Enriched Calling Technical Specification

Version 5.0

06 December 2018

This is a Non-binding Permanent Reference Document of the GSMA

Security Classification: Non-confidential

Access to and distribution of this document is restricted to the persons permitted by the security classification. This document is confidential to the Association and is subject to copyright protection. This document is to be used only for the purposes for which it has been supplied and information contained in it must not be disclosed or in any other way made available, in whole or in part, to persons other than those permitted under the security classification without the prior written approval of the Association.

Copyright Notice

Disclaimer

Antitrust Notice

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.

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 and 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 call composer) the picture delivery notification is received by the A-Party.

Term	Description
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
Multimedia Telephony	An IMS based Multimedia Telephony service as defined in [PRD-IR.92], [PRD-IR.94], [PRD-IR.51] or [PRD-NG.110]
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.
PIDF	Presence Information Data Format
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]	GSMA PRD IR.51 - IMS Profile for Voice, Video and SMS over untrusted Wi-Fi access Version 6.0 01 May 2018 http://www.gsma.com/
5	[PRD-IR.92]	GSMA PRD IR.92 - IMS Profile for Voice and SMS Version 12.0 02 May 2018 http://www.gsma.com/
6	[PRD-IR.94]	GSMA PRD IR.94 - “IMS Profile for Conversational Video Service” Version 13.0 21 June 2018 http://www.gsma.com/
7	[PRD-NG.110]	GSMA PRD NG.110 - “Multi Device” Version 2.0 12 November 2018 http://www.gsma.com/
8	[PRD-RCC.07]	GSMA PRD RCC.07 - “Rich Communication Suite 9.0 Advanced Communications Services and Client Specification” Version 10.0 06 December 2018 http://www.gsma.com/
9	[RFC3261]	SIP (Session Initiation Protocol) IETF RFC http://tools.ietf.org/html/rfc3261
10	[RFC3264]	An Offer/Answer Model Session Description Protocol IETF RFC http://tools.ietf.org/html/rfc3264
11	[RFC3326]	The Reason Header Field for the Session Initiation Protocol (SIP) IETF RFC http://tools.ietf.org/html/rfc3326
12	[RFC3863]	Presence Information Data Format (PIDF) IETF RFC http://tools.ietf.org/html/rfc3863
13	[RFC4028]	The Session Timers in the Session Initiation Protocol (SIP) IETF RFC http://tools.ietf.org/html/rfc4028

Ref	Document Number	Title
14	[RFC4119]	A Presence-based GEOPRIV Location Object Format IETF RFC http://tools.ietf.org/html/rfc4119
15	[RFC4122]	The Universally Unique IDentifier (UUID) URN Namespace IETF RFC http://tools.ietf.org/html/rfc4122
16	[RFC4479]	A Data Model for Presence IETF RFC http://tools.ietf.org/html/rfc4479
17	[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
18	[RFC4975]	The Message Session Relay Protocol (MSRP) IETF RFC http://tools.ietf.org/html/rfc4975
19	[RFC5491]	GEOPRIV Presence Information Data Format Location Object (PIDF-LO) Usage Clarification, Considerations, and Recommendations IETF RFC http://tools.ietf.org/html/rfc5491
20	[RFC6442]	Location Conveyance for the Session Initiation Protocol IETF RFC http://tools.ietf.org/html/rfc6442
21	[GML]	OpenGIS® Geography Markup Language (GML) Implementation Specification, Version 3.1.1, OGC 03-105r1 http://www.opengeospatial.org/

2 Service realisation

2.1 Client configuration

HTTP(s) based client configuration as defined in section 2.3 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 based on an Enriched Calling MSRP session as per section 2.4.3. If set to **2** the Call Composer service based on Multimedia Telephony INVITE request as per section 2.4.4 is enabled. If set to **3**, both the Call Composer service based on an Enriched Calling MSRP session as per section 2.4.3 and Multimedia Telephony INVITE request as per section 2.4.4 are 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

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	int	Get, Replace

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

- Values: 0, 1, 2, 3

  0, Indicates that the Enriched Calling Call Composer service is disabled
  The SIP REGISTER for RCS Services shall not include the ICSI pertaining to the Enriched Calling Call Composer in its Contact header nor shall the SIP REGISTER

for Multimedia Telephony include the feature tag defined in section 2.4.4. Also, the ICSI, the feature tag or the corresponding Service Tuples 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 only based on an Enriched Calling MSRP session as per section 2.4.3
The SIP REGISTER for RCS Services shall include the ICSI pertaining to Enriched Calling Call Composer if used by installed applications in its Contact header. Also, the ICSI or the corresponding Service ID are included as part of capability discovery as defined in section 2.4.3. Enriched Calling Call Composer entry points will be displayed based on the capability discovery.
2, Indicates that the Enriched Calling Call Composer service is enabled only based on Multimedia Telephony INVITE request as per section 2.4.4
The SIP REGISTER for Multimedia Telephony Services shall include the feature tag defined in section 2.4.4 in the Contact header field. Also, the feature tag or the corresponding Service Tuple shall be included as part of capability discovery if in coverage conditions where Multimedia Telephony calling is available as defined in section 2.4.4. Enriched Calling Call Composer entry points will be displayed based on the capability discovery and coverage conditions.
3, Indicates that the Enriched Calling Call Composer service is enabled based on both an Enriched Calling MSRP session as per section 2.4.3 and the Multimedia Telephony INVITE request as per section 2.4.4
The SIP REGISTER for RCS Services shall include the ICSI pertaining to Enriched Calling Call Composer if used by installed applications in its Contact header and the SIP REGISTER for Multimedia Telephony Services shall include the feature tag defined in section 2.4.4 in the Contact header field. Also, the ICSI or the corresponding Service ID for the Enriched Calling Call Composer service based on an Enriched Calling MSRP session shall be included as part of capability discovery as defined in section 2.4.3. If in coverage conditions where Multimedia Telephony calling is available, also the feature tag or the corresponding Service Tuple for the Enriched Calling Call Composer service based on Multimedia Telephony INVITE request shall be included as part of capability discovery as defined in section 2.4.4. Enriched Calling Call Composer entry points will be displayed based on the capability discovery.

