Enriched Calling Technical Specification

Version 1.0

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Contents

1 Introduction

1.1 Overview

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

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

All these features are to be provided in conjunction with Telco voice reliable bearers (i.e. traditional circuit switched (CS) call and VoLTE).

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

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.
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.
IMDN	Instant Messaging Disposition Notification
MSRP	Message Session Relay Protocol
Native RCS Device	A device with a pre-installed RCS client (as opposed to a downloaded RCS client).
Offline	A contact who is Enriched Calling enabled but is NOT currently available.
Online	A contact who is Enriched Calling enabled and is currently available.
Plain Voice Call	A voice call with no enriched content.
SDP	Session Description Protocol
UX	User Experience
XML	Extensible Markup Language

1.3 Document Cross-References

Ref	Document Number	Title
1	[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
2	[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
3	[PRD-IR.90]	GSMA PRD IR.90 - "RCS Interworking Guidelines" Version 11.0 28 February 2015 http://www.gsma.com/
4	[PRD-IR.94]	GSMA PRD IR.94 - "IMS Profile for Conversational Video Service"

		Version 8.0 17 November 2014 http://www.gsma.com/
5	[PRD-RCC.07]	GSMA PRD RCC.07 - “Rich Communication Suite 5.3 Advanced Communications Services and Client Specification” Version 6.0 28 February 2015 http://www.gsma.com/
6	[PRD-RCC.60]	joyn Blackbird Product Definition Document Version 4.0 06 October 2014 http://www.gsma.com/
7	[RFC3261]	SIP (Session Initiation Protocol) IETF RFC http://tools.ietf.org/html/rfc3261
8	[RFC3264]	An Offer/Answer Model Session Description Protocol IETF RFC http://tools.ietf.org/html/rfc3264
9	[RFC3326]	The Reason Header Field for the Session Initiation Protocol (SIP) IETF RFC http://tools.ietf.org/html/rfc3326
10	[RFC4028]	The Session Timers in the Session Initiation Protocol (SIP) IETF RFC http://tools.ietf.org/html/rfc4028
11	[RFC4122]	The Universally Unique IDentifier (UUID) URN Namespace IETF RFC http://tools.ietf.org/html/rfc4122
12	[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
13	[RFC4975]	The Message Session Relay Protocol (MSRP) IETF RFC http://tools.ietf.org/html/rfc4975

2 Service realisation

2.1 Client configuration

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

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

2.1.1 Enriched Calling specific configuration request

The https configuration request for Enriched Calling clients shall include a dedicated parameter to identify the client as an Enriched Calling capable client.

An Enriched Calling capable client shall indicate its capability via an additional rcs_profile parameter included in the HTTP GET requests with a value set to "enriched_call". This rcs_profile parameter is set in addition to other parameters indicating the support of RCS Services, e.g. joyn_blackbird.
Example of HTTP GET query:

https://config.rcs.mnc002.mcc262.pub.3gppnetwork.org?vers=0&rcs_version=5.1B&**rcs_profile=joyn_blackbird**&**rcs_profile=enriched_call**&client_vendor=OEMxyz&client_version=RCSAndr-2.0&terminal_vendor= OEMxyz&......

2.1.2 Additional configuration parameters

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

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

Table 1: Enriched Calling Related Configuration Parameters

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

2.1.2.1 RCS Management Tree Additions

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

<u>Node: /<x>/Services/composerAuth</u>
Leaf node that represents the authorization for the user to use the Enriched Calling Call Composer service
It shall be instantiated if the network supports any Enriched Calling Service.

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

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

- Values: 0, 1

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

  1- Indicates that the Enriched Calling Call Composer service is enabled
  The SIP REGISTER shall include the ICSI pertaining to Enriched Calling Call Composer if used by installed applications in its Contact header. Also, they are included as part of capability discovery. Enriched Calling Call Composer entry points will be displayed based on the capability discovery.

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

Node: /<x>/Services/sharedMapAuth

Leaf node that represents the authorisation for the user to use the Enriched Calling Shared Map service during a call.
It shall be instantiated if the network supports any Enriched Calling Service.

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

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

- Values: 0, 1

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

  1- Indicates that the Enriched Calling Shared Map service is enabled
  The SIP REGISTER shall include the ICSI pertaining to the Enriched Calling Shared Map if used by installed applications in its Contact header. Also, they are

included as part of capability discovery. Shared Map entry points will be displayed during the call based on the capability discovery.

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

Node: /<x>/Services/sharedSketchAuth

Leaf node that represents the authorisation for the user to use the Enriched Calling Shared Sketch service during a call.
It shall be instantiated if the network supports any Enriched Calling Service.

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

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

- Values: 0, 1

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

  1- Indicates that the Enriched Calling Shared Sketch service is enabled
  The SIP REGISTER shall include the ICSI pertaining to the Enriched Calling Shared Sketch if used by installed applications in its Contact header. Also, they are included as part of capability discovery. Shared Sketch entry points will be displayed during the call based on the capability discovery.

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

Node: /<x>/Services/postcallAuth

Leaf node that represents the authorisation for the user to use the Enriched Calling Post Call service.
It shall be instantiated if the network supports any Enriched Calling Service.

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

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

- Values: 0, 1

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

  1- Indicates that Post Call service is enabled
  The SIP REGISTER shall include the ICSI pertaining to Enriched Calling Post Call if used by installed applications, in its Contact header. Also, they are included as part of capability discovery. Enriched Calling Post Call entry points will be displayed based on the capability discovery.

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

Node: /<x>/Other/callComposerTimerIdle

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

Status	Occurrence	Format	Min. Access Types
Required	ZeroOrOne	Int	Get

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

- Values: < Timer value in seconds>
  When set to 0, there shall be no timeout.
  The default value is 180 (seconds).
- Optional parameter
  (It is mandatory and becomes relevant only if composerAuth is set to 1)
- Post-reconfiguration actions: There are no additional actions apart from application of the new value from the time of re-configuration onwards.
- Associated HTTP XML parameter ID: “callComposerTimerIdle”

As indicated, all parameters apart from the callComposerTimerIdle parameter shall be included in the Services MO defined in [PRD-RCC.07]. This leads to the following tree structure for the Services MO if Enriched Calling is supported:

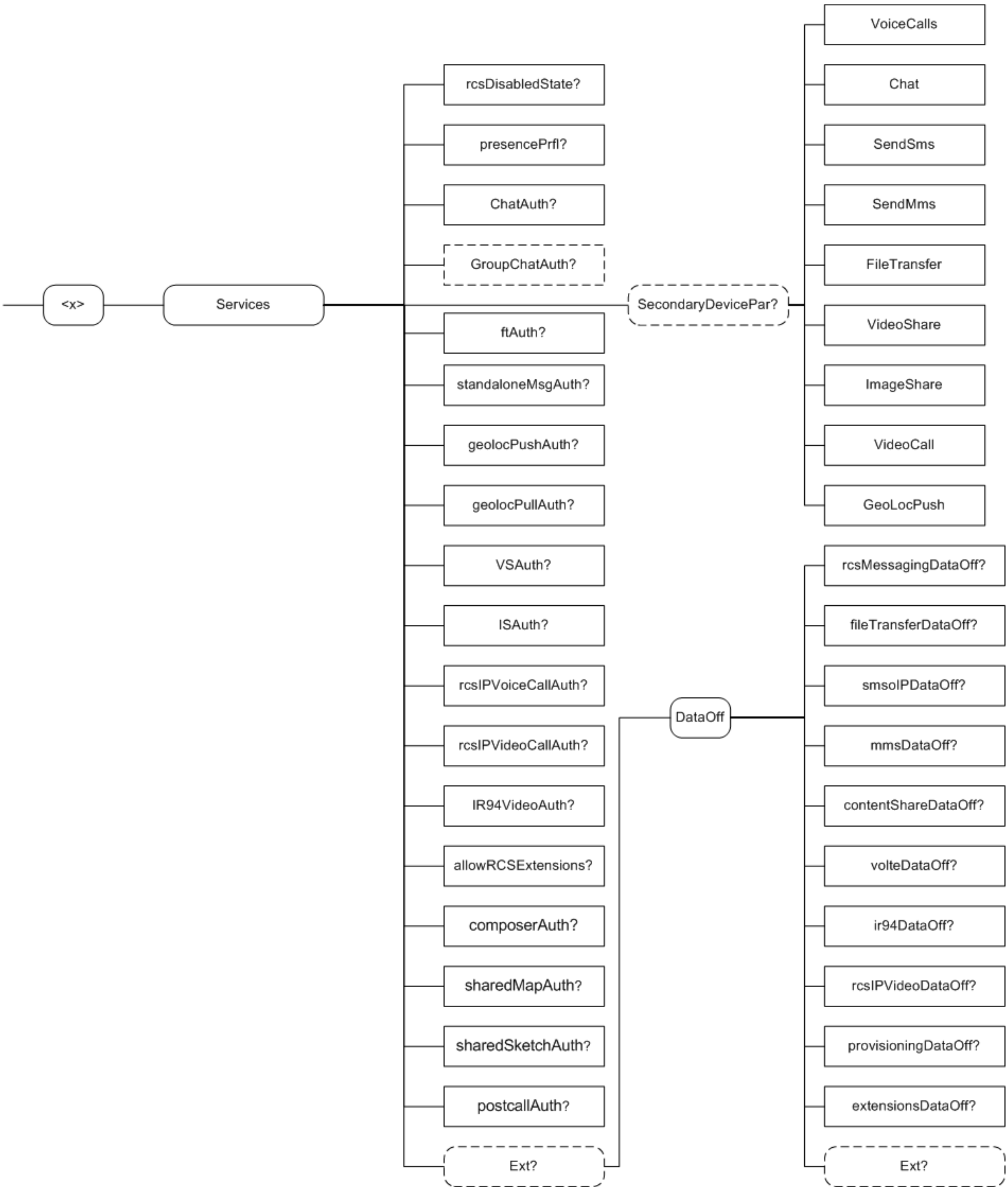


Figure 1: Services sub tree

The associated HTTP configuration XML structure is presented in the table below:

```
<characteristic type="SERVICES">
        <parm name="presencePrfl" value="X"/>
        <parm name="ChatAuth" value="X"/>
        <parm name="GroupChatAuth" value="X"/>
        <parm name="ftAuth" value="X"/>
        <parm name="standaloneMsgAuth" value="X"/>
        <parm name="geolocPullAuth" value="X"/>
        <parm name="geolocPushAuth" value="X"/>
        <parm name="vsAuth" value="X"/>
        <parm name="isAuth" value="X"/>
        <parm name="rcsIPVoiceCallAuth" value="X"/>
        <parm name="rcsIPVideoCallAuth" value="X"/>
        <parm.name="IR94VideoAuth" value="X"/>
        <parm.name="allowRCSExtensions" value="X"/>
        <parm.name="composerAuth" value="X"/>
        <parm.name="sharedMapAuth" value="X"/>
        <parm.name="sharedSketchAuth" value="X"/>
        <parm.name="postCallAuth" value="X"/>
        <characteristic type="Ext">
                <characteristic type="DataOff">
                        <parm name="rcsMessagingDataOff" value="X"/>
                        <parm name="fileTransferDataOff" value="X"/>
                        <parm name="smsoIPDataOff" value="X"/>
                        <parm name="mmsDataOff" value="X"/>
                        <parm name="contentShareDataOff" value="X"/>
                        <parm name="volteDataOff" value="X"/>
                        <parm name="ir94DataOff" value="X"/>
                        <parm name="rcsIPVideoDataOff" value="X"/>
                        <parm name="provisioningDataOff" value="X"/>
                        <parm name="extensionsDataOff" value="X"/>
                        <characteristic type="Ext"/>
                </characteristic>
        </characteristic>
        <characteristic type="SecondaryDevicePar">
                <parm name="VoiceCall" value="X"/>
                <parm name="Chat" value="X"/>
                <parm name="SendSms" value="X"/>
                <parm name="FileTransfer" value="X"/>
                <parm name="VideoShare" value="X"/>
                <parm name="ImageShare" value="X"/>
                <parm name="VideoCall" value="X"/>
                <parm name="GeoLocPush" value="X"/>
        </characteristic>
</characteristic>
```

Table 7: Services MO sub tree associated HTTP configuration XML structure

When Enriched Calling is supported, the Data Off behaviour for all Enriched Calling Services including pre- and postcall shall be controlled through the contentShareDataOff parameter.

