



**VIDEO TELEPHONY INTEROPERABILITY**  
**Video Telephony Circuit Switched**  
**Implementation Guidelines**  
**Version 1.0**  
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# 1 INTRODUCTION

The interoperability of circuit switched 3G Video Telephony (VT) systems is key to mass adoption of VT services. The purpose of this document is to accelerate the uptake of VT services by making available implementation guidelines and recommended best practice for 3G operators implementing such services.

These guidelines have been developed out of the experience of operators that have already implemented and deployed CS Video Telephone services. This work augments GSMA PRD SE34.

The principal audience for this document is 3G MNOs considering deployment of peer-to-peer VT services, or those MNOs which have already deployed VT and wish to improve inter-operability.

This document is organized as follows:

- Section 2 deals with the Use Cases for VT.
- Section 3 deals with Implementation Guidelines for CS VT Calling intra-network
- Section 4 deals with implementation Guidelines for CS VT Calling inter-network
- Section 5 deals with roaming
- Section 6 summarises the recommendations
- Section 7 is an Appendix detailing the format and inter-relationship of the use cases

## 1.1 Glossary

The following acronyms are used:

Term	Definition
3G	3rd Generation
3GPP	3 <sup>rd</sup> Generation Partnership Project
AMR	Adaptive Multi Rate
BC	Bearer Capability
BS	Bearer Service
CAP	CAMEL Application Part
CDR	Call Duration Record
CN	Core Network
CS	Circuit Switched
CSCF	Call State Control Function
DCME	Digital Circuit Multiplication Equipment
ECD	Echo Control Device
GW	Gateway
HLC	High Layer Compatibility
HLR	Home Location Register
IDD	International Direct Dialling
IE	Information Element
IoT	Interoperability Testing
IP	Internet Protocol
ISDN	Integrated Service Digital Network
ISP	Internet Service Provider
ITSP	Internet Telephony Service Provider
ISC	International Switching Centre
KPI	Key Performance Indicator
LLC	Low Layer Compatibility
MAP	Mobile Application Part

Term	Definition
MNO	Mobile Network Operator
MNP	Mobile Number Portability
MOC	Mobile Originated Call
MS	Mobile Station
MSC	Mobile Switching Centre
MTC	Mobile Terminated Call
NRT	Non Real Time
OC	Outgoing Call (from Mobile Station to network)
PC	Personal Computer
PLMN	Public Land Mobile Network
PS	Packet Switched
PSTN	Public Switched Telephone Network
QoS	Quality of Service
R99	Release 1999
RAN	Radio Access Network
SCCP	Signalling Connection Control Part
SDO	Standards Development Organisation
SS7	Signalling System No. 7
TAP	Transfer Account Procedure
TMR	Transmission Medium Requirement
TC	Terminating Call (from network to Mobile Station)
UDI	Unrestricted Digital Information
UMTS	Universal Mobile Telecommunications System
UTRAN	UMTS Terrestrial Radio Access Network
VLR	Visitor Location Register
VT	Video Telephony
WG	Working Group

## 2 USE CASES

This section presents a basic set of use cases that are sufficient for a new 3G operator to implement VT services. The format and inter-relationship of the use cases is given in Appendix A.

### 2.1 Make a VT call

#### 2.1.1 User A calls User B (UC1)

Use Case Name	User A calls User B
I want to	Call another person on video
ID Number	UC1
Preconditions – What must be true before Use Case begins	3G GSM network(s) must be up and running, there should be no network problems. User A (call owner – initiator of video call) and User B (video call recipient) must: <ul style="list-style-type: none"> <li>- be within 3G covered area of the 3G network(s)</li> <li>- have 3G terminals supporting video telephony based on the circuit switched bearer service 30 (BS30)</li> <li>- have a 2G SIM card or 3G UICC,</li> <li>- be provisioned for BS30 (or at least for BS30 multimedia) in the HLR</li> <li>- be allowed (and not barred) from calling national and international numbers</li> <li>- Must be post paid subscribers or prepaid users with credit on their account (before and during the call).</li> </ul>
Use Case Body	
Narrative	User A enters the number of User B into handset and presses the call button. User A waits for call to be established and for face of User B to appear.
Supplementary Details and Constraints	
Exceptions— something expected that does not fulfil the user's goal during the execution of a use case e.g. system variations, MT preferences, errors	User A is not registered for video telephony in network and an according error message is displayed on user A UE System cannot establish call to User B Call is established but handsets do not negotiate codecs correctly User loses 3G coverage during call set-up. Prepaid customer runs out of credit
KPIs	User A/B hears ring tone < 5 seconds (after initiating the call by pressing the call button)  Audio/Video Connection fully established < 5 seconds (after B party accepted the call)
Constraints — constraints this use case must conform to	UE time to divert must be greater than call set-up time.
Technical Comment	The 5 second audio/video connection is a challenging KPI. Improvements in call set-up times are currently a work item in 3GPP. See also Section 3 Implementation Guidelines for

<b>Use Case Name</b>	<b>User A calls User B</b>
	intra-network VT; also see section 3.8.4 on KPIs for more details on call set-up measurements.

**2.1.2 User A calls User B on another network (UC2)**

<b>Use Case Name</b>	<b>User A calls User B on another network</b>
I want to	Call another person on video who is on another network
ID Number	UC2
Preamble	
Preconditions – What must be true before Use Case begins	<p>3G GSM network(s) must be up and running, there should be no network problems.</p> <p>User A (call owner – initiator of video call) and User B (video call recipient) must:</p> <ul style="list-style-type: none"> <li>- be within 3G covered area of the 3G network(s) supporting 64 Kbps CS bearers</li> <li>- have 3G terminals supporting video telephony based on the circuit switched bearer service 30 (BS30), have a 2G SIM card or 3G UICC,</li> <li>- be provisioned for BS30 (or at least for BS30 multimedia) in the HLR, be allowed (and not barred) from calling national and international numbers,</li> <li>- must be post paid subscribers or prepaid users with credit on their account (before and during the call).</li> </ul> <p>User A is on PLMN A and User B is on their home network PLMN B</p>
Use Case Body	
Narrative	User A enters the number of User B into handset and presses the video call button. The call is being established from User A to User B. A ringing tone is indicated to User A and User B.
Supplementary Details and Constraints	
Exceptions— something expected that does not fulfil the user's goal during the execution of a use case e.g. system variations, MT preferences, errors	User B is either in another home network (e.g. of operator B) or roaming in another network (e.g. of operator C).
KPIs	<p>User A/B hears ring tone &lt; 5 seconds (after initiating the call by pressing the call button)</p> <p>Audio/Video Connection fully established &lt; 5 seconds (after B party accepted the call)</p>
Constraints — constraints this use case must conform to	The inter-network connections must support video telephony transmission and billing.
Technical Comment	Reference UC1 – same conditions apply regarding call set-up time. See Section 4 for guidelines relating to inter-networking.

**2.1.3 User A calls User B, while User B is roaming on another network (UC3)**

<b>Use Case Name</b>	<b>User A calls User B while B is roaming on another network</b>
I want to	Call on video another person who is roaming on

Use Case Name	User A calls User B while B is roaming on another network
	another network
ID Number	UC3
Preconditions – What must be true before Use Case begins	<p>3G GSM network(s) must be up and running, there should be no network problems.</p> <p>User A (call owner – initiator of video call) and User B (video call recipient) must:</p> <ul style="list-style-type: none"> <li>- be within 3G covered area of the 3G network(s) supporting 64 Kbps CS bearers,</li> <li>have 3G terminals supporting video telephony based on the circuit switched bearer service 30 (BS30),</li> <li>- have a 2G SIM card or 3G UICC,</li> <li>be provisioned for BS30 (or at least for BS30 multimedia) in the HLR,</li> <li>- be allowed (and not barred) from calling national and international numbers,</li> <li>- must be post paid subscribers or prepaid users with credit on their account (before and during the call).</li> </ul> <p>User A is on PLMN A and User B is roaming on another network PLMN B</p>
Use Case Body	
Narrative	User A enters the number of User B into handset and presses the video call button. The call is then established from User A to User B. A ringing tone is indicated to User A and User B.
Supplementary Details and Constraints	
Exceptions— something expected that does not fulfil the user's goal during the execution of a use case e.g. system variations, MT preferences, errors	User B is either in another home network (e.g. of operator B) or roaming in another network (e.g. of operator C).
KPIs	<p>User A/B hears ring tone &lt; 5 seconds (after initiating the call by pressing the call button)</p> <p>Audio/Video Connection fully established &lt; 5 seconds (after B party accepted the call)</p>
Constraints — constraints this use case must conform to	The inter-network connections must support video telephony transmission and billing.
Technical Comment	Reference UC1 – same conditions apply regarding call set-up time. See Section 5 for guidelines relating to roaming.