- Post-reconfiguration actions:
  - If the re-configuration transits from values 0 or 2 to values 1 or 3 the client shall re-register in the IMS used for RCS services to include the ICSI pertaining to the Enriched Calling Call Composer based on an Enriched Calling MSRP session as per section 2.4.3.
  - If the re-configuration transits from values 1 or 3 to values 0 or 2 the client shall re-register in the IMS used for RCS services to remove the ICSI pertaining to Enriched Calling Call Composer based on an Enriched Calling MSRP session as per section 2.4.3. The Enriched Calling Call Composer entry points will be no longer displayed.

    - If the re-configuration transits from values 0 or 1 to values 2 or 3 the client shall re-register in the IMS used for Multimedia Telephony Services to include the feature tag defined in section 2.4.4.
    - If the re-configuration transits from values 1 or 3 to values 0 or 2 the client shall re-register in the IMS used for Multimedia Telephony Services to remove the feature tag defined in section 2.4.4.

- 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”.

As indicated, all parameters shall be included in the Services MO 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 and media feature 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
- Share any file (file transfer via http)
- Share text during a call (1 to 1 chat)
- Location share (Geolocation PUSH during a call)

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

For Enriched calling services ICSI and media feature tags and service tuples definitions see sections 2.4.3, 2.4.4, 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 –as defined in section 2.9.12
- 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 (for the realisation based on an Enriched Calling session) and Post-Call sessions a client shall close an Enriched Calling session when it has been idle for longer than 180 seconds. 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.

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 for an RCS Enriched Calling Session

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 for an RCS Enriched Calling Session

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 2.11 of [PRD-RCC.07].

2.4 Call Composer

2.4.1 Call Composer Services and Selection Rules

This document defines two implementations of the Call Composer service. The client shall provide the Call Composer service using

- an Enriched Calling session as defined section 2.4.3
  - if the value of the configuration parameter COMPOSER AUTH defined in section 2.1.2 is set to "1", and, if based on the capability discovery, the called B-party supports the Call Composer service using Enriched Calling session, or
  - if the value of the configuration parameter COMPOSER AUTH defined in section 2.1.2 is set to "3" and
    - if a Multimedia Telephony service is not available for the call, or
    - if, based on the capability discovery, the called B-party supports the Call Composer service using Enriched Calling session only.
- the Multimedia Telephony session with Enriched Calling Composer elements as defined in section 2.4.4
  - if the value of the configuration parameter COMPOSER AUTH defined in section 2.1.2 is set "2" or "3", and
  - if a Multimedia Telephony service is available for the call, and
  - if, based on capability discovery, the B-party supports the Call Composer service using a Multimedia Telephony session.

2.4.2 Picture transfer procedures

If a picture is sent as part of the Call Composer service, 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 3.2.5.3.1.1 of [PRD-RCC.07].

If the file upload cannot be completed the client MAY apply the Upload Resume procedure defined in section 3.2.5.3.1.2 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 Call Composer procedures using an Enriched Calling Session

The Call Composer service using an Enriched Calling session will use the following elements and is defined as a separate service:

<table>
<tr><th colspan="2">Element</th><th>Value / Description</th></tr>
<tr><td colspan="2">Value carried in an Accept-Contact or Contact header</td><td>+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.gsma.callcomposer"</td></tr>
<tr><td colspan="2">Value carried in a P-Preferred-Service or P-Asserted-Service header</td><td>urn:urn-7:3gpp-service.ims.icsi.gsma.callcomposer</td></tr>
<tr><td colspan="2">Service Tuple</td><td>org.3gpp.urn:urn-7:3gpp-service.ims.icsi.gsma.callcomposer
Version: 1.0</td></tr>
<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 6: Enriched Calling Composer service realisation summary

2.4.3.1 Session management

If the Call Composer service via the Enriched Calling session is selected as defined in section 2.4.1 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 2.4.3.

A client receiving an invitation for a Call Composer Enriched Calling 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

NOTE: 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.

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 180 seconds.

For a client compliant to this version of the specification, the importance call composer element cannot be changed by the A-party after the call has been initiated. However, a client compliant to an earlier version of this specification may change the call importance, therefore the client shall support receiving a changed call importance before the call is answered.

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 2.4.3.

In accordance with the receiver procedures in section 2.4.3.3 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 2.4.3.2 of this document.

2.4.3.2 Call Composer elements

Using the Enriched Calling session established for the Call Composer, the service employs an 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 Enriched Calling 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 7: XML message based on HTTP Content Server response (see the File Transfer procedure in section 3.2.5.3.1 of [PRD-RCC.07])

This XML body shall be carried in a CPIM body even though delivery and displayed notifications are no longer required (see section 2.4.3.4). 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 XML message, 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]"/>
        </rcscalldata>
```

```
</rcsenvelope>
```

Table 8: Call composer service using Enriched Calling session; Complete XML example

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 elements of rcscalldata (this is to secure future service extensibility).
The elements under rcscalldata are described below:

- <subject>: The maximum length of the subject field shall 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>: provides the characteristics of the picture file on the FT content server. The element includes an attribute "url" containing 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 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]"/>
    </rcscalldata>
</rcsenvelope>
```

Table 9: complete XML example after the call button has been pressed

The content of the XML only includes the picture URL for outgoing requests after the call button has been pressed, and may include either or both the importance element and the picture element for incoming requests. The picture URL replaces any value that may have been included in the previous XML sent for this call. The composerid is always mandatory. The composerid value shall be set to the same value that was used in the Call Composer data which had been sent previously for the same Enriched Calling session.

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

2.4.3.3 Receiver procedures

After the Call Composer Enriched Calling session is established, the receiving party 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 for 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 via Enriched Calling session service, 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 3.2.5.3.2 and 4.1.15.4 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.

2.4.3.4 Delivery and display notifications

When a Call Composer Enriched Calling session is set up, the pre-call picture file information, as per Table 7 in section 2.4.3.2, is transported in the Enriched Calling session.

A client compliant to this version of the specification shall not request IMDNs for the MSRP SEND request carrying this file information.

If the request arrives from a client supporting an earlier version of this specification, the MSRP SEND request carrying this file information contains a request to receive an Instant Messaging Disposition Notification (IMDN) ‘delivery’ notification, and a request to receive an IMDN ‘display’ notification.