The callComposerTimerIdle parameter is added to the Other MO tree defined in [PRD-RCC.07], leading to the following tree structure:

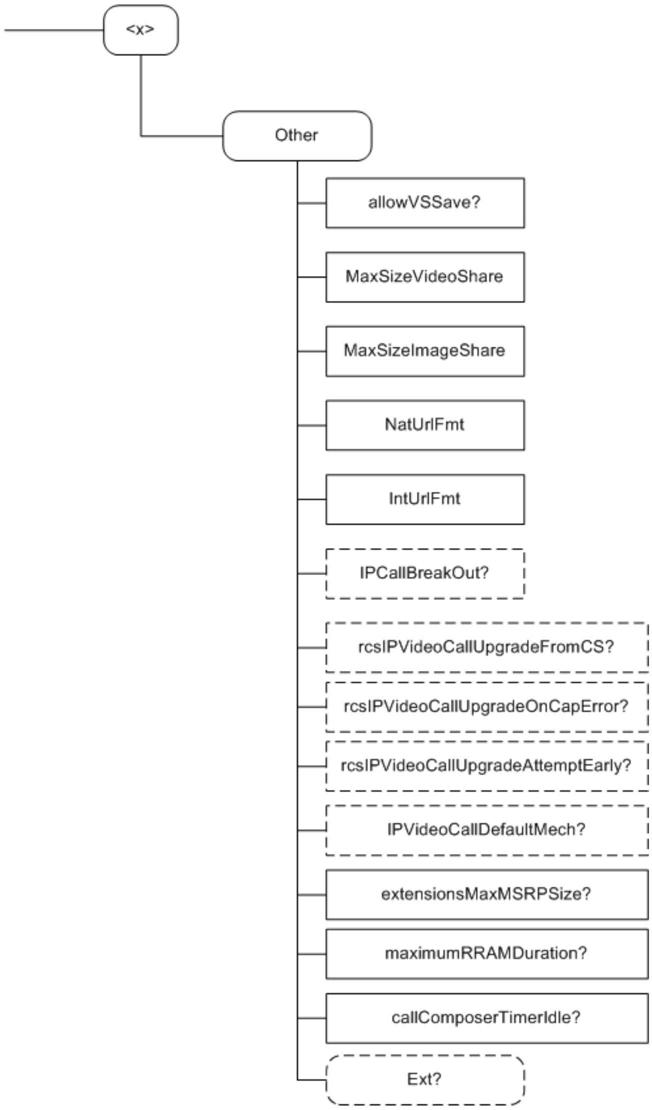


Figure 2: Other MO sub tree

The associated HTTP configuration XML structure is presented in the table below:

```
<characteristic type="OTHER">
        <parm name="allowVSSave" value="X"/>
        <parm name="MaxSizeImageShare" value="X"/>
        <parm name="MaxTimeVideoShare value="X"/>
        <parm name="NatUrlFmt" value="X"/>
        <parm name="IntUrlFmt" value="X"/>
        <parm name="IPCallBreakOut" value="X"/>
        <parm name="rcsIPVideoCallUpgradeFromCS" value="X"/>
        <parm name="rcsIPVideoCallUpgradeOnCapError" value="X"/>
        <parm name="rcsIPVideoCallUpgradeAttemptEarly" value="X"/>
        <parm name="IPVideoCallDefaultMech" value="X"/>
        <parm name="extensionsMaxMSRPSize" value="X"/>
        <parm name="maximumRRAMDuration" value="X"/>
        <parm name="callComposerTimerIdle" value="X"/>
        <characteristic type="Ext"/>
</characteristic>
```

Table 8: Other sub tree associated HTTP configuration XML structure

2.2 Capability discovery

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

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

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

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

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

- Video share (as defined in RCS)
- Share any file (file transfer via http)
- Share text during a call (1 to 1 chat)
- Location share (Geolocation PUSH during a call)

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

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

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

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

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

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

For Enriched calling services ICSI tags and service tuples definitions see sections 2.4.1, 2.5.1, 2.9.7 and 2.9.8 **Error! Reference source not found.**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:

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

2.3.1 Initiating an RCS Enriched Calling Session

Handling at Initiating Nodes

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

- Shall include the address of the target RCS contact in the Request-URI;
- Shall include an Accept-Contact header field with the ICSI for the Enriched Calling service this session is related to (for Enriched Calling ICSI tag definition, see sections 2.3, 2.4, 2.5, 2.9.7 and 2.9.8 of this document) percent encoded as per [3GPP TS 24.229] section 7.2A.8.2 "*Coding of the ICSI*" in a g.3gpp.icsi-ref media feature tag with the *require* and *explicit* parameters;
- Shall set the P-Preferred-Service header field with the value of the Enriched Calling ICSI tags (for Enriched Calling ICSI tag definition, see sections 2.3, 2.4, 2.5, 2.9.7 and 2.9.8 of this document);
- Shall include a Contact header field with the ICSI for the Enriched Calling specific service this session is related to percent encoded as per [3GPP TS24.229] section 7.2A.8.2 "*Coding of the ICSI*" in a *g.3gpp.icsi-ref* media feature tag;
- Shall include the address of the originating RCS Client that has been authenticated as per section 2.5.3.2 of [PRD-RCC.07]and [3GPP TS 24.229];
- Shall include a User-Agent header field as specified in [RFC3261];
- Should include a Session-Expires header field with the refresher parameter set to "uac" according to the rules and procedures of [RFC4028];
- Shall include a MIME SDP body as an SDP offer as described in section 2.3.4 of this document.
- Shall send the SIP INVITE request according to the rules and procedures of [3GPP TS 24.229]

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

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

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

1. The RCS Client may indicate to the user that the session could not be established;

2. The RCS Client shall generate and send a SIP ACK request as an acknowledgement of the final response according to the rules and procedures of [RFC3261].

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

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

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

2. If Enriched Calling services are running on the device, the RCS Client:

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

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

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

2.3.2 Terminating an RCS Enriched Calling Session

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

1. Shall generate a SIP BYE request according to the rules and procedures of [3GPP TS 24.229], with the Reason Header field as defined in [RFC3326] with the protocol-value set to SIP, the protocol-cause set to *200* (e.g., *SIP;cause=200*);

2. Shall send the SIP BYE request according to the rules and procedures of [3GPP TS24.229];
3. Shall release all Media Plane resources corresponding to the Enriched Calling Session being closed.

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

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

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

When receiving a SIP BYE request, the client shall

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

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

2.3.3 SDP Contents

SDP Contents when Initiating a Session

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

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

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

- The *a=accept-types* shall only include:
  - In case of Call Composer *application/vnd.gsma.encall+xml* and *message/cpim* MIME types

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

In case of Shared Sketch and Shared Map refer to section 2.9.9.2

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

In case of Shared Sketch and Shared Map refer to section 2.9.9.2

2.3.4 MSRP Session Handling

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

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

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

2.3.5 NNI and IOT considerations

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

2.4 Call composer

2.4.1 Call composer procedures

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

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

Table 9: Enriched Calling Composer service realisation summary

2.4.2 Picture transfer procedures

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

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

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

The format of the picture should be compliant with section 4.2 and 4.3 of [3GPP TS 26.141]. In case the picture file size exceeds a recommended value of 80kB or in case its resolution would not give an advantage for displaying on a mobile screen, the picture should be resized before it is uploaded.

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

2.4.3 Session management

If the capability discovery is successful (B-Party supports Call Composer service) A-Party shall initiate an Enriched Calling session using procedures described in section 2.3 and ICSI defined in section 2.4.4 as soon as entering the call composer screen. B-Party shall auto accept the session in order to establish the MSRP session.

The initiator shall re-establish the session when additional data related to the call composer session needs to be sent or when resending data is necessary.