## 2.2 Accept a VT call

### 2.2.1 User B Receives Call from User A (UC4)

Use Case Name	User B receives VT call from user A
I want to	Receive a call from another person
ID Number	UC4
Preconditions – What must be true before Use Case begins	<p>3G GSM network(s) must be up and running, there should be no network problems.</p> <p>User A (call owner – initiator of video call) and User B (video call recipient) must:</p> <ul style="list-style-type: none"> <li>- be within 3G covered area of the 3G network(s) supporting 64 Kbps CS bearers,</li> <li>- have 3G terminals supporting video telephony based on the circuit switched bearer service 30</li> </ul>

Use Case Name	User B receives VT call from user A
	(BS30), - have a 2G SIM card or 3G UICC, - be provisioned for BS30 (or at least for BS30 multimedia) in the HLR, - be allowed (and not barred) from calling national and international numbers, - must be post paid subscribers or prepaid users with credit on their account (before and during the call).
Use Case Body	
Narrative	User B's handset rings to notify of incoming call. User B notes that the call is a VT call from User A and answers the call. User A appears on screen of User B and User B appears on screen of User A and conversation begins, hearing and seeing each other.
Supplementary Details and Constraints	
Exceptions— something expected that does not fulfil the user's goal during the execution of a use case e.g. system variations, MT preferences, errors	User B rejects call (see UC6) User B ignores call and call is diverted (see UC10) User B answers the call with video muted (see UC5) User B UE is on GSM 2G network of a roaming partner (see UC3) System cannot establish call to User B Call is established but handsets do not negotiate codecs correctly User loses 3G coverage during call set-up. Prepaid customer runs out of credit
KPIs	Audio/Video Connection fully established < 5 seconds (after B party accepted the call)
Constraints — constraints this use case must conform to	UE time to divert must be greater than call set-up time.
Technical Comment	Reference UC1 for note on set-up times.

### 2.3 Answer a Call with video mute (UC5)

Use Case Name	User B receives a VT call and opts to answer the call in mute.
I want to	Accept a video call but not show myself
ID Number	UC5
Preconditions – What must be true before Use Case begins	3G GSM network(s) must be up and running, there should be no network problems. User A (call owner – initiator of video call) and User B (video call recipient) must: - be within 3G covered area of the 3G network(s) supporting 64 Kbps CS bearers, - have 3G terminals supporting video telephony based on the circuit switched bearer service 30 (BS30), - have a 2G SIM card or 3G UICC, be provisioned for BS30 (or at least for BS30 multimedia) in the HLR, - be allowed (and not barred) from calling national and international numbers, - must be post paid subscribers or prepaid users with credit on their account (before and during the call).  Terminal of User B must support video mute.

Use Case Name	User B receives a VT call and opts to answer the call in mute.
Use Case Body	
Narrative	User B's handset rings to notify of incoming call. User B notes that the call is a VT call from User A and opts to answer the call in video mute i.e. User B will be able to see User A's image but User B will not be sending their live image – so their picture is muted. User A will hear User B's voice only.
Supplementary Details and Constraints	
Exceptions— something expected that does not fulfil the user's goal during the execution of a use case e.g. system variations, MT preferences, errors	System cannot establish call to User B Call is established but handsets do not negotiate codecs correctly User loses 3G coverage during call set-up. Prepaid customer runs out of credit
KPIs	Call Connection < 10 seconds User A hears ring tone < 5 seconds Hang-up = 1 second
Post-conditions— what must be true about the system after a use case completes	System must be able to free resources for next calls Update "Calls Made" on handset System generates and stores CDR for call Handset generates "Favourite numbers" Other Handset functions can be called – add to address book Pre-paid active credit management will decrease
Constraints — constraints this use case must conform to	UE time to divert must be greater than call set-up time.
Technical Comment	It is possible that as a handset-resident feature, that User B may have the ability to send a pre-defined image that acts as a "comfort page" to User A.

## 2.4 End the VT call

### 2.4.1 User wishes to end VT call (UC6)

Use Case Name	Users wishes to end video call
I want to	Finish the video call
ID Number	UC6
Preconditions – What must be true before Use Case begins	3G GSM network(s) must be up and running, there should be no network problems. User A (call owner – initiator of video call) and User B (video call recipient) must: <ul style="list-style-type: none"> <li>- be within 3G covered area of the 3G network(s) supporting 64 Kbps CS bearers,</li> <li>- have 3G terminals supporting video telephony based on the circuit switched bearer service 30 (BS30),</li> <li>- have a 2G SIM card or 3G UICC,</li> <li>- be provisioned for BS30 (or at least for BS30 multimedia) in the HLR,</li> <li>- be allowed (and not barred) from calling national and international numbers,</li> <li>- must be post paid subscribers or prepaid users with credit on their account (before and during the call).</li> </ul>

Use Case Name	Users wishes to end video call
Use Case Body	
Narrative	User A is in video call with user B. User A wishes to end video call. User A press end call button. The call is terminated. An end call message is displayed on both UEs.
Supplementary Details and Constraints	
Exceptions— something expected that does not fulfil the user's goal during the execution of a use case e.g. system variations, MT preferences, errors	User B wishes to end the video call. User B presses the end call button. The call is terminated. An end call message is displayed on both UEs. The call is terminated because user A UE has lost 2G or 3G coverage. The call is terminated because user B UE has lost 2G or 3G coverage. The call is terminated because user A UE has lost 3G coverage and a handover to 2G is performed (see UC7). The call is terminated because user B UE has lost 3G coverage and a handover to 2G is performed (see UC7).
KPIs	Hang-up < 1 second
Constraints — constraints this use case must conform to	
Technical Comment	The hang-up KPI is a realistic user experience expectation, however in practice the end to end VT CS data application, the underlying UDI bearer and the CS radio bearers must be torn down, and this process requires optimisation by the MNOs at each end of the call. This may take longer than 1 second. A user therefore is unlikely to be able to make a new CS video call 1 second after hang-up.

## 2.5 Continue the call

### 2.5.1 Switch to voice if VT coverage is lost (UC7)

Use Case Name	Switch to voice if VT coverage is lost
I want to	Initiate a video call and the call reverts to a voice call should VT service be lost.
ID Number	UC7
Preconditions – What must be true before Use Case begins	3G and/or GSM network(s) must be up and running, there should be no network problems. User A (call owner – initiator of video call) and User B (video call recipient) must: <ul style="list-style-type: none"> <li>- be within 3G covered area of the 3G network(s) supporting 64 Kbps CS bearers,</li> <li>- have 3G terminals supporting video telephony based on the circuit switched bearer service 30 (BS30),</li> <li>- have a 2G SIM card or 3G UICC,</li> <li>- be provisioned for BS30 (or at least for BS30 multimedia) in the HLR,</li> <li>- be allowed (and not barred) from calling national and international numbers,</li> <li>- must be post paid subscribers or prepaid users with credit on their account (before and during the call).</li> </ul>
Use Case Body	
Narrative, Scenario or Conversation	During the video call one of the parties moves to such a degree that they move from the 3G

Use Case Name	Switch to voice if VT coverage is lost
	coverage area to that of a non VT environment.  If the video call fails, there shall be the option for User A to start a simple voice call.
Supplementary Details and Constraints	
Exceptions— something expected that does not fulfil the user's goal during the execution of a use case e.g. system variations, MT preferences, error	Neither 3G nor 2G coverage is available and call will be released. User A moves back into 3G coverage but the call will not revert back to a video call. User A is not registered for video telephony in network and a corresponding error message is displayed on user A UE System cannot establish call to User B Call is established but handsets do not negotiate codecs correctly User loses 3G coverage during call set-up. Prepaid customer runs out of credit
KPIs	Audio Connection fully established < 5 seconds (after initiating the call by pressing the call button)
Constraints — constraints this use case must conform to	The serving network or the terminal must be able to detect that the call cannot continue as video and offers to reconnect as voice while ensuring integrity and transparency of billing.
Technical Comment	It is possible to provide voice services on either 3G or 2G networks when video service is unavailable. As such the service change required above can be supported inter-network (3G VT to 2G voice) or intra-network (3G VT to 2G voice). This is being discussed as a work item in 3GPP ( SCUDIF may be one solution)