When IMDNs are requested by a client compliant to an earlier version of this specification, the receiving devices must generate an MSRP SEND request containing the IMDN delivered status when the picture file information is delivered and another MSRP SEND request containing the IMDN displayed status when the picture is downloaded.

If the B-Party has delivered / displayed notifications to be sent after the Enriched Calling session has been torn down, the B-Party sends them through SIP MESSAGE requests using the Call Composer via Enriched Calling session ICSI (similar to what is done for File Transfer).

For more details see also Annex A.

2.4.3.5 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).

2.4.3.6 Legacy support for Call Composer via Enriched Calling session

Call Composer via Enriched Calling session will not have any legacy support since it can only be used if both sides support Call Composer via an Enriched Calling session.

2.4.4 Call composer procedures using the Multimedia Telephony session with Enriched Calling Call Composer elements

The Call Composer service using the Multimedia Telephony session with Enriched Calling Call Composer elements is based on [PRD-IR.92], [PRD-IR.51], [PRD-IR.94] and [PRD-NG.110] and will use the following elements:

Element	Value / Description
Value carried in Contact header of SIP REGISTER request for Multimedia Telephony	In addition to the values defined for the Multimedia Telephony service: +g.gsma.callcomposer
Value carried in Accept-Contact in SIP INVITE request for Multimedia Telephony	Same as defined for the Multimedia Telephony service NOTE: No need to include the Call Composer media feature tag since there is no requirement to limit the forking on the terminating side to only devices that support call composer.
Values carried in Contact header of SIP INVITE request and 200 OK response for Multimedia Telephony	In addition to the values defined for the Multimedia Telephony service:

	+g.gsma.callcomposer
Values carried in Contact header in SIP OPTIONS request and 200 OK response	In addition to the values defined for the Multimedia Telephony service: +g.gsma.callcomposer
Value carried in P-Preferred-Service or P-Asserted-Service header	urn:urn-7:3gpp-service.ims.icsi.mmtel
Service Tuple	org.3gpp.urn:urn-7:3gpp-service.ims.icsi.gsma.callcomposer Version: 2.0
Service realisation type	Multimedia Telephony service with Enriched Calling Call Composer elements

Table 10: Multimedia Telephony service with Enriched Calling Call Composer elements

2.4.4.1 Multimedia Telephony session with Enriched Calling Call Composer elements

If the Call Composer service via the Multimedia Telephony session is selected as defined in section 2.4.1 and the user enters a call composer screen associated with this recipient address then the user can set the importance of the call, the subject of the call, a location and a picture.

The client shall store the values of the call composer elements set by the user in the call composer screen.

When the call button is pressed and the client is setting up the call as defined for the applied Multimedia Telephony service, then the client shall include the relevant Call Composer elements in headers of the SIP INVITE request for Multimedia Telephony as defined in section 2.4.4.2 and identify the call as specified in Table 10.

Once the Multimedia Telephony session invitation is sent, the values of the Call Composer elements cannot be changed anymore.

Call Composer elements shall be stored in the A-party call log as soon as the SIP session has been initiated. Call composer elements shall be stored in the B-party call log as soon as the Multimedia Telephony session invitation has been received.

2.4.4.2 Call Composer Elements

The same SIP INVITE request used to set-up the call for a Multimedia Telephony call shall be used for the Call Composer elements subject, importance, location and picture for Enriched Calling services.

The list of headers for the SIP INVITE to carry the Call Composer elements is shown in Table 11:

Call Composer Element	SIP INVITE Header	Details
Importance	Priority	Format: As defined in [RFC3261] Section 20.26. Specifically, Priority value of “normal” will be

		mapped to “standard” call and Priority value of “urgent” will be mapped to “important” call Usage: Optional. May be present if the Priority is set to "normal" and shall be present if Priority is set to "urgent"
Subject	Subject	Format: As defined in [RFC3261] Section 20.36 Usage: Optional. Shall be present if the user has set a subject for the call. Length: As defined in section 2.4.3.2
Picture	Call-Info	Format: As defined in [RFC3261] Section 20.9 and in the procedure for sending a picture below. The “info” and “card” parameters are not applicable for Enriched Calling. Usage: Optional. Shall be present if the user has set a picture for the call.
Location	Geolocation, Geolocation-Routing	Format: As defined in [RFC6442] and in the procedure for setting of the location below Usage: Optional. Shall be present if location is set by the user

Table 11: Call Composer elements in Headers of the SIP INVITE for Multimedia Telephony

If the user sets a picture for the call, then the client shall invoke the picture transfer procedures defined in section 2.4.2. The Call-Info header value shall be set to the value of the "url" attribute of the "data" element of the "file-info" element in the XML returned from the Content Server and shall include the “purpose=icon” header parameter as described in [RFC3261].

If the user sets a location for the call, then the client shall

- create a Presence Information Data Format (PIDF) document following the definitions of [RFC3863],
- add to the PIDF document a "presence" element with an "entity" attribute containing the tel or sip URI of the originating user,
- add to the "presence" element a "person" element following the requirements of [RFC4479] with an "id" attribute containing a value satisfying the constraints of the xsd type ID,
- add to the "person" element a "geopriv" element following the requirements of [RFC4119],
- add to the "geopriv" element an empty "usage-rules" element,
- add to the "geopriv" element a "location-info" element.
- represent in the "location-info" element either an exact position by providing a [GML] <point> element or an inaccurate position as a [GML] <circle> element, both referring to the European Petroleum Survey Group EPSG::4326 spatial reference schema as described in [RFC5491]. The coordinates of either the centre of this circle or the exact position will be represented with a single [GML] <pos> element with the actual coordinates as value. The radius of the circle will be represented in meters, which will

be indicated by setting the unit of measure attribute of the radius element to the value of EPSG::9001 as described in [RFC5491].

- include the PIDF document in a SIP INVITE body with the Content-Type "application/pidf+xml" and a Content-ID URL,
- add to the SIP INVITE a Geolocation header containing the Content ID URL of the PIDF document body as defined in [RFC6442],
- add to the SIP INVITE a Geolocation-Routing header as defined in [RFC6442] with value set to "no", unless a service interaction as described below applies.

