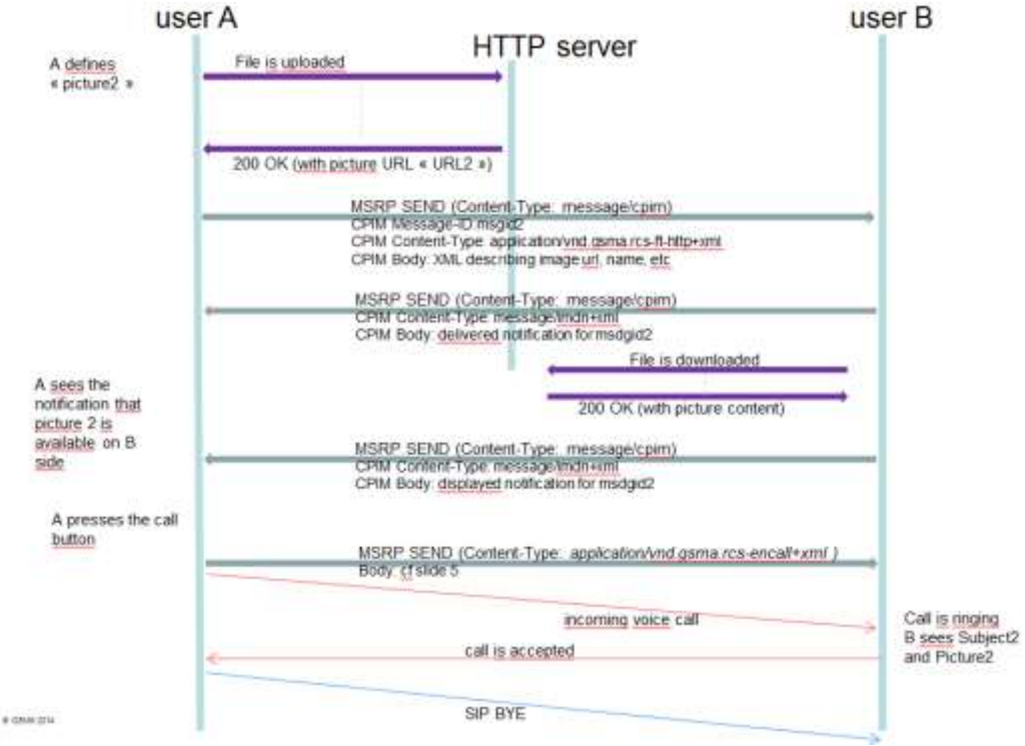
- The subject «subject1».
- The picture «picture1.jpg»
- Updates the subject with «subject2»
- Updates the picture with «picture2.jpg»
- User A presses the call button

Assumption: B has the time to download picture2 before the call is received

Body sent when pressing the call button

```
<?xml version="1.0" encoding="UTF-8"?>
<rcsenvelope xmlns="urn:gsma:params:xml:ns:rcs:rcs:calldata">
<rcscalldata >
   <subject>subject2</subject>
   <composerid>12345</composerid>
   <pictureurl>URL2</pictureurl>
</rcscalldata>
</rcsenvelope>
```