The A-Party shall only close the Enriched Calling session:

- After the pre-call data has been received by the B-Party (200 OK MSRP received by A)
- AND after the call state left the 'ringing' state (as A-Party can still set the urgency flag during this ringing phase)
- AND if there is a pre-call picture set, after the A-Party has received the displayed notification for the latest picture sent to the B-Party.

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

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

The Enriched Calling session is also closed when the timeout on call composer session occurs.

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

Exception case:
If the A-Party has set a call composer picture, but presses the call button later than 60 minutes, the A-Party has to send the picture again according to the procedures described in step 1 of section 2.4.4 of this document.

2.4.4 Call Composer elements

Service media specification

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

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

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

Table 10:XML message based on HTTP content server response (see the File Transfer procedure in section 3.5.4.8.3 of [PRD-RCC.07])

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

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

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

Table 11: Call composer service; Complete XML sample

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

The elements under rcscalldata are described below:

- <subject> : The maximum length of the subject field should be 60 characters.
- <importance> : The values are “1” for an important call; “0” for a standard call.
- <location>: The location shall be a geo URI. Example: geo:37.786971,-122.399677
- <picture url>: The URL of the picture file on the FT content server to be displayed.
- <composerid>Random generated number to identify the session. The maximum length should be 10 characters. The composerid is a mandatory field.

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

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

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

Table 12: complete XML example after the call button has been pressed

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

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

2.4.5 Receiver procedures

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

In case a picture has been sent as part of Call Composer, the receiver side will download the call composer picture file automatically as soon as it has received the link in the call composer picture XML. For download of the call composer picture the procedures in sections 3.5.4.8.3.2 and 3.5.4.8.6.4 of [PRD-RCC.07] apply.
The call composer information shall be automatically accepted by the receiving side. This also includes the picture, which is limited in size.

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

The client displays the call composer information when the Call Composer XML is received AND the corresponding CS or VoLTE call is incoming.

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

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

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

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

2.4.6 Delivery and display notifications

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

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

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

For more details see also Annex B

2.4.7 Expected Call forwarding behaviour

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

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

2.4.8 Legacy support for call composer

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

2.5 Post-call Service

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

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

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

2.5.1 Post-call procedures

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

<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.callunanswered"</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.callunanswered</td></tr>
<tr><td colspan="2">Service Tuple</td><td>org.3gpp.urn:urn-7:3gpp-service.ims.icsi.gsma.callunanswered
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 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 3.11.4.2.2 of [PRD-RCC.07].

2.5.3 Session Management

If 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 3.11.4.2.2 of [PRD-RCC.07]

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

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

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

The audio message shall not exceed the duration as defined in MAX RRAM DURATION (as defined in Annex A of [PRD-RCC.07]) 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.

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

2.5.6 Call forwarding behaviour

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

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

2.5.7 Legacy and offline support for Post-call Services

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

2.6 XML Schema

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

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

<!--
  Enriched Calling document
    version - 2.1
    date    - 19 May 2015

FILE INFORMATION

        GSMA OIG Enriched Calling 1.0: Technical Specification Document
    File: OIG-SUP-XSD_Enriched_Calling-V2_1-20150519.xsd
    Type: Text - Schema Description
-->

<xs:schema
    targetNamespace="urn:gsma:params:xml:ns:rcs:rcs:calldata"
    xmlns="urn:gsma:params:xml:ns:rcs:rcs:calldata"
    xmlns:xs="http://www.w3.org/2001/XMLSchema"
    elementFormDefault="qualified" attributeFormDefault="unqualified">

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

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

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

Table 16: Complete XML Schema supporting for the EnCall Call Composer and Post call service

2.7 Enriched call logs

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

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

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

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

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

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

As the content shall also be made accessible from the call logs, the selection of the content is left to implementation choice. For instance:
The devices apply a filter using the call start and ending date to filter the messages attached to that specific call: any message with time stamp is included in the call time frame shall be selected for the call entries.

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

2.8 Video call

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

2.9 In call sharing

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

The following in call sharing features are possible:

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

2.9.1 Video sharing:

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

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

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

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

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

Table 17:Standard RCS Release 1-4 SIP OPTIONS tag

The network parameter to allow video share is:

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

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

2.9.2 Share any file during a call

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

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

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

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

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

2.9.3 Exchanging messages during a call:

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

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

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

2.9.4 Location push during a call:

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

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

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

Table 18: 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 19: 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)+>
```