If there are interactions with other geolocation services adding location information for the call, then the location information shall be added to the SIP INVITE request considering the rules defined in [RFC5491] for multiple location objects. If another geolocation service requires the use the geolocation information in the network for routing, then the Geolocation-Routing header value shall be set to "yes".

```
INVITE tel:+491715551212 SIP/2.0
…
Subject: This is an example!
Priority: urgent
Call-Info: <ftcontentserver.rcs.mnc001.mcc262.pub.3gppnetwork.org/dl?uid=1234>;purpose=icon
Geolocation: <cid:some_id@server.com>
Geolocation-Routing: no
Content-Type: multipart/mixed; boundary=boundary1

--boundary1
Content-Type: application/sdp

  [Session Description Protocol (SDP) goes here]

--boundary1
Content-Type: application/pidf+xml
Content-ID: <some_id@server.com>

<presence xmlns="urn:ietf:params:xml:ns:pidf"
xmlns:dm="urn:ietf:params:xml:ns:pidf:data-model"
xmlns:gp="urn:ietf:params:xml:ns:pidf:geopriv10"
xmlns:gml="http://www.opengis.net/gml"
xmlns:gs="http://www.opengis.net/pidflo/1.0"
entity="tel:+491711234567">
  <dm:person id="sh2204">
    <gp:geopriv>
        <gp:location-info>
          <gs:Circle srsName="urn:ogc:def:crs:EPSG::4326">
              <gml:pos>47.577866 -122.164080</gml:pos>
              <gs:radius uom="urn:ogc:def:uom:EPSG::9001">30</gs:radius>
          </gs:Circle>
        </gp:location-info>
        <gp:usage-rules/>
    </gp:geopriv>
  </dm:person>
</presence>

--boundary1--
```

Table 12: Call Composer service using Multimedia Telephony session; Complete example

An application server may choose to remove Priority, Subject, Call-Info and Geolocation headers and the PIDF document body if the terminating party does not support Call Composer capabilities based on the Call Composer registration tag in the Contact header.

In case of a file upload failure in the Call Composer procedures, or if coverage changes while in the Call Composer screen, the voice call shall not be impacted.

2.4.4.3 Receiver procedures

The receiving party shall display all available information for the call.

In case a picture has been sent as part of Call Composer, the receiver side shall download the Call Composer picture file automatically as soon as it has received the link in the call request. For download of the Call Composer picture the procedures in sections 3.2.5.3.2 and 4.1.15.4 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.

2.4.4.4 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 the Call Composer service. The interactions will be handled in the following way:

- In case of call forward towards another number (C-party) that supports Enriched Calling, the Call Composer information will be shown on the receiver side assuming the call stays within the IMS domain
- In case of call forward towards another number (C-party) that does not support Enriched Calling, the Call Composer information will not be shown on the receiver side even when the call stays within the IMS domain
- Call composer information shall always be stored in the B-party call logs if Call Forward Not Answered is set up and also in the call logs of the party to whom the call was forwarded if the call is not answered and forwarded
- Call composer information shall be stored in the forwarded party call logs if Unconditional Call Forwarding is setup

NOTE: Removal of Call Composer information when terminating network does not support it, may be provided at NNI.

2.4.4.5 Caller Line Identification Restriction (CLIR)

If a client does not receive calling party identification with the received call (e.g. due to calling line restriction services), then the Pre-call Call Composer information should also not be displayed.

NOTE: Removal of Call Composer information when calling party identification is not present may also be provided by the terminating network.

2.4.4.6 Legacy support for Call Composer via Multimedia Telephony session

The Call Composer headers defined in section 2.4.4.2 can only be used in a Multimedia Telephony session if both sides support the service.

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 13: 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 3.2.7.2.2 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 14: 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 15: Example of Audio Message Transfer using File Transfer via HTTP)

See section 3.2.7.2.2 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 3.2.7.1 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 2.4.3.3.

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 shall be offered on A-Party side by the client based on the Post-call service under consideration. The Post-Call Note information shall be sent by extracting the text included in the note element of the Post-Call xml and using the selected messaging technology based on the selection rules defined in section 3.2.1 of [PRD-RCC.07]. The Post-Call Voice Message shall be sent using the selected file transfer technology based on the selection rules defined in section 3.2.5.7 of [PRD-RCC.07].

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"?>
<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" type="xs:string" minOccurs="0" maxOccurs="1">
        <xs:complexType>
          <xs:attribute name="url" type="xs:anyURI" use="required"/>
          <xs:anyAttribute namespace="##other" processContents="lax"/>
        </xs:complexType>
      </xs:element>
      <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 16: Complete XML Schema supporting for the Enriched Calling 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 2.4.3.3 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 Void

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 –as defined in section 2.9.12
- Share any file (file transfer via http)
- Exchanging messages (1 to 1 chat)
- Location push (Geolocation PUSH during a call)

2.9.1 Void

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 service that is described in section 3.2.5 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 3.2.6 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 17: 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 18: 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 19: 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 in-call 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.

To initiate the in-call capability discovery request and to establish the media sessions for a Shared Map or Shared Sketch, the client shall derive the target address for the SIP signalling from the other party of the active call in accordance with the definitions in section 2.5.3.2 of [PRD-RCC.07].

The client shall match the phone number in the P-Asserted-Identity header of received in-call capability discovery requests or a media session for a Shared Map or Shared Sketch with an active ongoing call in accordance with the definitions in section 2.5.2.2 of [PRD-RCC.07].

For both cases, to identify the other party of the call, the client shall use

- 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.

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 the user of the client initiating the media session aborts the session prior to finalisation of the INVITE transaction, then the client shall cancel the INVITE request by sending a CANCEL request according to the rules and procedures of [3GPP TS 24.229].

If the call matching the Shared Map or Shared Sketch session is terminated by the user of the client initiating the media session or is put on hold prior to finalisation the INVITE transaction, then the client shall cancel the media session by sending a CANCEL request according to the rules and procedures of [3GPP TS 24.229].