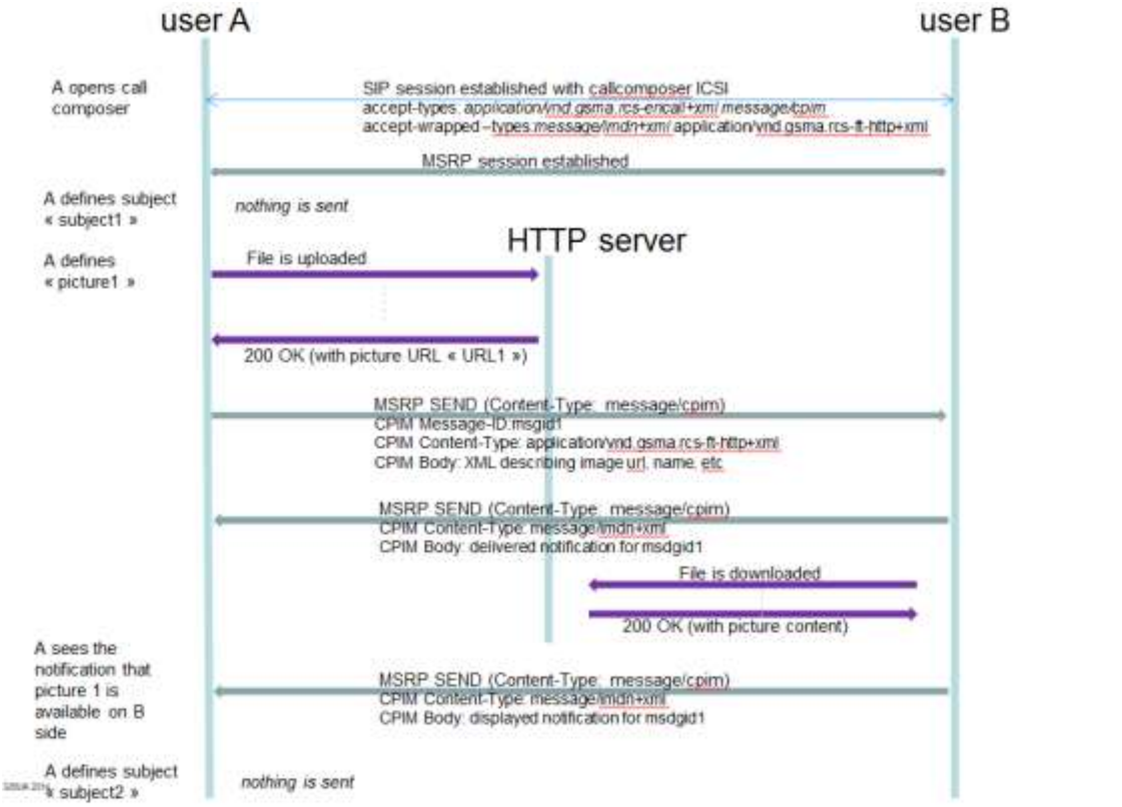


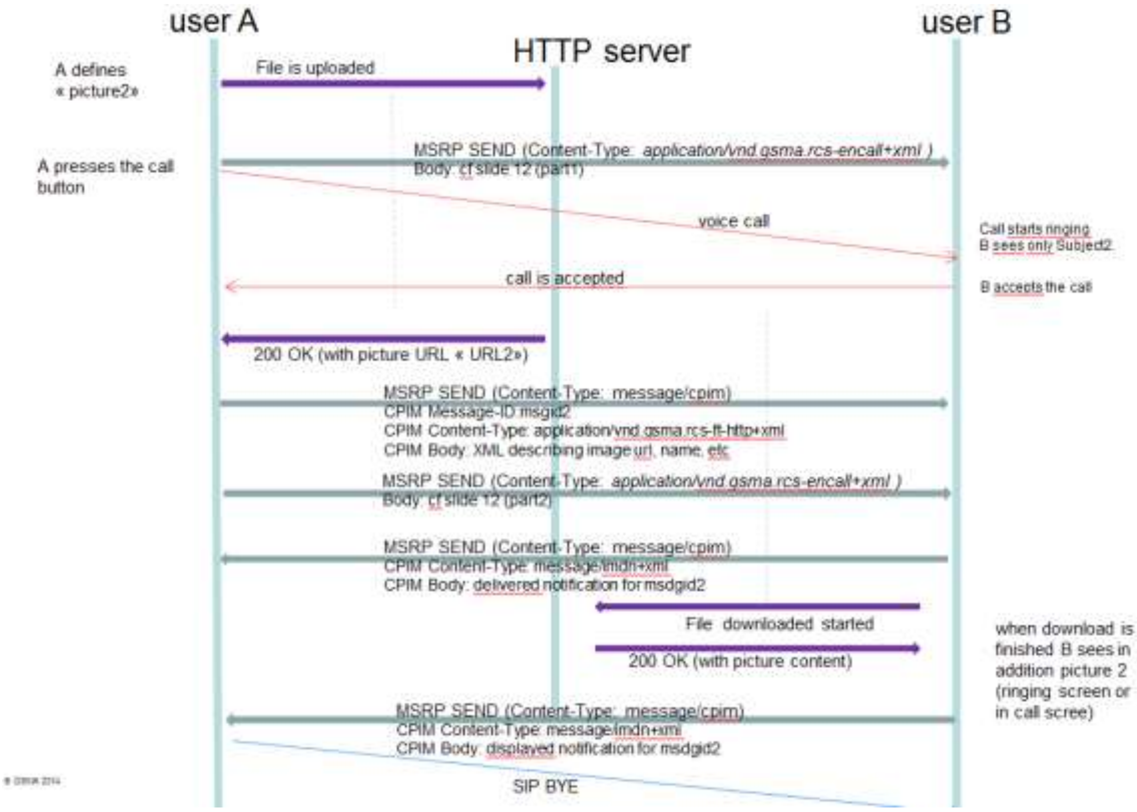
A.2 Example 2: Call Composer: Subject and picture with updates, B cannot download all information before call starts ringing.