### 2.5.2 Switch to voice if B not in 3G coverage (UC8)

Use Case Name	Switch to voice if B not in VT coverage.
I want to	Initiate a video call and the call reverts to a voice call as the person receiving is not within VT coverage.
ID Number	UC8
Preconditions – What must be true before Use Case begins	3G GSM network(s) must be up and running, there should be no network problems. User A (call owner – initiator of video call) must: - be within VT covered area of the 3G network(s) supporting 64 Kbps CS bearers, - have 3G terminals supporting video telephony based on the circuit switched bearer service 30 (BS30), - have a 2G SIM card or 3G UICC, - be provisioned for BS30 (or at least for BS30 multimedia) in the HLR, - be allowed (and not barred) from calling national and international numbers, - must be post paid subscribers or prepaid users with credit on their account (before and during the call). User B (video call recipient) is within a non VT coverage area.
Use Case Body	
Narrative, Scenario or Conversation	The B Party is not within a 3G coverage area, User A shall be prompted to start a simple voice

Use Case Name	Switch to voice if B not in VT coverage.
	call. The call successfully connects.
Supplementary Details and Constraints	
Exceptions— something expected that does not fulfil the user's goal during the execution of a use case e.g. system variations, MT preferences, error	A Party does not want to have the call complete as a voice call and thus terminates the call before completion.
KPIs	Audio Connection fully established < 5 seconds (after initiating the call by pressing the call button)
Constraints — constraints this use case must conform to	The ability for the network to identify the B Party network connection and appropriately connect the call. Successful conversion of call from video initiated call to that of a voice call.
Technical Comment	It should be noted that 3G coverage and VT coverage are not the same. 3G coverage can provide video and audio coverage, depending on the availability of a CS 64 Kbps bearer. I.e. party B could be in a 3G coverage area, but without a 64kbps bearer available would only have voice services supported.

### 2.5.3 Call switches to voice as B Party is roaming (UC9)

Use Case Name	Switch to voice as B party is roaming
I want to	Initiate a video call and the call reverts to a voice call as the person taking the call is roaming.
ID Number	UC9
Preconditions – What must be true before Use Case begins	3G GSM network(s) must be up and running, there should be no network problems. User A (call owner – initiator of video call) must: <ul style="list-style-type: none"> <li>- be within VT covered area of the 3G network(s),</li> <li>- have 3G terminals supporting video telephony based on the circuit switched bearer service 30 (BS30),</li> <li>- have a 2G SIM card or 3G UICC,</li> <li>- be provisioned for BS30 (or at least for BS30 multimedia) in the HLR,</li> <li>- be allowed (and not barred) from calling national and international numbers,</li> <li>- must be post paid subscribers or prepaid users with credit on their account (before and during the call).</li> </ul> User B (video call recipient) is roaming and is unable to establish a VT call.
Use Case Body	
Narrative, Scenario or Conversation	The B Party is roaming within a non VT coverage area. User A shall be prompted to start a simple voice call. The call successfully connects.
Supplementary Details and Constraints	
Exceptions— something expected that does not fulfil the user's goal during the execution of a use case e.g. system variations, MT preferences, error	A Party does not want to have the call complete as a voice call and thus terminates the call before completion.
KPIs	Audio Connection fully established < 5 seconds

<b>Use Case Name</b>	<b>Switch to voice as B party is roaming</b>
	(after initiating the call by pressing the call button).
Constraints — constraints this use case must conform to	Successful conversion of call from video initiated call to that of a voice call. The ability for the network to identify the B Party network connection and appropriately connect the call. Alternatively the system may renegotiate the call as voice if diverting to voicemail as a customer preference when roaming.
Technical Comment	Reference UC7 and UC8.

## 2.6 Divert incoming VT calls

### 2.6.1 User A Redirects Incoming Video Calls (UC10)

<b>Use Case Name</b>	<b>User A establishes call directs for video calling</b>
I want to	Divert incoming video calls
ID Number	UC10
Preconditions – What must be true before Use Case begins	3G GSM network(s) must be up and running, there should be no network problems. User A (call owner – initiator of video call) and User B (video call recipient) must: <ul style="list-style-type: none"> <li>- within 3G covered area of the 3G network(s) supporting 64 Kbps CS bearers,</li> <li>- have 3G terminals supporting video telephony</li> <li>- based on the circuit switched bearer service 30 (BS30),</li> <li>- have a 2G SIM card or 3G UICC,</li> <li>- be provisioned for BS30 (or at least for BS30 multimedia) in the HLR,</li> <li>- be allowed (and not barred) from calling national and international numbers,</li> <li>- must be post paid subscribers or prepaid users with credit on their account (before and during the call).</li> </ul>
Use Case Body	
Narrative	User A selects to redirect their video calls to another number. They find the settings for video calling and enter a number for the redirection of video calls.
Supplementary Details and Constraints	
Exceptions— something expected that does not fulfil the user's goal during the execution of a use case e.g. system variations, MT preferences, errors	System cannot establish call to User B Call is established but handsets do not negotiate codecs correctly User loses 3G coverage during call set-up. Prepaid customer runs out of credit
KPIs	User A/B hears ring tone < 5 seconds (after initiating the call by pressing the call button)  Audio/Video Connection fully established < 5 seconds (after B party accepted the call)  Hang-up < 1 second
Constraints — constraints this use case must conform to	UE time to divert must be greater than call set-up time.
Technical Comment	The most common example of this is with diversion to Videomail. This diversion can be triggered by the network when the handset is not

<b>Use Case Name</b>	<b>User A establishes call directs for video calling</b>
	switched on otherwise unavailable. Alternatively it could be triggered by user choice via the handset UI. However the diversion mechanism remains a network feature and is not terminal resident. See also 3.6.1 re VT Supplementary Services.

### 2.6.2 User A is diverted to User C on 3<sup>rd</sup> network (UC11)

<b>Use Case Name</b>	<b>User A calls User B and is diverted to User C.</b>
I want to	Divert incoming calls to someone on a different network
ID Number	UC11
Preconditions – What must be true before Use Case begins	<p>3G GSM network(s) must be up and running, there should be no network problems.</p> <p>User A (call owner – initiator of video call), User B (video call recipient) and User C must:</p> <ul style="list-style-type: none"> <li>- be within 3G covered area of the 3G network(s),</li> <li>- have 3G terminals supporting video telephony based on the circuit switched bearer service 30 (BS30),</li> <li>- have a 2G SIM card or 3G UICC,</li> <li>- be provisioned for BS30 (or at least for BS30 multimedia) in the HLR,</li> <li>- be allowed (and not barred) from calling national and international numbers,</li> <li>- must be post paid subscribers or prepaid users with credit on their account (before and during the call).</li> </ul> <p>User A shall be on network X, User B shall be on network Y and User C shall be on network Z.</p>
Use Case Body	
Narrative	<p>User B's number is dialled and a connection is attempted.</p> <p>Call is then redirected to User C.</p> <p>The call is established from User A to User C.</p> <p>A ringing tone is indicated to User A and User C.</p>
Supplementary Details and Constraints	
Exceptions— something expected that does not fulfil the user's goal during the execution of a use case e.g. system variations, MT preferences, errors	User A, B and C can either in their home network (e.g. of operator X, Y or Z) or roaming in another network (e.g. of operator X, Y or Z).
KPIs	<p>User A/B hears ring tone &lt; 5 seconds (after initiating the call by pressing the call button)</p> <p>Audio/Video Connection fully established &lt; 5 seconds (after B party accepted the call)</p> <p>Hang-up &lt; 1 second</p>
Constraints — constraints this use case must conform to	
Technical Comment	This is known as VT call forwarding in a direct analogy of traditional voice call forwarding.