```
  <!ELEMENT close EMPTY>
  <!ELEMENT version EMPTY>
  <!ELEMENT drawing (points)>
  <!ELEMENT remove (id)>
  <!ELEMENT undo EMPTY>
  <!ELEMENT bounds (points)>
  <!ELEMENT marker (title?, snippet?, point, id)>
  <!ELEMENT user (point)>
  <!ELEMENT points (#PCDATA)>
  <!ELEMENT point (#PCDATA)>
  <!ELEMENT title (#PCDATA)>
  <!ELEMENT snippet (#PCDATA)>
  <!ELEMENT id (#PCDATA)>
  <!ATTLIST version  id CDATA #REQUIRED>
  <!ATTLIST drawing color CDATA #REQUIRED>
  <!ATTLIST drawing width CDATA #REQUIRED>
  <!ATTLIST drawing erase CDATA “true”>
  <!ATTLIST points 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 20: 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)+>
   <!ELEMENT version EMPTY>
   <!ELEMENT close EMPTY>
   <!ELEMENT drawing (points)>
   <!ELEMENT undo EMPTY>
   <!ELEMENT image (#CDATA)>
   <!ELEMENT points (#PCDATA)>
   <!ELEMENT background_color EMPTY>
   <!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 “true”>
   <!ATTLIST points encoding CDATA “Base64”>
   <!ATTLIST image encoding CDATA “Base64”>
]>
```

2.9.9 Shared Map / Shared Sketch session management

A user can only engage in one Shared Sketch/Map with a specific contact at a given time. E.g. it is not possible to share two sketches/maps with the same contact at the same time. To start a new Shared Sketch/Map the current one needs to be closed first.

The session is considered active until one party closes the session. Either by closing the MSRP session or by sending <close>.

If an error occur on the MSRP session, or if the MSRP session is unexpectedly closed, the Shared Sketch/Map session remains active and the client should try to reconnect the MSRP session as follows:.

- The client should try to reconnect the session for a reasonable amount of time.
- If the other party receives an INVITE from the same user that he is already in a session with, he should consider the new INVITE as a reconnect attempt and accept this, and send data on the new session.
- To avoid race conditions it is recommended to listen for data on all open sessions, a possible scenario is that both clients tries to reconnect at the same time and sends an INVITE to the other party. This could lead to two open sessions, which should be considered OK and data transferred on both sessions must be handled.

The client should close the session when the call ends.

2.9.9.1 Session established

When the session has been established both users are expected to send the <version> element as soon as possible.

For Shared Map both users should also send the <user> element as soon possible.

2.9.9.2 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 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 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:

None

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 should be sent directly after the session has been established by both parties. The lowest version should be used by both parties, and the UI updated to remove unsupported features.

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

If this tag is omitted then the other party should assume version 1.

Attributes:

id – The protocol version suggested to be used by the sender. Should be an integer value.

Elements/PCDATA:

None

2.9.10.1.3 <close>

This tag is used to tell the other party that the session should end. Any actions received after this should be discarded.

Attributes:

None

Elements/PCDATA:

None

2.9.10.1.4 <drawing>

This tag describes a drawing.

Attributes

Width: describes the width of the drawing (in dp) with float precision.

Colour: describes the colour of the drawing in ARGB.

Erase: a Boolean value that decides if this is an erase or a drawing. If attribute is not present it is interpreted as being a drawing. Shared Sketch only

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 should be discarded.

Attributes:

None

Elements/PCDATA:

None

2.9.10.1.6 <points>

Tag containing coordinates. The amount of points allowed differs depending on the outer tag. It must however be a multiple of 2. 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.

Attributes:

Encoding – The encoding used. Defaults to Base64. 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. This is only used Shared Map since the canvas encompasses such a large area.

Attributes:

Encoding: defaults to the string Base64. Currently this is the only encoding supported

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,
- points (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 by shared map

Attributes:

None

Elements/PCDATA:

Points (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 <snippet>

Contains the subtitle of a marker.

Attributes:

None

Elements/PCDATA:

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

2.9.10.2.7 <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_colour>

This elements specifies the background colour of the canvas.

Attributes

Colour – The new canvas background colour in ARGB

Elements/PCDATA:

None

2.9.10.3.2 <image>

Tag containing an image that will be set as background. Recommended size < 500 kB.

Attributes:

Encoding – The encoding used. Defaults to Base64. Currently the only supported format

Elements/PCDATA:

Base64 encoded image.

2.9.10.3.3 <point>

Tag containing a coordinate, latitude and longitude.

Attributes:

Encoding – The encoding used. Defaults to Base64. 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.

2.9.11 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="-9580554" width="7.0">

<points encoding="Base64">

QEvdCvCXCWZAKmRCbAAAAEBL3QrwlwlmQCpkRESAAABAS90KrhG8cEAqZEU8AAAA
QEvdCos5n/RAKmRFPAAAAEBL3QpItFGmQCpkRihAAABAS90KBJmMvUAqZEYoQAAA

</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 Video share options

The share video service is provided via two different means:

- ViLTE (IR.94) service (when available from the network) as per [PRD-IR.94] provide:
    - The user initiating the share is using a ViLTE capable phone,
    - The call leg of the user initiating the share is provided via VoLTE and,
    - The capability exchange prior to the call shows that the ViLTE (IR.94) service is supported by the other end
- RCS video share service as defined in [PRD-RCC.07].

2.9.12.1 Service realisation information

RCS Service		Tag
RCS/ViLTE service tag		<u>ViLTE:</u> +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel";video <u>Video share:</u> +g.3gpp.cs-voice
Service realisation type		<u>ViLTE (IR.94)</u> Standard GSMA PRD IR.94 service <u>Video share:</u> Standard RCS service
Extension service	Media type	N/A
	Auto accept	N/A
	Store & Forward	N/A
	Max size	N/A

Table 21: Upgrade to video service realisation summary

2.9.12.2 Service logic

In addition to the procedures described in the relevant specifications ([PRD-RCC.07], [PRD-IR.90], [PRD-RCC.60] and [PRD-IR.94]) and as a consequence that for Enriched Calling, a dual IMS stack is employed, the following procedures shall be implemented by the client:

- If feasible, the client shall be aware on whether
    - The handset is currently on a VoLTE call (the handset is on a call and the data coverage is 4G), and,
    - The handset is ViLTE capable.
- If the preconditions described above are both true, the client needs to make sure that the ViLTE capability tag is included in any capability exchange taking place during the call.

    - There is an exception; if the other party does not support capability exchange (VoLTE only device), the ViLTE call button shall be offered provided the preconditions in the first bullet can be fulfilled.
    - This means that there is a possible failure case where the video call cannot be established (down to UX to handle the error).

- If the capability exchange returns that ViLTE and video share are possible end-2-end, then ViLTE shall prevail.
- Provided video share is available and the ViLTE establishment fails, video share shall be employed instead seamlessly to the user.

In case a ViLTE call is supported by the network:

Node: /<x>/ SERVICEPROVIDEREXT /videoUpgradeMethod

Leaf node that describes whether Video Share and IP Video Call should share a common button to upgrade a call in the absence of end-2-end ViLTE availability.

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

Table 22:SERVICEPROVIDEREXT MO sub tree addition parameters (videoUpgradeMethod)

- Values:
  0, In case of conflict, the video share service prevails.
  1, In case of conflict, the IP video call service prevails (IR 94).

- Post-reconfiguration actions: As the client remains unregistered during configuration, there are no additional actions apart from de-registering using the old configuration and registering back using the new parameter.
- Associated HTTP XML characteristic type: “videoUpgradeMethod”

The associated HTTP configuration XML structure is presented in the table below:

```
<characteristic type=” SERVICEPROVIDEREXT ”>
   …
   <characteristic type=”EnCall”>
      <parm name=”videoUpgradeMethod” value=”X”/>
   </characteristic>
</characteristic>
```

Table 23: SERVICEPROVIDEREXT sub tree associated HTTP configuration XML structure

Annex A Summary of required network provisioning parameters

To allow the use of Enriched Calling services, the following network parameters are required as part of the configuration XML. Those parameters have also been mentioned in the corresponding chapters in this document.