User A does the following action (in this order)

- the subject « subject1 ».
- the picture « picture1.jpg »
- updates the subject with « subject2 »
- updates the picture with « picture2.jpg »
- user A presses the call button

Assumption: When A presses the call button, A has uploaded picture 2 but B Does NOT have the time yet to download picture2 at the moment the call starts ringing





A.3 Example 3: Call Composer: Subject and picture with updates, A is still uploading the picture when the call starts ringing.

User A does the following action (in this order)

- the subject « subject1 »
- the picture « picture1.jpg »
- updates the subject with « subject2 »
- updates the picture with « picture2.jpg »
- user A presses the call button

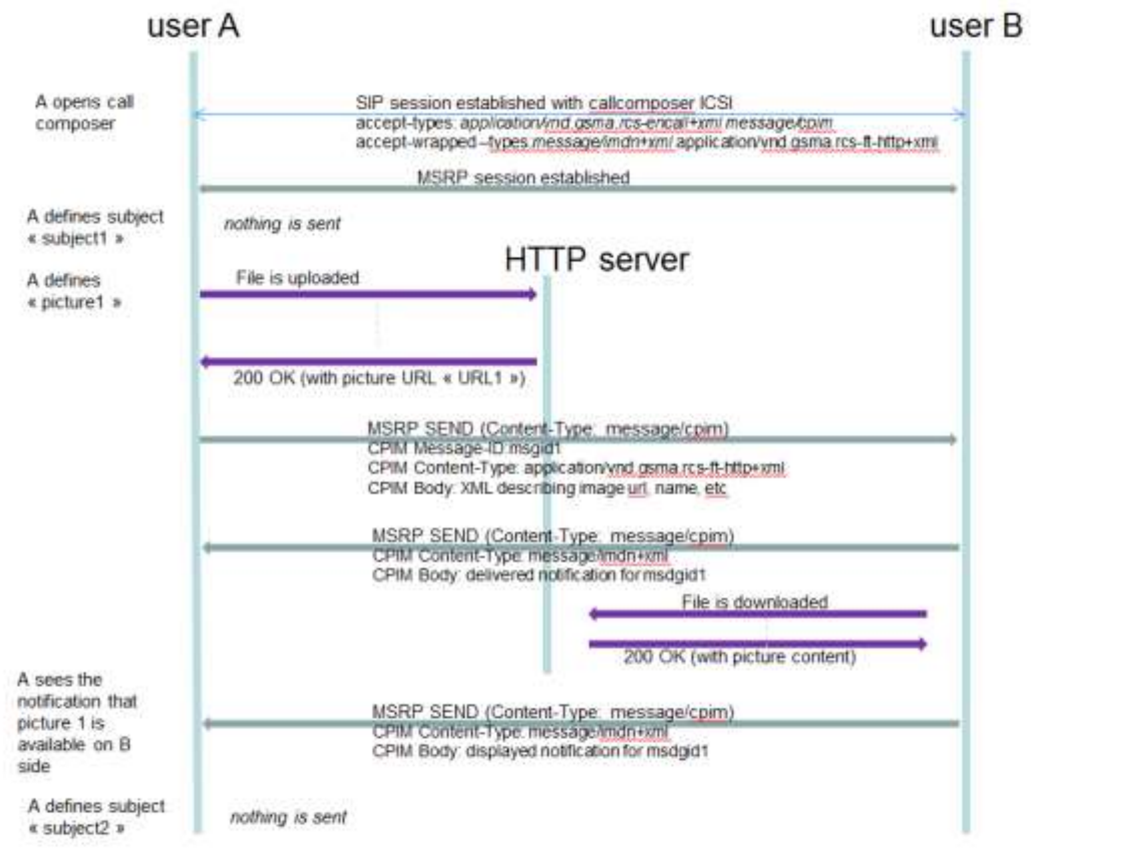
Assumption: A party is still uploading the picture2 when A presses the call button.