## 2.7 Put call on hold

### 2.7.1 User A makes a second call, voice or video, putting B on hold. (UC12)

Use Case Name	User A puts User B on hold.
I want to	Put B on hold and make a 2 <sup>nd</sup> call
ID Number	UC12
Preconditions – What must be true before Use Case begins	<p>3G GSM network(s) must be up and running, there should be no network problems.</p> <p>User A (call owner – initiator of video call) and User B (video call recipient) must:</p> <ul style="list-style-type: none"> <li>- be within 3G covered area of the 3G network(s),</li> <li>- have 3G terminals supporting video telephony based on the circuit switched bearer service 30 (BS30),</li> <li>- have a 2G SIM card or 3G UICC,</li> <li>- be provisioned for BS30 (or at least for BS30 multimedia) in the HLR,</li> <li>- be allowed (and not barred) from calling national and international numbers,</li> <li>- must be post paid subscribers or prepaid users with credit on their account (before and during the call).</li> </ul>
Use Case Body	
Narrative	<p>User A and User B are in a live VT call. User A decides to hold the call notifies User B and selects the hold option.</p> <p>User B receives on-hold clip. User A performs second call.</p> <p>Video call continues as soon as User A decides to take User B off hold.</p>
Supplementary Details and Constraints	
	<p>User B decides to hold the call, notifies User A and selects the hold option. User A receives User B's hold clip.</p> <p>User A network can send a signal to User B UE to display an on hold video.</p>
Exceptions— something expected that does not fulfil the user's goal during the execution of a use case e.g. system variations, MT preferences, errors	<p>System cannot establish call to User B</p> <p>Call is established but handsets do not negotiate codecs correctly</p> <p>User loses 3G coverage during call set-up.</p> <p>Prepaid customer runs out of credit</p>
KPIs	
Design Note	Do the users exchange hold clips or is there only one clip sent by the user who opted to hold the call?
Constraints — constraints this use case must conform to	UE time to divert must be greater than call set-up time.
Technical Comment	The Supplementary Service "Hold" is not supported on BS30 as this is primarily a data circuit. A change was requested at 3GPP but this has not been progressed. The VTIOF team is to consider approaching 3GPP CN4 for advancing this issue.

## 2.8 Accept a second call

### 2.8.1 User A takes 2<sup>nd</sup> Incoming Call (UC13)

Use Case Name	User C makes a call to User A whilst user A is in a VT call with user B.
I want to	Accept a 2 <sup>nd</sup> call while on a call
ID Number	UC13
Preconditions – What must be true before Use Case begins	<p>3G GSM network(s) must be up and running, there should be no network problems.</p> <p>User C calls (call owner) User A (initiator of video call to User B) and User B (video call recipient). User A, User B and User C must:</p> <ul style="list-style-type: none"> <li>- be within VT coverage area of the 3G network(s),</li> <li>- have 3G terminals supporting video telephony based on the circuit switched bearer service 30 (BS30),</li> <li>- have a 2G SIM card or 3G UICC,</li> <li>- be provisioned for BS30 (or at least for BS30 multimedia) in the HLR,</li> <li>- be allowed (and not barred) from calling national and international numbers,</li> <li>- must be post paid subscribers or prepaid users with credit on their account (before and during the call).</li> </ul>
Use Case Body	
Narrative	User A and User B are in a live VT call. User C calls user A via video or voice call. User A is notified of user C's call and opts to answer the call and place user B on hold. User A speaks to user C then ends that call and retrieves user B to continue their call.
Supplementary Details and Constraints	
Exceptions— something expected that does not fulfil the user's goal during the execution of a use case e.g. system variations, MT preferences, errors	<p>User A opts not to answer user C's call</p> <p>User A opts to reject user C's call</p> <p>User A ends their call with user B and answers user C's call</p>
KPIs	<p>User A/B hears ring tone &lt; 5 seconds (after initiating the call by pressing the call button)</p> <p>Audio/Video Connection fully established &lt; 5 seconds (after B party accepted the call)</p>
Constraints — constraints this use case must conform to	UE time to divert must be greater than call set-up time.
Technical Comment	User B must be notified that they have been placed on hold through either an audio or video message. In general hold is an issue as regards Supplementary Services. See UC 12.

## 2.9 Additional Use Cases

The following use cases were also identified as part of the process to identify all use cases in relation to Video Telephony. However they were not felt to be a core part of the VT experience and were thus not further developed after the VTIOF Meeting in Amsterdam,

December 7, 2005. They are however listed here for reference and the Use Cases which were presented in Amsterdam are stored separately on the Infocentre.

<b>Conferencing Functionality</b>
1. Conference in another calling party as a video call
2. Conference in another calling party as an audio call
3. The B party can conference in another calling party as a video call
4. The B party can conference in another calling party as an audio call
<b>Videomail Functionality</b>
5. Reply to a video mail
6. Save a Videomail
7. Receive receipts for Videomail
8. User is diverted to Answer Phone
<b>Automatic Service Change</b>
9. Network downgrades call to voice
10. A Party upgrades call from voice to video
11. A Party downgrades from video to voice
12. Revert to video after using voice
<b>Other</b>
13. User B receives a VT call as voice
14. Call is continued as voice after network prompt caused by call degradation

### 3 INTRA-NETWORK IMPLEMENTATION GUIDELINES FOR CS VT CALLING IN A 3G NETWORK

#### 3.1 Purpose and scope

- The purpose of this section is to provide implementation guidelines to 3G operators for implementation of Video Telephony within a 3G network. The scope is therefore strictly intra-network CS VT implementation.
- Please refer to section 4 of this document for implementation guidelines relating to inter-network CS VT.
- Please refer to section 5 of this document for implementation guidelines relating to CS VT roaming.

#### 3.2 Reference documents

1. 3GPP TS 26.110 Codec for Circuit switched Multimedia Telephony Service; General Description
2. 3GPP TS 26.111 Codec for Circuit switched Multimedia Telephony Service; Modifications to H.324
3. 3GPP TR 26.911 Codec for Circuit switched Multimedia Telephony Service; Terminal Implementer's Guide
4. 3GPP TR 26.912 Codec for Circuit switched Multimedia Telephony Service; Quantitative performance evaluation of H.324 Annex C over 3G
5. 3GPP TR 23.972 Circuit Switched Multimedia Telephony
6. 3GPP TS 26.101: "AMR Speech Codec Frame Structure"
7. 3GPP TR 24.008
8. 3GPP TR 22.002
9. 3GPP TR 27.001
10. 3GPP TR 29.007
11. 3GPP TR 23.008
12. ITU-T Recommendation H.324: "Terminal for low bitrate multimedia"
13. ITU-T Recommendation H.245: "Control Protocol for multimedia communication"
14. ITU-T Recommendation G.723: "Dual Rate Speech Coder for multimedia communications transmitting at 5.3 and 6.3kbps"
15. ITU-T Recommendation H.223: "Multiplexing protocol for low bitrate multimedia communication"
16. ITU-T Recommendation H.223/Annex B: "Multiplexing protocol for low bitrate multimedia communication over moderate error-prone channels"
17. ITU-T Recommendation H.223/Annex A: "Multiplexing protocol for low bitrate multimedia communication over low error-prone channels"
18. ITU-T Recommendation H.263: "Video coding for low bitrate communication"
19. [19] ITU-T Recommendation H.263: "Annex X, Profiles and Levels Definition"
20. Reference to ITU WNSRP Proposal
21. Fast session set-up extensions to H.324, Submission to ITU Study Group 16 on 16-26 November 2004 from Dilithium Networks
22. ETSI Spec (ETSI/STQ Mobile (04) 03TD09, ETSI number TS 102 250-2, CR number 250-2 009

#### 3.3 Terminals

##### 3.3.1 Overview

3G Video Telephony in its current form, based on the 3G-324M standard and using the 64Kbps circuit switched BS30 bearer can be considered terminal centric since when the

underlying bearer is established, the network acts as a transparent “pipe”. There are two clear phases to the establishment of a VT call:

1. Establishment of the underlying UDI Bearer between a terminal and an end-point. The end-point in the scope of this activity is considered to be another wireless 3G-324M terminal on the same network.
2. Establishment of the Video Telephony session using 3G-324M.

This chapter of the document examines the terminal perspective of Video telephony and specifically the interoperability issues and best practices that can be implemented. The chapter is divided into two sections that consider the bearer and 3G-324M session establishment separately.

Both these sections cover known implementation issues that contributing operators have experienced, along with recommendations to optimise Video Telephony implementation from an interoperability perspective.

### **3.3.2 Bearer Requirements**

The focus of this document are the intra-network scenarios that inherently have the least risk of bearer interoperability issues, because the end-to-end connection should be within the control of one operator and there is unlikely to be a reliance on third party carriers or interconnect providers.

However, there are some measures which should be taken from the terminal perspective to ensure the bearer is established correctly and with the least risk of a failure occurring due to IOT.

Obviously the terminal must support the circuit-switched UDI transparent bearer at 64 Kbps as specified in [7], [8], [9], [10], and [11]. In addition to general support of the bearer, the terminal must differentiate a multimedia call from a standard 64 Kbps transparent call by using the Bearer Capability Information Element (BC IE) and the Lower Layer Capability (LLC) fields in the SET-UP message when originating a call. Specifically the BC IE “other rate adaptation” (ORA) parameter must equal H.223&H.245 to signal that the 64 Kbps call is intended to be a multimedia call. The LLC must set the 'user information layer 1 protocol' equal to H.223&H.245.