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.
- if the media session matches the call, the client shall send a SIP 180 Ringing response and notify the user about the incoming Shared Map or Shared Sketch session. The client receiving the invitation for the media session shall start a timer "s" for the INVITE transaction. Subsequently,

    - if 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 and stop the timer "s" for the INVITE transaction. 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.
    - if the user accepts the invitation to the Shared Map or Shared Sketch, then the client shall accept the media session as defined for terminating nodes in section 2.3.1 and stop the timer "s" for the INVITE transaction.
    - if the call matching the Shared Map or Shared Sketch session is terminated by the user of the client terminating the media session or if the user puts the call on hold, the client shall reject the media session with a SIP 486 Busy Here response and stop the timer "s" for the INVITE transaction.
    - if the client receives a CANCEL request for the Shared Map or Shared Sketch session, then the client shall process the CANCEL request according to the rules and procedures of [3GPP TS 24.229] and stop the timer "s" for the INVITE transaction.
    - if the timer "s" for the INVITE transaction expires, then the client shall reject the media session with a SIP 408 Request Timeout response.

    The recommended value for the timer "s" is:

    - s = 30 sec.

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 an established 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 occurs for 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 try reconnecting 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/RAKmR
FPAAAAEBL3QpltFGmQCpkRihAAABAS90KBJmMvUAqZEYoQAAA
</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>
```

2.9.12 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 uni-directional or bi-directional 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.

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 [PRD-IR.92]
Feature Tag value in Contact Header for support of video as streaming media	video see [PRD-IR.94]

Table 20: Live Video Service identification summary

2.9.12.2 Service logic

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.

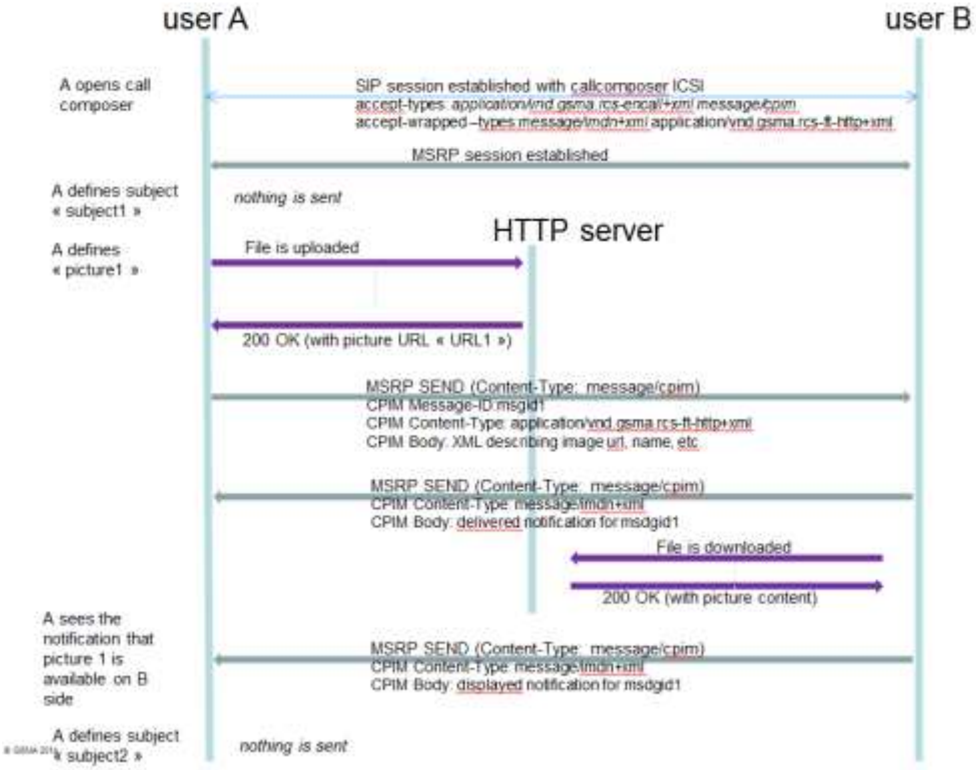
User A does the following action (in this order)

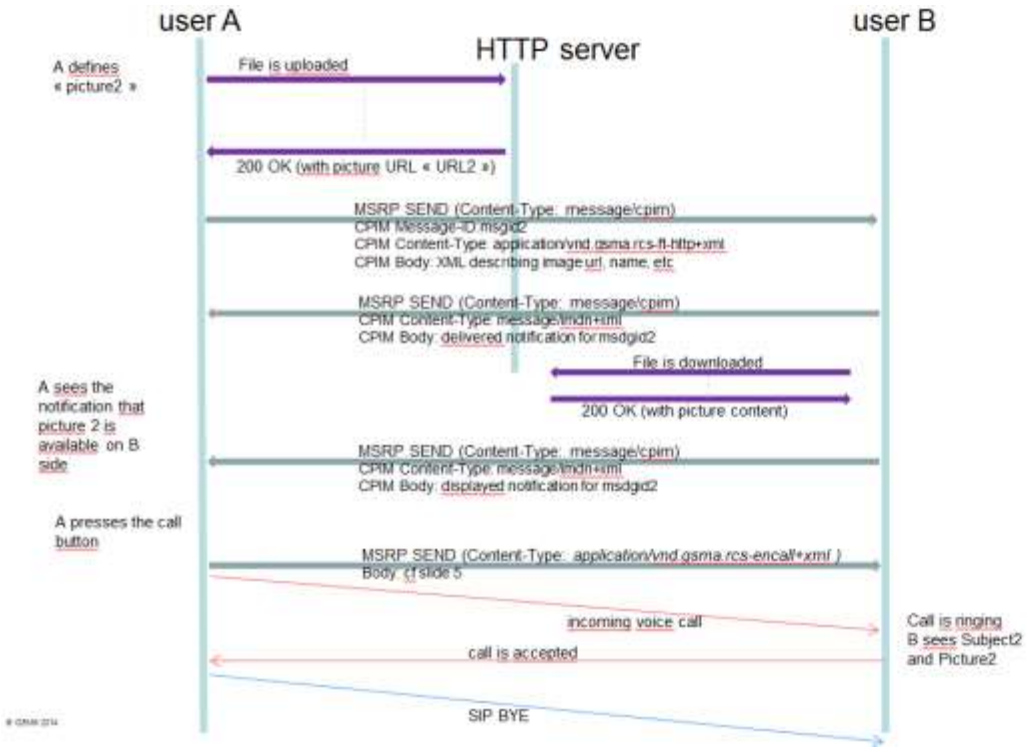
- The subject «subject1».
- The picture «picture1.jpg»
- Updates the subject with «subject2»
- Updates the picture with «picture2.jpg»
- User A presses the call button

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>
```