```
<parm name=”vsAuth” value=”1” />
```

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

Enriched Calling Specific parameters:
<parm name=”composerAuth” value=”1”/>
<parm name=”sharedMapAuth” value=”1”/>
<parm name=”sharedSketchAuth” value=”1”/>
<parm name=”postCallAuth” value=”1”/>
<parm name=”callComposerTimerIdle” value=”X”

In case ViLTE is supported by the network:
<parm name=”videoUpgradeMethod” value=”X”/>

Annex B Call flow examples

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 >

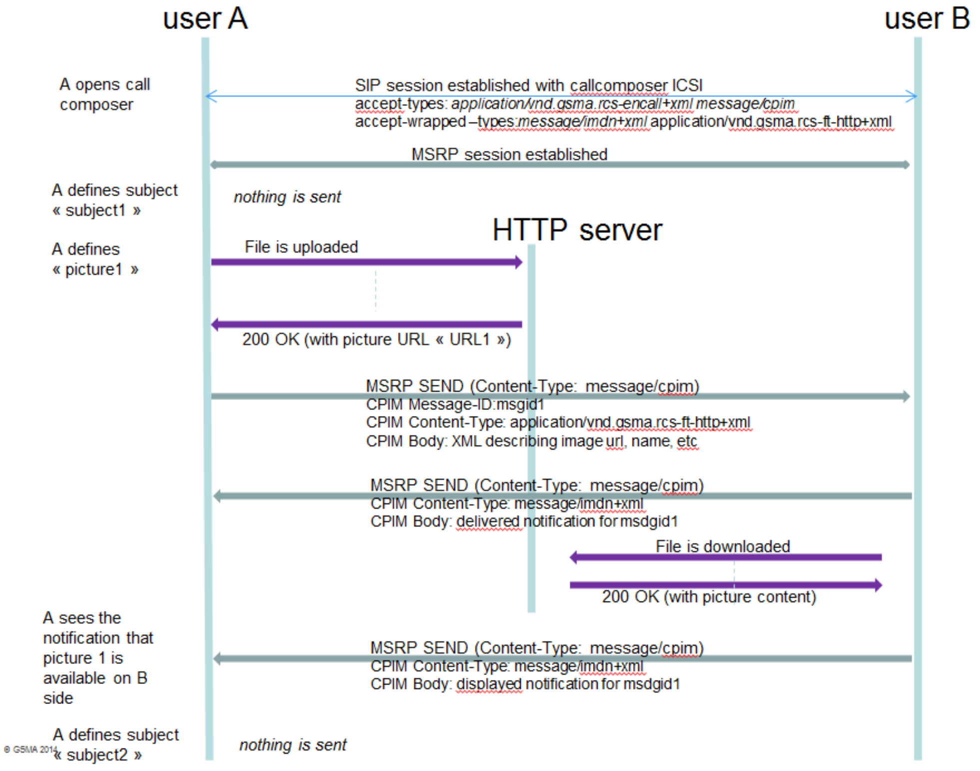
<subject>subject2</subject>

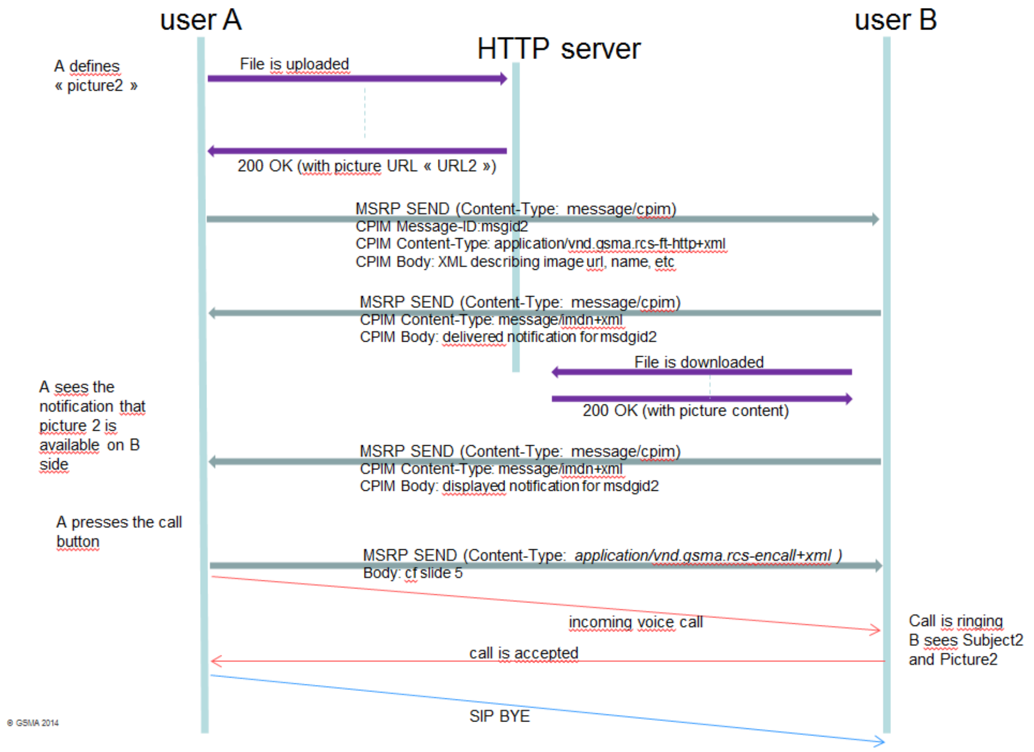
<composerid>12345</composerid>

<pictureurl>URL2</pictureurl>

</rcscalldata>

</rcsenvelope>



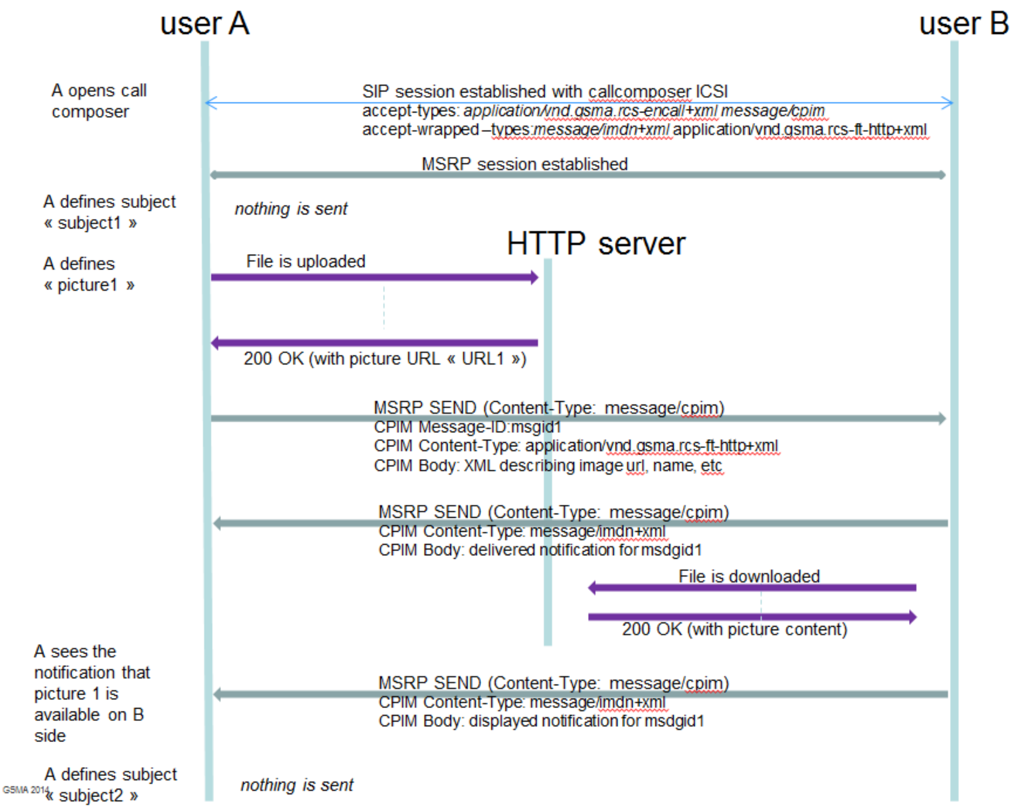


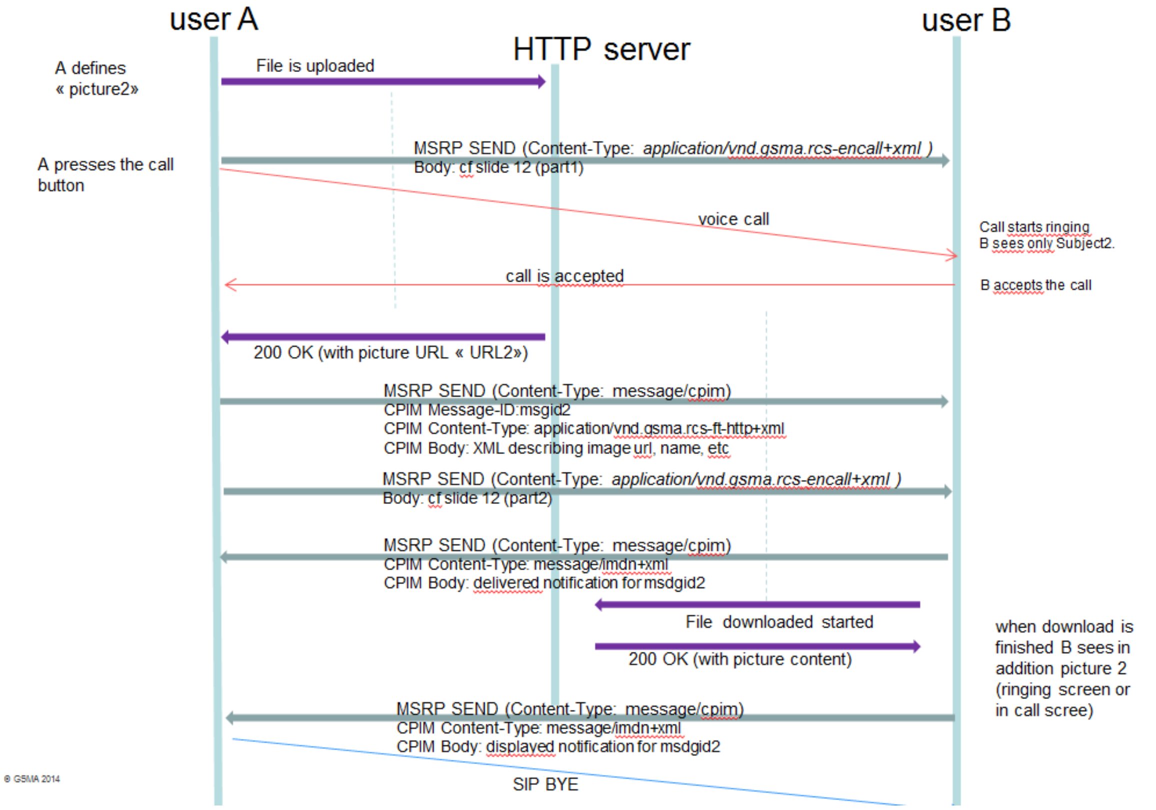