Equally on the Mobile Terminating side the terminal must use these indicators to correctly accept the call as a Multimedia 64 Kbps CS call rather than a standard transparent circuit switched call.

The requirement for both fields is to overcome the case where data from the BC IE field can be lost in some scenario's e.g. ISUPv1 interconnect due to truncation of the fields.

### **3.3.3 Multimedia Protocol & Codec Implementation**

As discussed in 2.1, the 3G-324M standard specifies the protocols used by devices to implement a conversational Video Telephony session across the underlying CS bearer. 3G-324M is defined in [2], [12] and incorporates the H.223 [15], [16], [17], H.245 [13] standards and also the codecs defined for use in audio and video transmission.

From a terminal interoperability perspective, issues can occur in intra-network scenarios with both codecs and protocol interoperability. In the case of both areas implementation choices can be made to reduce the risk of call failure or degradation due to compatibility differences.

3G-324M implementations use H.223 for multiplexing and H.245 for control signalling respectively, each of these have a number of definable parameters and configurable options

that give rise to variation between implementations and an inherent risk of interoperability failures. The standard is now mature and has been well stressed in organisations such as IMTC (International Multimedia Telecommunications Consortium) and commercial roll-outs globally, therefore, interoperability issues are typically misinterpretations of the specification or errors in implementation due to lack of interoperability testing.

### **3.3.4 Multiplexing Protocols**

In the case of H.223, different annexes are specified which provide different levels of robustness against erroneous environments as are also different Adaptation Layers (AL) offering either unidirectional or bi-directional logical channels for media.

The 3GPP implementers Guide [3] already makes recommendations in this area that should be followed. Specifically both video and audio media should be transmitted over H223 AL2 unidirectional logical channels; this differs from the original H.324 standard where AL3 was recommended for video media. AL3 offers retransmission capabilities which had some merit in the fixed line environment but are inappropriate for wireless networks where excessive delay can be introduced to a conversational service.

The basic H.223 bit-stream does not have sufficient error resiliency to operate reliably in mobile wireless environments. Therefore additional annexes A, B and C were added to the H.324 standard to provide progressively increased robustness at the cost of increased overhead and complexity.

- H.223 Level 0 standard H.223 without any additional annexes for error robustness
- H.223 Level 1 refers to the use of H.223 Annex A. The HDLC flag in H.223 used to delimit MUX-PDUs in level 0 is replaced with a longer flag that leads to improved MUX-PDU synchronisation. HDLC bit stuffing is not used. The control channel segmentation and reassembly layer (CCSRL) is introduced for transmission of the control channel.
- H.223 Level 2 refers to the use of H.223 Annex B. In addition to the features of Annex A, the header describing the MUX-PDU contents includes error protection.
- H.223 Level 3 refers to the use of H.223 Annex C. In addition to the features of Annex B, error protection and other features are provided to increase the protection of the AL-PDUs.

The recommendation is to use either annex A or B (Level 1 or Level 2) as level 0 offers no benefit and the overhead introduced through level 3 redundancy makes it inefficient for low bandwidth usage. The de facto and preferred annex for mobile operation is B (level 2). However, all terminals must support level 0 as a baseline implementation.

### **3.3.5 Control Protocols**

The control protocol, H.245 provides the functionality for master-slave determination, capability exchange, management of logical channel and multiplex tables in addition to other commands and indications.

#### **3.3.5.1 SRP/NSRP**

H.245 uses Simple Retransmission Protocol (SRP) and/or Numbered SRP (NSRP) and Control Channel Segmentation and Reassembly Layer (CCSRL). When terminals communicate using H.223 Level 0 then SRP is used and NSRP can be employed when H.223 Level 2 is used.

A number of interoperability issues can occur if rules of SRP are not followed, typical examples of where implementations typically cause IOT failures and where clear recommendations can be made are:

A terminal must not transmit a new SRP frame until an acknowledgement to the previous frame has been received. This has been witnessed in implementations as a technique to accelerate 3G-324M establishment time but contravenes the specification and can lead to interoperability issues. Currently the introduction of a new technique called Windowed NSRP (WNSRP) [20] is under discussion and when/if standardised allows for accelerated establishment by allowing new SRP frames to be transmitted without receiving acknowledgements for the previous frame.

Even if the terminal (transmitter) retransmits an SRP command frame at short interval (100ms or less), the other terminal (receiver) should be able to read in all the SRP frames correctly and continue the processing. (It may occur that the retransmitted SRP command frame (SN=n) and the initial sending of the subsequent SRP command frame (SN=n+1) are transmitted in a short period.)

The terminal shall be able to continue the normal call procedures even if it receives a SRP frame including multiple H.245 messages as typical implementations use this method of multiple messages/SRP frames to reduce call establishment delay.

Terminals must not expect the SRP frames sequence number to always start from 0; it is valid for the initial sequence number to be any value.

#### *3.3.5.2 Capability Exchange*

Another problem area of 3G-324M implementations can be the Terminal Capability Set (TCS); this is where an implementation declares its capabilities and available options. Issues arise due to incorrect declarations of unsupported capabilities such as codec configurations that are not possible when validly requested by the other device based on the TCS declaration. It is recommended that terminal implementations follow the H.324 specifications relating to TCS declarations and only declare a capability or support for a configuration where the device supports that capability or configuration.

#### *3.3.5.3 Multiplex Table Entries*

The phase of Multiplex Table Entry Exchange (MTE) is also an area where interoperability issues occur due to specification misinterpretations or implementation errors, it is recommended that terminals follow the specification and ensure the flexibility is available to both support multiple MTE requests in the same H.245 message and the reception of MTE requests over a number of H.245 messages as both implementations are valid according to the specification.

#### *3.3.5.4 Logical Channel Establishment*

Problems with Logical Channel establishment for media transmission are possible between implementations where different configurations are applied and particularly if errors have been made in the TCS as described above.

In most cases the 3G-324M standard allows for logical channel conflicts and provides ability for controlled closing and reopening with agreed configurations. However, although this overcomes in many cases call failure scenarios it can significantly delay the establishment of the call as additional round trip delays are introduced before media transmission commences.

A specific issue related to logical channel configuration is a common misinterpretation of the standard whereby the configuration information sent as a DCI parameter in the non collapsing field of the Open Logical Channel (MPEG4) must include the Visual Object Sequence end code. Such misinterpretation contravenes the specification of ISO and can cause issues of interoperability between devices in an intra-network scenario.

The behaviour of a terminal in the scenario of LC failure should also be considered. If there is a failure in logical channel set-up that would cause video in one or both directions to fail it is recommended that the terminal terminates the call in a controlled way when it will not be possible to complete the establishment successfully. (The reason of this recommendation is that this case is expected to be an unusual scenario but it has been observed by operators that terminals are not consistent in their behaviour following partial LC establishment failure. Implementations have been observed to allow an “incomplete” call e.g. video only working in one direction to continue. This is of importance because the end user is charged based on the UDI connection and not the successful transmission of media.)

**3.3.6 Media Exchange**

The final step in the call establishment is the transmission of media bit-streams for audio and video across the logical channels. 3G-324M defines a number of codecs that can be used in Video and Audio which can lead to severe user impacting interoperability issues, in the worst case, the audio or video bit-stream will not be decoded due to an incompatibility issue.

The CODEC options are as follows:

Media	CODEC	Comment
Video	H263 Profile 0 Level 10	Mandatory in 3GPP
	H.263 Profile 3 Level 10	
	MPEG-4 Simple Visual Profile Level 0	Recommended by This Working Group
Audio	3GPP AMR	Mandatory in 3GPP
	G723.1	

For video the default standard of most implementations is MPEG4 SVP. H.263 baseline is mandatory in 3GPP and common to MPEG4 at the core but MPEG4 is marginally superior to H.263 in performance and preferred generally by the industry.

In the case of Audio, AMR is the default codec used in Video Telephony; G723.1 is still implemented in some cases and has potential uses for interoperability with fixed line environments when transcoding gateways are not in place but is currently not typically used.

In the interest of interoperability it is preferred that the industry settle on common codecs that are superior from an achievable quality and implementation perspective, therefore it is recommended that all devices use MPEG4 SVP Level 0 as default for video and AMR @ 12.2 kbps for Audio.

H263 Profile 0 Level 10 is mandated in 3GPP and must also be supported by all devices as a fallback in the case where MPEG4 is not available.

H263 Profile 3 Level 10 (baseline with Annex I, J, K, and T) is not considered a high priority as it provides no quality advantages over MPEG4 and is less commonly implemented.