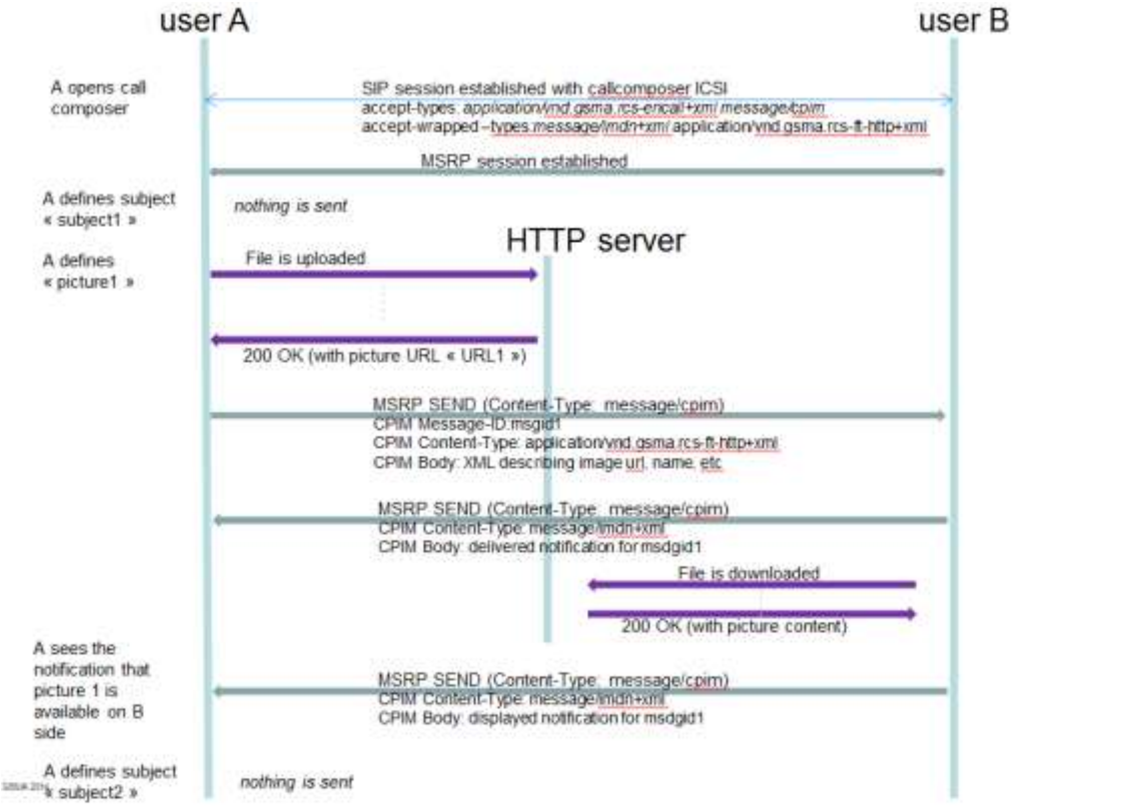


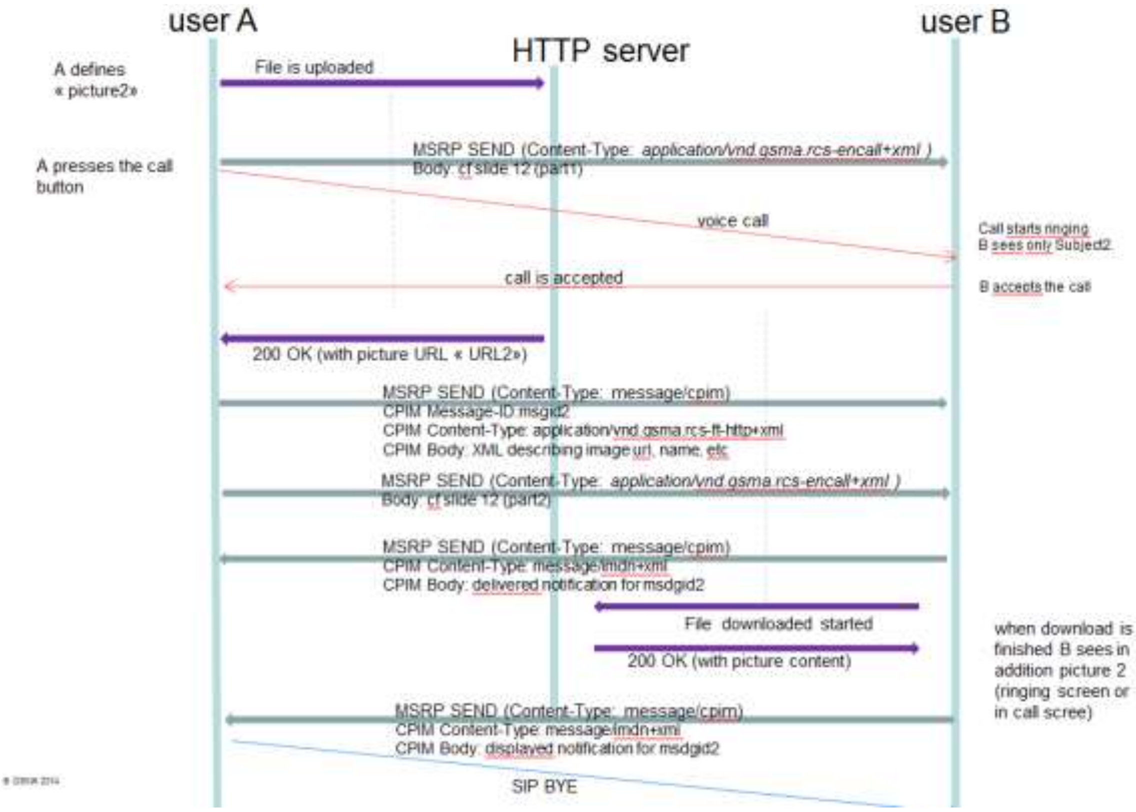
A.2 Example 2: Call Composer: Subject and picture with updates, B cannot download all information before call starts ringing.

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.