Example 2: Call Composer: Subject and picture with updates, B cannot download all information before call starts ringing.

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





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

User A does the following action (in this order)

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

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

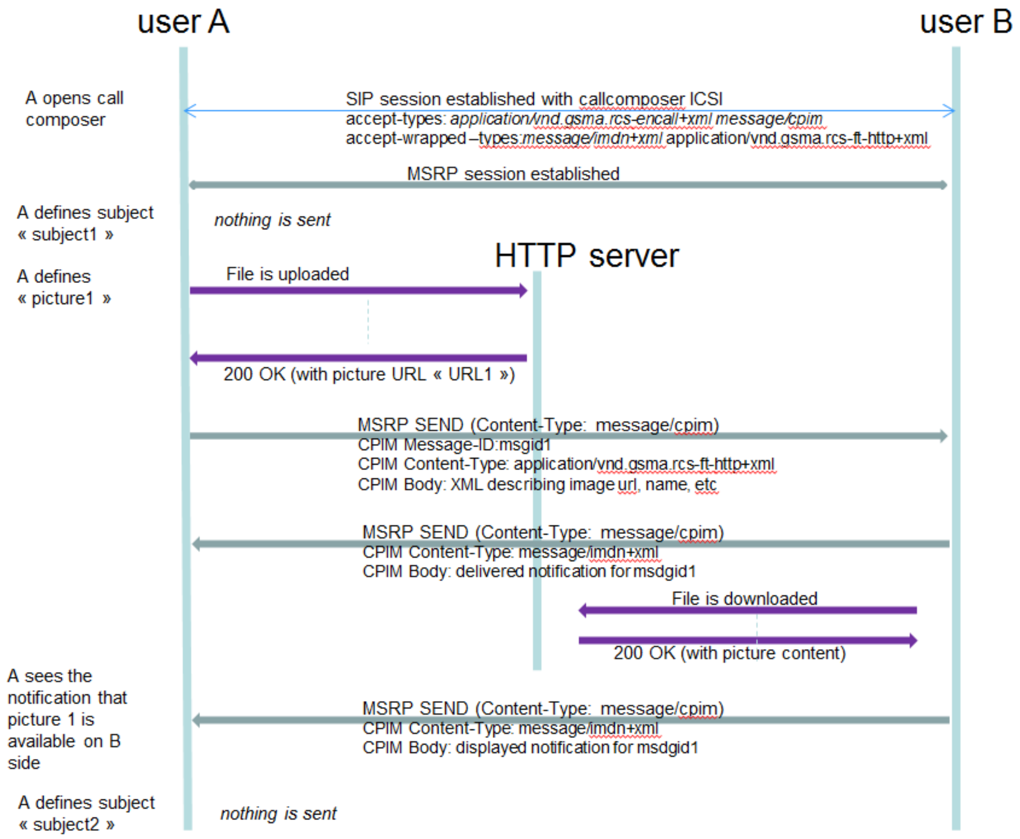
MSRP Send content

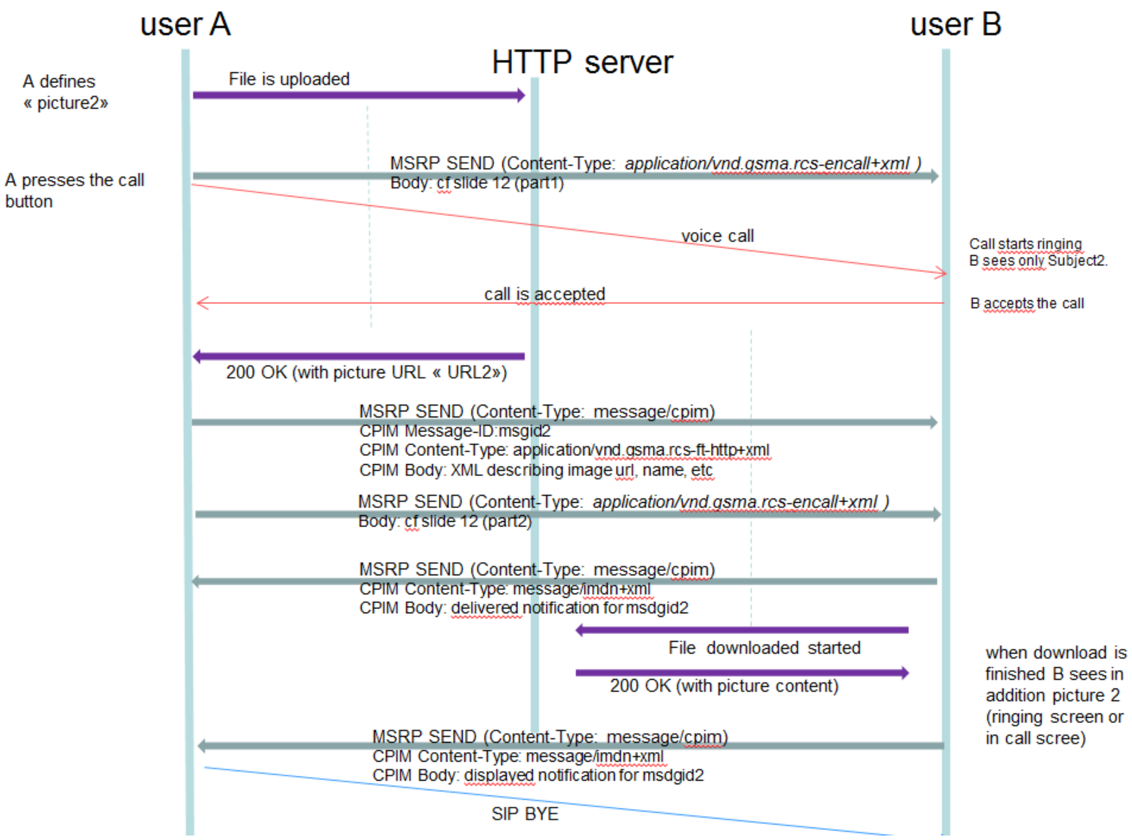
First MSRP SEND when A presses the call button

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

Second MSRP SEND when A has finished to upload picture 2

< ?xml version=on=n= encoding=ncoding=o
<rcsenvelope xmlns=murn :gsma :params :xml :ns :rcs :rcs :calldata”>
<rcscalldata >
<subject>subject2</subject>