**3.3.7 Summary of Terminal Recommendations**

Section	Area	Recommendation
2.3.1-1	H.223 Adaptation Layers	Audio and Video Media should use H.223 AL2 unidirectional logical channels
2.3.1-2	H.223 Multiplex Annexes	All devices must support level 0 as a baseline implementation and its recommended that all terminals also support Annex B (Level 2)
2.3.2-1	SRP	Terminals must not transmit a new SRP frame until an acknowledgement to the previous frame has been received
2.3.2-2	SRP	The terminal shall be able to continue the normal call

Section	Area	Recommendation
		procedures even if it receives a SRP frame including multiple H.245 messages
2.3.2-3	SRP	Terminals must not expect the SRP frames sequence number to always start from 0; it's valid for the initial sequence number to be any value.
2.3.2-4	MTE	It is recommended that terminals follow the specification and ensure the flexibility is available to both support multiple MTE requests in the same H.245 message
2.3.2-5	MTE	It is recommended that terminals follow the specification and ensure the flexibility is available to support the reception of an MTE request over a number of H.245 messages
2.3.2-6	LC Establishment	If there is a failure in logical channel set-up that would cause video in one or both directions to fail it is recommended that the terminal terminates the call in a controlled way
2.3.3-2	Codecs	All devices use MPEG4 SVP Level 0 as default for video and AMR @ 12.2 kbps for Audio

### 3.4 Core Network

The purpose of this chapter is to provide guidelines for an appropriate Core Network implementation in a 3G network in order to deliver CS VT services.

In the 3G Core Network the Synchronous Circuit Switched General Bearer Service 30 (BS30) is supported with the following characteristic according to 3GPP TS 22.002, TS 24.008, TS 27.001 and TS 29.007 chapter 9.4:

- 64 Kbps
- Multimedia Mode
- UDI
- Layer1 protocol=H.223 & H.245 – for video telephony (VT)

The *Video Telephony* service provides real-time Multimedia services with a transmission rate of 64 Kbps. It is realized transparent, synchronous, circuit switched and unrestricted digital and uses the UMTS system, which enables significantly higher data rates over the air interface.

This feature is based on the H.324/M (H.324 with extensions for mobile transmission). This standard is not compatible to the H.324 standard and therefore only Mobile-to-Mobile calls are possible (except the case that a fixed network terminal is capable to handle H.324/M calls).

A multimedia gateway is required for calls from H.324/M (mobile) to H.320 (ISDN).

#### 3.4.1 VT Supplementary Services

##### 3.4.1.1 Billing Problem with Call Forwarding Unconditional (CFU)

The following problem for mobile terminating calls with CFU invocation has been revealed. In certain GMSC implementations the CDR-field "Basic Service" will be assigned with a default value (e.g. TS11). The Basic Service cannot be derived in the GMSC.

It is possible to derive BS30 at the GMSC, as the ISUP signalling will have all the relevant information. In this case the switch manufacturer has chosen not to extract the information and update the CDR.

This implies that the leg from subscriber B (forwarding subscriber) to subscriber C (forwarded-to-number) is billed as for speech calls. As long as the price is the same this is not a big problem, but for different BS30 billing it is clearly a problem.

3.4.1.2 Billing Problem with CFNRy (single numbering scheme)

The following problem for mobile terminating calls with CFNRy has been revealed. It may occur for single-numbering subscribers and only when the basic service cannot be derived from the information received from the originating side (which is likely for ISDN originated mobile terminating calls).

In cases where the basic service cannot be derived from the information received from the originating side for a single numbering scheme a SET-UP message will be sent out to the MS containing an empty Bearer Capability information element. The MS itself may now choose a service and include the respective Bearer Capability information in CALL CONFIRMED.

The MSC will apply a network facility check on the received Bearer Capability received in CALL CONFIRMED to determine whether the BC is supported by the network but will not apply the subscription check defined as optional. In that way the subscriber might even use services he is not subscribed to. However this applies for terminating calls only.

It shows how a call with undefined basic service is received for a single-numbering subscriber with activated CFNRy supplementary service. The subscriber does not connect the call and so it is forwarded according to the forwarded-to-number to a C-party. The original BC-information (basic service can not be derived) is used. So the call A-C can still become a BS30 call, if whatever terminal C confirms it with that basic service. However for the CDR in VMSCb the information received in CALL CONFIRMED is stored, in our example a BS20. In case of different BS30 billing the leg B-C will be charged wrongly.

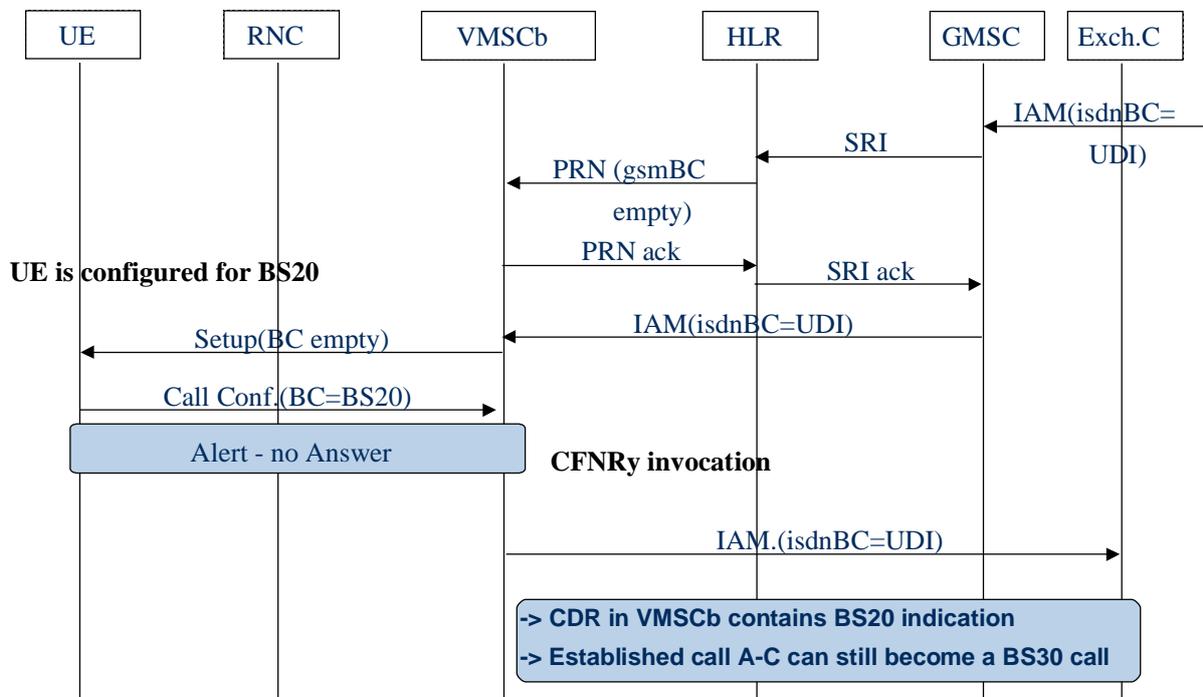


Figure 1: Example of ISDN originated mobile terminating call with CFNRy invocation

3.4.1.3 Call Hold

The SS Call Hold is not supported in the Core Network by BS 30.

3.4.2 Numbering

For simplicity and from a Product Management perspective it is preferred that the video telephony service be implemented using the same MSISDN as for TS11 (speech).

For the existing multi-numbering customers we recommend supporting a mixed numbering scheme with a single number for voice and VT while keeping the other user numbers (e.g. fax) for other services.

**Note:** however that this is not supported by certain HLR vendors nor IT systems.

3.4.3 Core Network Settings

The BS30 in a network must be configured and activated. Otherwise the BS30 can't be used for the application Video Telephony.

- BasicService= 64 Kbps (Value vendor depending e.g. "BS30GENR")
- ConnectionElement = Transparent (Value vendor depending e.g. "TR")
- BS30 = Active (Value vendor depending e.g. "ACT")

3.4.3.1 Connection Element (CONNEL) regarding BS30GENR

This is an example of a vendor implementation:

```

1  MODMSERVOPT:BSERV=BS30GENR,CONNEL=TR;
2
3  DISPMSERVOPT:BSERV=BS30GENR;
4
5  SERVICE/      CONNECTION  RADIO CHANNEL
   ALTERNATIVE
6  FEATURE  STATUS ELEMENT   REQUIREMENT  SCI  OCI
   SERVICE
7  -----+-----+-----+-----+-----+-----+-----
8  BS30GENR  ACT    TR      FRCH
9
    
```

3.4.3.2 PLMN Bearer Capability Info Element

The BS30 basic service is determined by Bearer Capability and LLC information element. The HLC information element is not used by the PLMN.