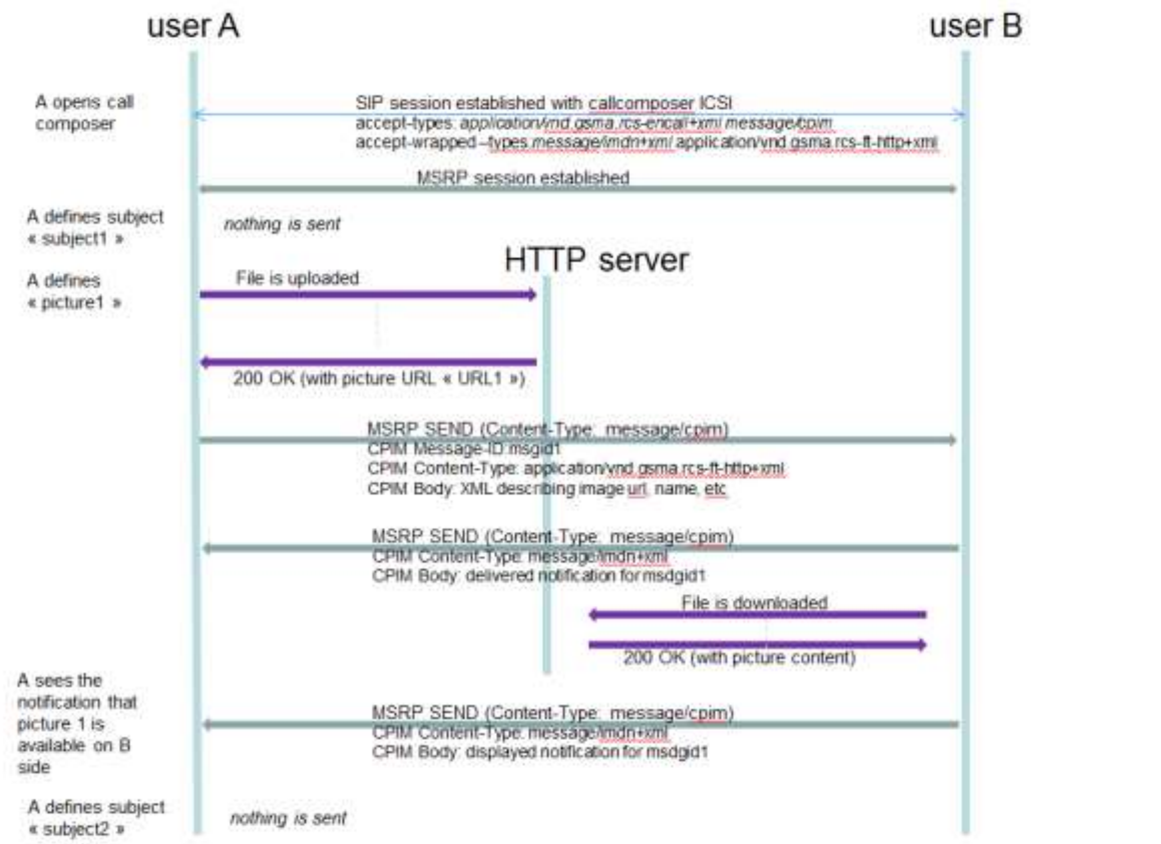
MSRP Send content

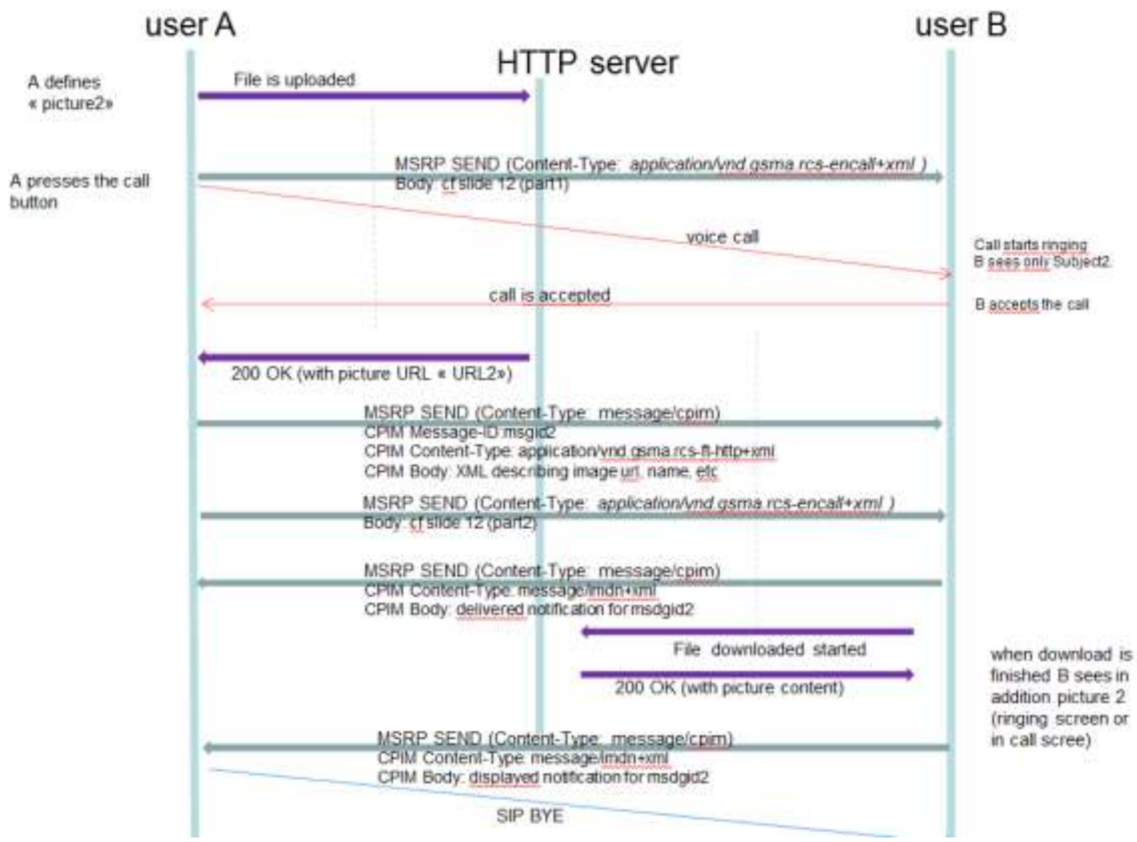
First MSRP SEND when A presses the call button