<pictureurl>URL2</file-info>
<composerid>12345</composerid>
</rcscalldata>
</rcsenvelope>



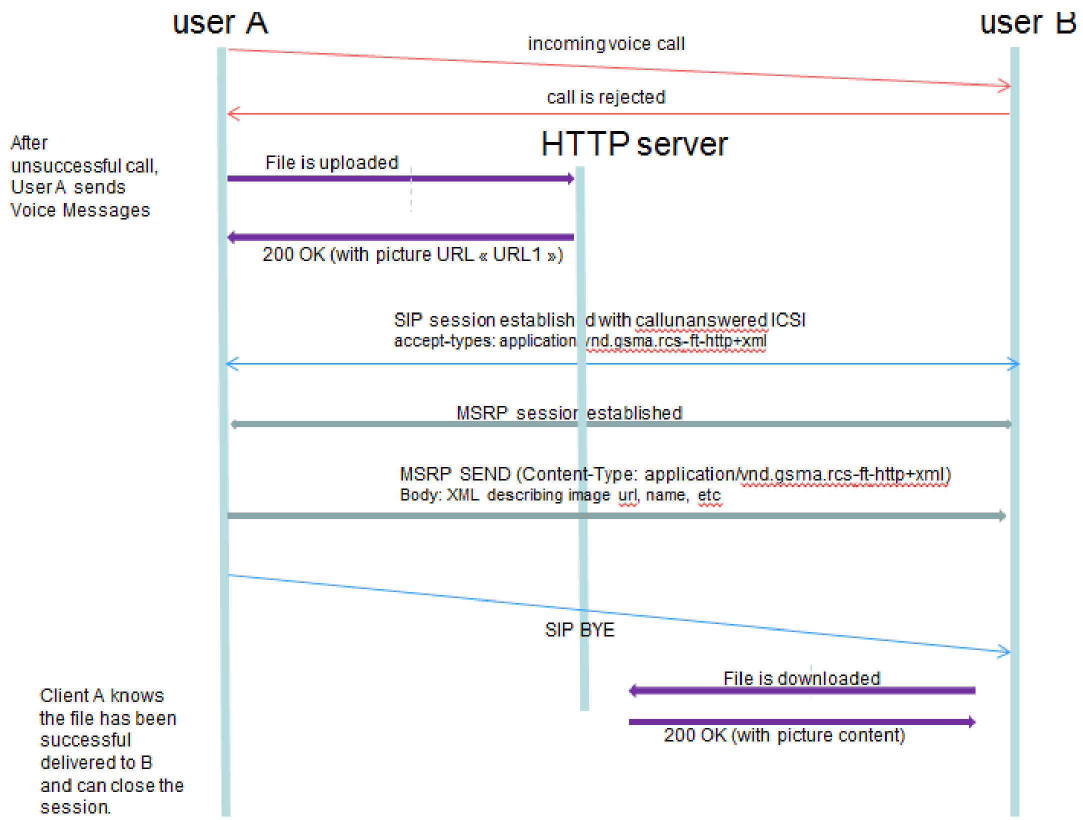


Example 4: Add reason after unanswered call.

User A does the following action

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

User B is ON-LINE



Annex C Content of the current backlog

https://infocentre2.gsma.com/gp/pr/V2020/N2020/IPCOIG/EnCallOIG/WorkingDocuments/OIG_EnrCalling_Feature%20Backlog.xlsx

Annex D Document Management

D.1 Document History

Version	Date	Brief Description of Change	Approval Authority	Editor / Company
1.0	13/07/2015	New document	PSMC	Andreas Brock / Vodafone

D.2 Other Information

Type	Description
Document Owner	Global Specification Group
Editor / Company	Andreas Brock / Vodafone

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.