Relevant parameter layout for VT:

PLMN BC (mandatory)	
Transfer mode	circuit mode
Information transfer capability	unrestricted digital
Structure	unstructured
Rate adaptation	other rate adaptation
Other rate adaptation	H.223 & H.245
Signalling Access protocol	I.440
Synchronous/Asynchronous	synchronous
Connection element	transparent
Modem type	none

**PLMN BC (mandatory)**

User info layer 2 protocol not available

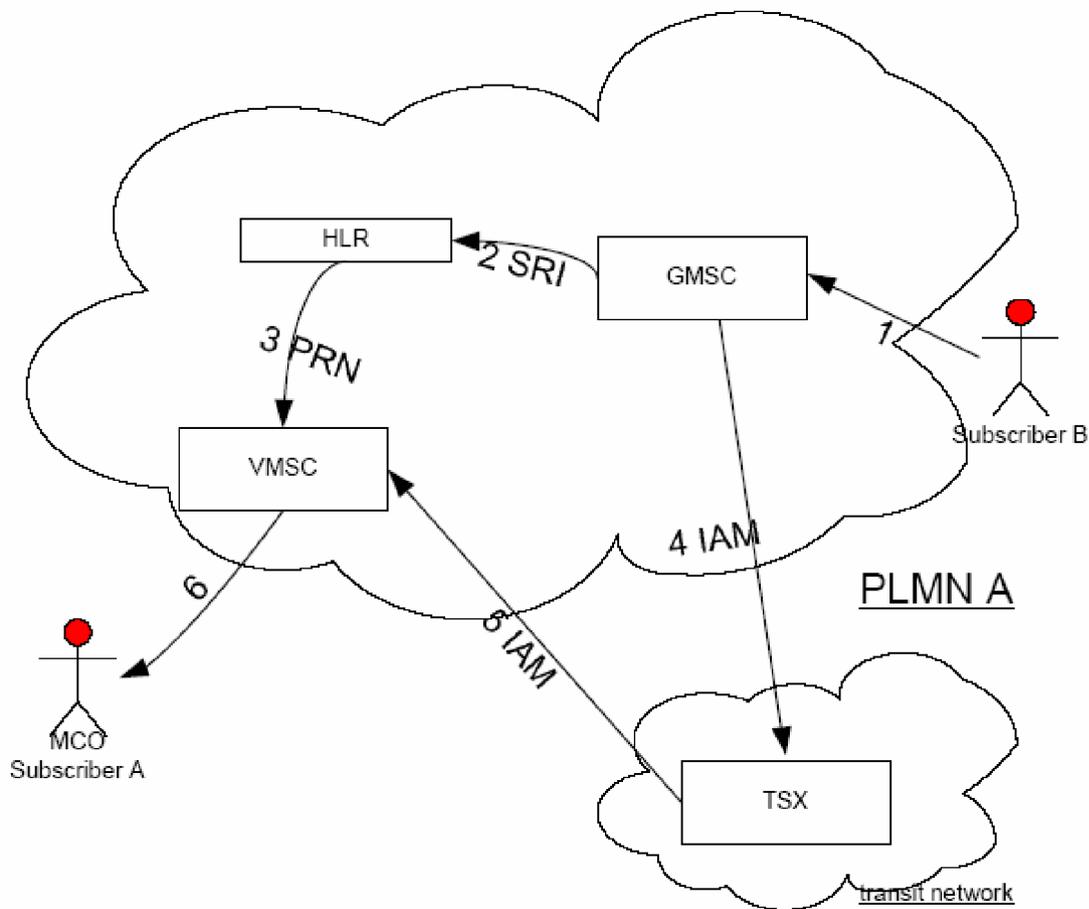
8	7	6	5	4	3	2	1	Octet
0	0	0	0	0	1	0	0	octet 1 = 04 <sub>H</sub>
Bearer capability IEI								
Length of the bearer capability contents								octet 2 =
1	0	1	0	0	0	0	1	octet 3 = A1 <sub>H</sub>
ext	radio channel requirement	coding std= GSM	Trans Fer Mode= Circuit Mode	information transfer capability (ITC)= Unrestricted Digital Information				
1	0	1	1	1	0	0	0	octet 4 =B8 <sub>H</sub>
ext	comp-ress.	Structure= Unstructured	dupl. Mode= Full Duplex	confi gur.= Point to point	NIRR	establi.		
0	0	0	1	1	0	0	1	octet 5 = 19 <sub>H</sub>
ext	access id.	Rate Adaption= Other Rate adaptation	signalling access protocol= I.440/450					
1	0	0	0	1	0	0	0	octet 5a = 88 <sub>H</sub>
ext	Other ITC	Other rate Adaption (ORA) H.223 & H.245	Spare					
0	0	1	0	0	0	0	0	octet 6 = 20 <sub>H</sub>
ext	layer 1 id.	User information layer 1 protocol= default layer 1 protocol	sync/ Async = Sync					
0	0	0	1	0	1	0	1	octet 6a = 15 <sub>H</sub>
ext	numb. stop bits	negotiation= In band neg not poss	numb. data bits	user rate = 9600 bits/s				
0	1	1	0	0	0	1	1	octet 6b = 63 <sub>H</sub>
ext	intermed. Rate	NIC on TX	NIC on RX	Parity				
0	0	0	0	0	0	0	0	octet 6c = 00 <sub>H</sub>
ext	connection element= Transparent	Modem type = None						
1	0	0	0	1	0	0	0	octet 6d = 88 <sub>H</sub>
ext	Other modem type	Fixed network user rate (FNUR)= 64.0 Kbps						

The fields are set to the following values (according to 3G TS 24.008):

- Radio channel requirement: Full Rate channel
- Coding standard: GSM standardized coding
- Transfer mode: Circuit mode
- Information Transfer Capability: Unrestricted Digital
- Compression: data compression not possible
- Structure: Unstructured

- Duplex mode: full duplex
- Configuration: point-to-point
- Establishment: demand
- Rate adaptation: Other Rate Adaptation
- Other Rate adaptation: H.223 and H.245
- Signalling access protocol: I.440/I.450
- User info. layer 1 protocol: default layer 1 protocol
- Synchronous/asynchronous: Synchronous
- Number of stop bits: stop bit (value used in case of synchronous mode)
- Negotiation: in-band negotiation not possible
- Number of data bits: 8 bits (value used in case of bit oriented protocol)
- User rate: 9.6kbit/s
- Intermediate rate: 16kbit/s
- NIC on Tx: not required to send data with NIC
- NIC on Rx: not accepted to receive data with NIC
- Parity: none
- Connection Element: Transparent
- Modem Type: none
- Other Modem Type: no other modem type specified in this field
- Fixed Network User Rate: 64 Kbps bit transparent

Priority at VMSC has been observed in certain operators but not in all. This has similarities to Use Case 1, although UC1 does not request the call to be routed through an external transit network.



- (1) An originating VT call is initiated by a PLMNA sub to another PLMNA sub (Originating MSC and GMSC are likely to be the same equipment)
- (2) SRI message sent to the HLR has the correct information and the HLR is able to assess the right GSM Bearer Capability.
- (3) The GSM BC is transmitted to the VMSC when asking for a roaming number. The MSC gets the VT information.  
The Roaming number is sent back to the GMSC through MAP messages and the GMSC routes the call to the VMSC.
- (4) In this case, the PLMN has no transit network so the call is routed through a national carrier. The carrier does not transport correctly the VT parameters.
- (5) The ISUP parameters received by the VMSC in the incoming call are not accurate and VT cannot be assessed from them.

**Note:** that this case can be extended to roaming out case where it becomes likely to happen. If priority is given to the ISUP parameters and if a PLMN BC (different from VT) can be assessed, the VT call will fail.

If priority is given to the MAP parameters then the call will be successful.

**Recommendation:** VMSC prioritises the MAP information over the ISUP information received via the transit network.

### **3.4.4 Billing Aspects**

From product management perspective it is required to have a possibility to bill BS30 calls different from other basic services like speech (TS11), fax (TS62) or asynchronous data (BS26).

This can be implemented for straightforward MOCs and MTCs because the charging records in MSC contain a basic service field.