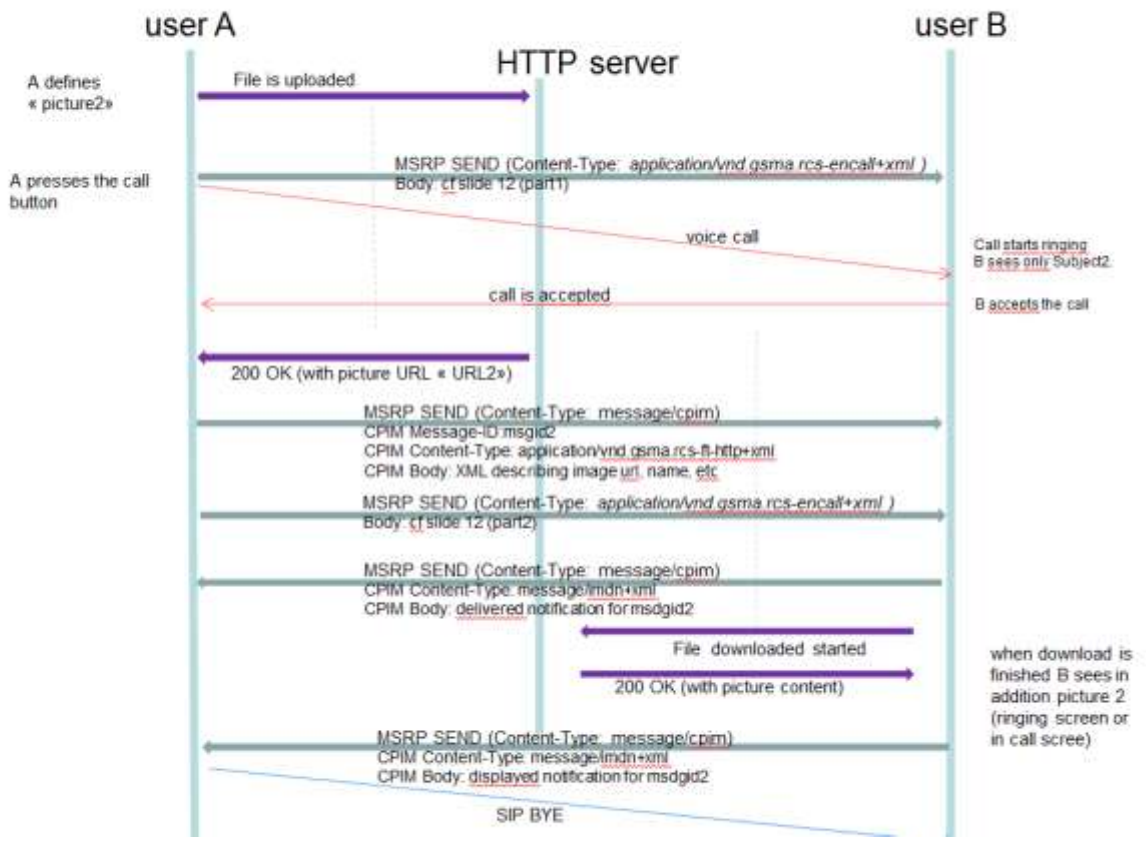
MSRP Send content

First MSRP SEND when A presses the call button

```
< ?xml version=on=n= encoding=ncoding=o
<rcsenvelope xmlns=murn :gsma :params :xml :ns :rcs :rcs :calldata”>
<rcscalldata >
   <subject>subject2</subject>
   <composerid>12345</composerid>
</rcscalldata>
</rcsenvelope>

Second MSRP SEND when A has finished to upload picture 2
< ?xml version=on=n= encoding=ncoding=o
<rcsenvelope xmlns=murn :gsma :params :xml :ns :rcs :rcs :calldata”>
<rcscalldata >
   <subject>subject2</subject>
   <pictureurl>URL2</file-info>
   <composerid>12345</composerid>
</rcscalldata>
</rcsenvelope>
```



user A
user B
A opens call composer
SIP session established with callcomposer ICSI
accept-types: application/vnd.gsma.rcs-encall+xml message/cpim
accept-wrapped –types message/imdn+xml application/vnd.gsma.rcs-ft-http+xml
MSRP session established
A defines subject « subject1 »
nothing is sent
HTTP server
A defines « picture1 »
File is uploaded
200 OK (with picture URL « URL1 »)
MSRP SEND (Content-Type: message/cpim)
CPIM Message-ID msgid1
CPIM Content-Type: application/vnd.gsma.rcs-ft-http+xml
CPIM Body: XML describing image url, name, etc
MSRP SEND (Content-Type: message/cpim)
CPIM Content-Type: message/imdn+xml
CPIM Body: delivered notification for msdgid1
File is downloaded
200 OK (with picture content)
A sees the notification that picture 1 is available on B side
MSRP SEND (Content-Type: message/cpim)
CPIM Content-Type: message/imdn+xml
CPIM Body: displayed notification for msdgid1
A defines subject « subject2 »
nothing is sent