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

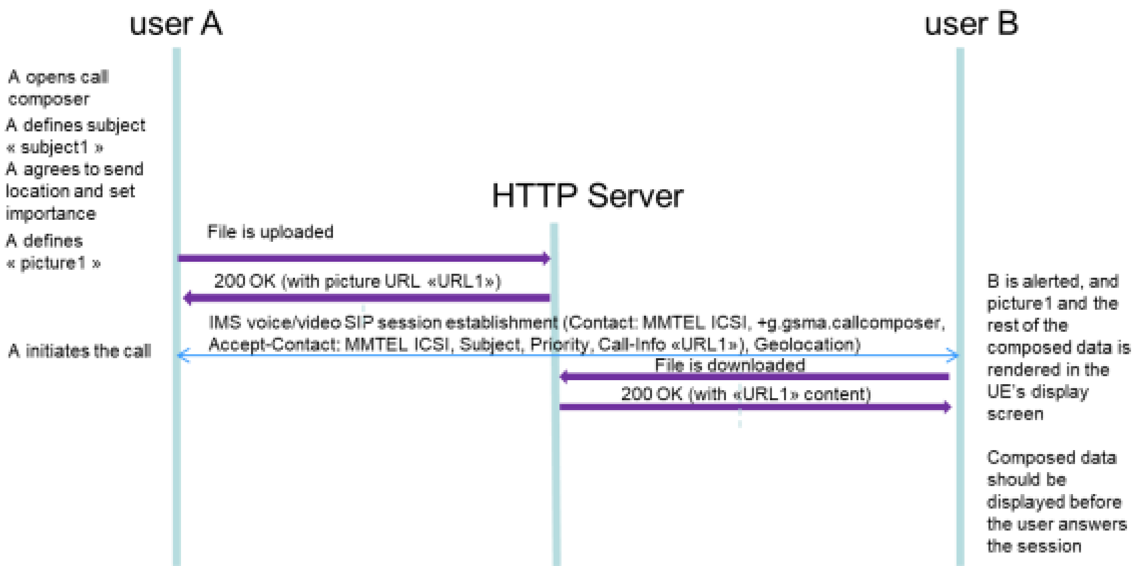

user A
user B
HTTP server
A defines « picture2»
File is uploaded
A presses the call button
MSRP SEND (Content-Type: application/vnd.gsma.rcs-encall+xml)
Body: cf slide 12 (part1)
voice call
Call starts ringing
B sees only Subject2
call is accepted
B accepts the call
200 OK (with picture URL « URL2»)
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: Call Composer: Multimedia Telephony Enriched Calling Composer service, B can download and display all information before call starts ringing.

User A performs the following actions (in this order):

- Sets the subject (Subject header)
- Sets the importance (Priority header)
- Sets the location (Geolocation header)
- Sets the picture «picture.jpg» (uploaded, then Call-Info header)
- User A presses the call button

Assumption: Capability exchange has indicated that User B is capable of receiving call composer elements in the Multimedia Telephony SIP INVITE request and User B downloads the picture before the call is received.

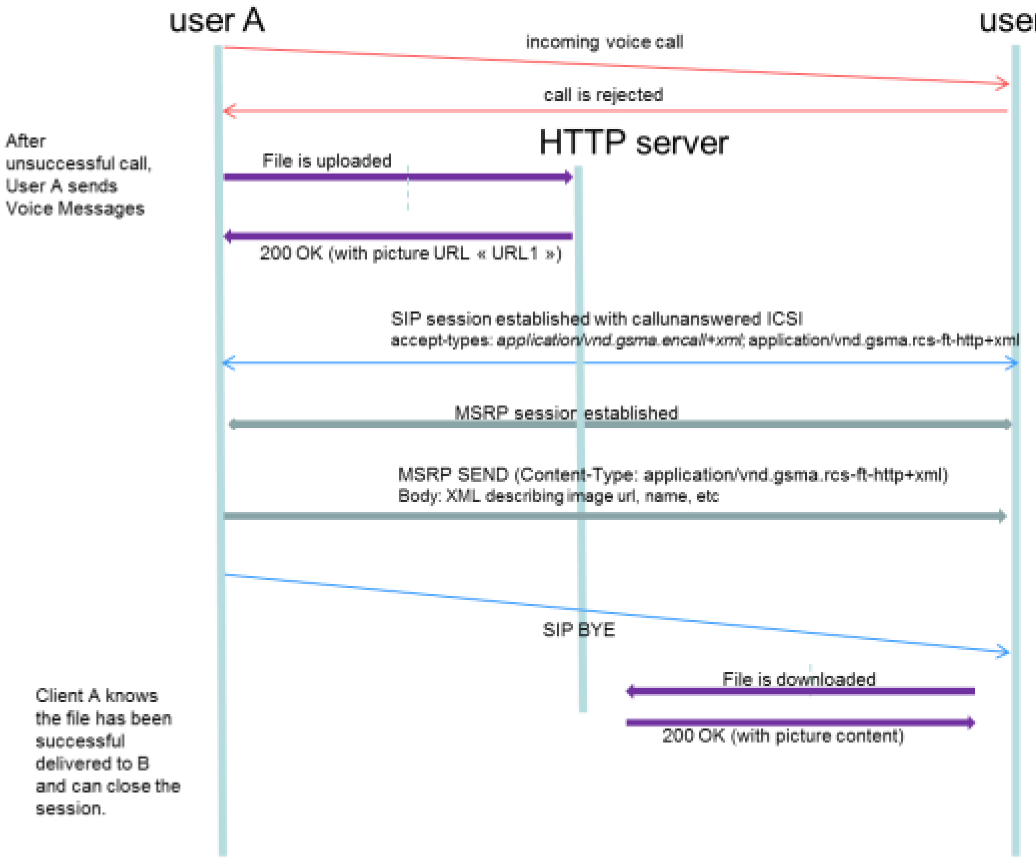


A.5 Example 5: Add reason after unanswered call.

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
3.0	28 June 2017	Include approved CR1003	TG	Tom Van Pelt / GSMA
4.0	16 May 2018	Include approved CR1004	TG	Tom Van Pelt / GSMA
5.0	06 December 2018	Include approved CR1005	TG	Tom Van Pelt / GSMA

B.2 Other Information

Type	Description
Document Owner	Network Group, 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

Your comments or suggestions & questions are always welcome.