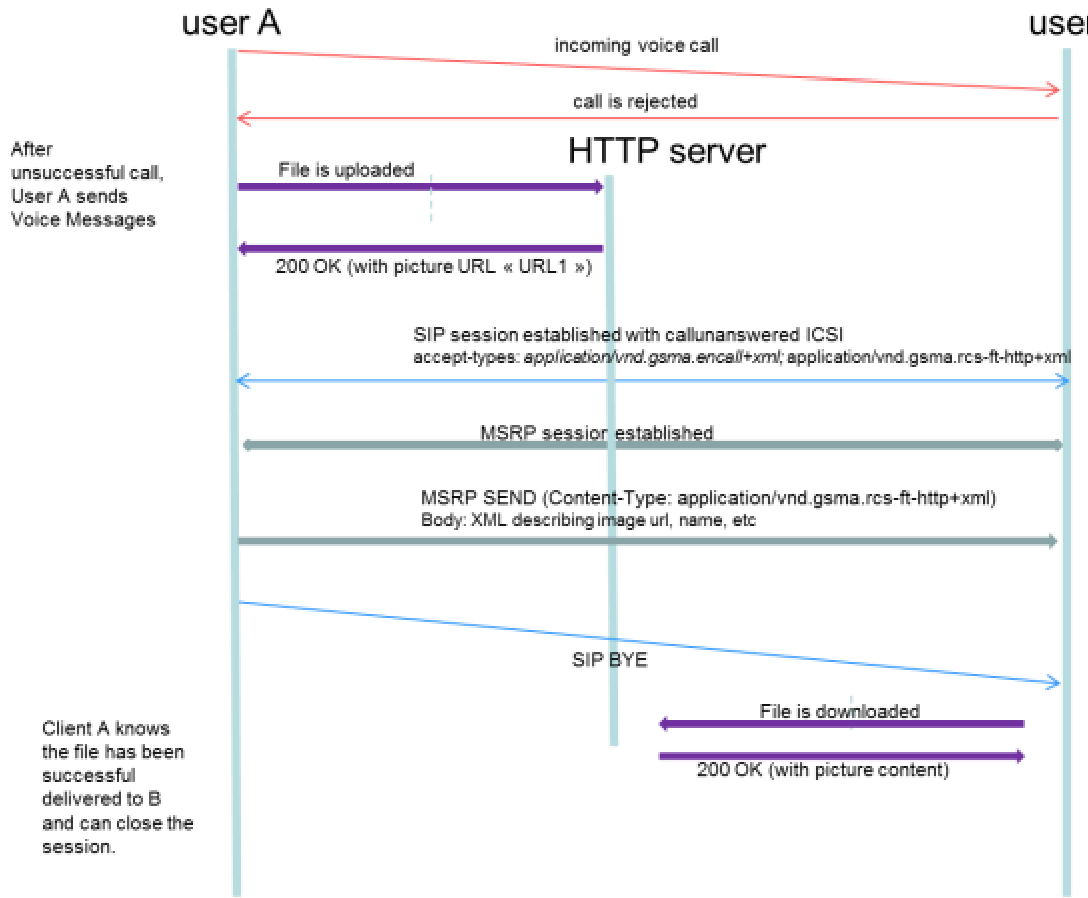

user A
user B
HTTP server
A defines « picture2»
File is uploaded
A presses the call button
MSRP SEND (Content-Type: application/vnd.gsma.rcs-encall+xml)
Body: cf slide 12 (part1)
voice call
Call starts ringing
B sees only Subject2
call is accepted
B accepts the call
200 OK (with picture URL « URL2»)
MSRP SEND (Content-Type: message/cpim)
CPIM Message-ID msgid2
CPIM Content-Type: application/vnd.gsma.rcs-ft-http+xml
CPIM Body: XML describing image url, name, etc
MSRP SEND (Content-Type: application/vnd.gsma.rcs-encall+xml)
Body: cf slide 12 (part2)
MSRP SEND (Content-Type: message/cpim)
CPIM Content-Type: message/imdn+xml
CPIM Body: delivered notification for msdgid2
File downloaded started
200 OK (with picture content)
when download is finished B sees in addition picture 2 (ringing screen or in call scree)
MSRP SEND (Content-Type: message/cpim)
CPIM Content-Type: message/imdn+xml
CPIM Body: displayed notification for msdgid2
SIP BYE

A.4 Example 4: Add reason after unanswered call.

User A does the following action

After unsuccessful call, User A sends Post-call Voice Message

User B is ON-LINE



Annex B Document Management

B.1 Document History

Version	Date	Brief Description of Change	Approval Authority	Editor / Company
1.0	13 July 2015	New document	PSMC	Andreas Brock / Vodafone
2.0	21 March 2016	Include approved CR1002	PSMC	Tom Van Pelt / GSMA

B.2 Other Information

Type	Description
Document Owner	Global Specification Group
Editor / Company	Tom Van Pelt / GSMA

It is our intention to provide a quality product for your use. If you find any errors or omissions, please contact us with your comments. You may notify us at prd@gsma.com

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