### **Bearer Service Code**

Only the values for 'General - data CDA' (Asynchronous) and 'General - data CDS' (Synchronous) are output for a UMTS call. This is an existing GSM field and does not indicate data rates higher than 9.6Kbit/s. For this reason it will be used in conjunction with the Fixed Network User Rate (FNUR) field to identify specific data rates. If the BSC value indicates a UMTS call, the FNUR field will be inspected.

### **FNUR Requested**

This field covers a wide range of data rates, but so far only 9.6 Kbps and 64 Kbps have been allocated a tariff in the billing chain. Any other data rates appearing in the FNUR field will result in the CDR being placed by the Mediator in a separate file for editing.

### **Multimedia Call**

Should it become necessary to differentiate between CSD64 and VT, this field will enable discrimination. It is a null field type and is only populated only in the case of a VT call.

## **3.5 RAN**

### **3.5.1 Radio Access Bearers required for video-telephony**

3GPP specifications specify in TS34.108 one RAB configuration to be used for conformance testing (section 6.10.2.2.13).

Conversational / unknown / UL:64 DL:64 kbps / CS RAB + UL:3.4 DL:3.4 kbps SRBs for DCCH

However, this RAB can be implemented with two different TTI values: 20 ms or 40 ms.

Although, R99 terminals are supposed to support both TTI values, experience shows that early terminals have been tested mainly against the 20 ms configuration. Therefore, the 20 ms configuration is recommended to be used by the RAN in order to limit the risk due to incorrect implementation by early UEs.

### **3.5.2 Radio Access Bearers combinations required**

Even if the user interface will often prevent the user from using a Packet Switched (PS) application during a video call, the terminal shall be able to receive or initiate a video call also when a background PS connection (PDP context with a PS I/B RAB) exists. Since a background PS connection can exist on the terminal transparently to the user (e.g. terminal receiving an MMS), most terminals will support the concurrent existence of a video call and a PS connection. Hence, when a video call is received or initiated whilst a PS connection exists, the terminal will attempt to set up this call normally.

The RAN shall be able to set up a video call to a terminal that already has a PS I/B RAB. Otherwise, there is a risk that the user experience will be affected if video calls fails to be initiated or received when a PS connection is up, especially if no application is alive on the terminal.

Several such RAB combinations (video call + background PS) have been defined for this purpose in 3GPP specifications (TS34.108 and TR25.993). The configuration most widely supported and tested by early terminals is the one specified in section 6.10.2.2.51a of TS34.108.

Conversational / unknown / UL:64 DL:64 kbps / CS RAB + Interactive or background / UL:8 DL:8 kbps / PS RAB + UL:3.4 DL:3.4 kbps SRBs for DCCH

Given that most early terminals are of uplink class 64 Kbps and will only do a limited usage of the PS I/B RAB during a video call, this RAB combination shall be sufficient to guarantee good user experience (i.e. allow the user to make or receive a video-call at any time) while limiting interoperability issues.

Nevertheless, the RAN shall be able to set up this RAB combination in all cases, i.e.:

- When a video call RAB request is received for a terminal that already has a PS RAB in FACH or PCH state.
- When a video call RAB request is received for a terminal that already has a PS RAB in DCH state.
- When a PS RAB request is received for a terminal that already has a video call RAB.

In the case where the RAN does not support any RAB combinations allowing the terminal to simultaneously have a video call and a PS connection, the RAN shall be able to pre-empt the existing PS background RAB in order to set up the video call.

### **3.5.3 Block Error Rate (BLER) Targets**

The BLER target is only one optimisation parameter of optimisation among several. The BLER target is operator-dependent and will be specific to each one. Experience shows that a total end-to-end BLER budget of 1.2% offers acceptable video quality.

## **3.6 KPI**

We recommend that KPIs based on the ETSI specifications referenced below should be introduced into each network. The values for each KPI should be defined by the individual network operators (between technology and marketing) to measure Video Telephony.

KPIs are based on ETSI Spec (ETSI/STQ Mobile (04) 03TD09, ETSI number TS 102 250-2, CR number 250-2 009.

### 3.6.1 VT Service Non-Accessibility [%]

**Abstract definition:**

Probability that the end-customer can't access the service when requested while it is offered by network indication on the mobile equipment.

Remark(s): A successful "Service Access" is when the A-party hears the alerting or busy tone after the send button is pushed.

**Computation**

Abstract formula:

$$VTServiceNon-Accessibility[\%] = \frac{\text{Number of Unsuccessful Video Telephony Call Access Attempts}}{\text{Number of All Video Telephony Call Access Attempts}} * 100\%$$

**Trigger Points:**

Event (from equation)	Trigger Point (from customer's point of view)
Video Telephony call access attempt	Push Send button
Unsuccessful Video Telephony call access attempt	Not Alerting or busy tone isn't heard by the A-party coming from B-party

### 3.6.2 VT Service Access Time [s]

**Abstract definition:**

The time between sending of complete address information and the receipt of VT call set-up notification.

Remark(s): This parameter is not calculated unless the video telephony call access attempt is successful.

**Computation**

Abstract formula:

$$VT\ Service\ Access\ Time\ [seconds] = t_{Connection\ Established} - t_{PushSendButton}$$

**Trigger Points:**

Event (from equation)	Trigger Point (from customer's point of view)
Video Telephony call access attempt	Push Send button
Connection established (Successful Video Telephony call attempt)	Alerting or busy tone isn't heard by the A-party coming from B-party

### 3.6.3 Audio/Video Set-up Failure Ratio [%]

**Abstract definition**

Probability of audio/video set-up failure after service access. The audio/video set-up is successful if audio and video output is performed at both sides.

Remark(s): This parameter reports a failure if the end-trigger isn't reached at both sides.

This parameter is not calculated unless the VT service access attempt is successful.  
 This parameter depends on the mobile used and on the multimedia protocol stack implemented (e.g. answer fast feature).

**Computation**

Abstract formula:

$$\text{VT Audio/Video Setup Failure Ratio [\%]} = \frac{\text{Number of Audio/Video Setup Failures}}{\text{Number of All Accepted Calls at MT side}} * 100$$

**Trigger Points:**

Event (from equation)	Trigger Point (from customer's point of view)
Accepted call at MT side	Pressing the accept key at the MT side to accept the incoming call
Audio/Video Set-up Failure	Not start of the audio and video output at both sides

**3.6.4 VT Audio/Video Set-up Time [s]**

**Abstract definition**

The elapsed time from accepting the call at the MT side until audio and video output starts at both sides.

Remark(s):

This parameter should report the worse time of both sides.  
 This parameter is not calculated unless the VT audio/video set-up attempt is successful.  
 This parameter depends on the mobile used and on the multimedia protocol stack implemented (e.g. answer fast feature).

**Computation**

Abstract formula:

$$\text{VT Audio/Video Setup Time [s]} = t_{\text{Audio / Videostart}} - t_{\text{MTacceptca ll}}$$

**Trigger Points:**

Event (from equation)	Trigger Point (from customer's point of view)
Accepted call at MT side	Pressing the accept key at the MT side to accept the incoming call
Audio/Video Start	Start of the audio and video output at both sides

**3.6.5 VT Cut-off Call Ratio [%]**

**Abstract definition**

Probability that a successful service access is ended by a cause other than the intentional termination of the user (calling or called party).

Remark(s): This parameter is not calculated unless the VT service access attempt is successful. A VT call is considered dropped if both audio and video are not established within the audio/video set-up timeout or if either the audio, the video or both are lost for at least 10 seconds.

**Computation**

Abstract formula:

$$\text{VT Cut - off Call Ratio } [\%] = \frac{\text{Number of VT Dropped Calls}}{\text{All successful VT Call Access Attempt}} * 100$$

**Trigger Points:**

Event (from equation)	Trigger Point (from customer's point of view)
Dropped Call	Lost of video and/or audio without any intention by A- or B-party
Call Access Attempt	Alerting or busy tone heard by the A-party

**3.6.6 VT Speech Quality on call basis [MOS-LQO]**

**Abstract definition**

Indicator representing the quantification of the end-to-end speech transmission quality of the Video Telephony Service. This parameter computes the speech quality on the basis of completed calls.

**Computation**

The validation of the end-to-end quality is made using the MOS<sub>LQO</sub> scale. This scale describes the opinion of customers with voice transmission and their perceived ranking of service trouble (noise, robot voice, echo, dropouts etc). The speech quality measurement is taken per call. An aggregation should be made on one value for speech quality per call. Reference: ITU-T P.862 (PESQ Algorithm)[1] in conjunction with ITU-T Rec. P.862.1.

Abstract formula:

$$\begin{matrix} \text{SpQ} & - & \text{C(received} & & \text{A - side)} & = & \text{f(MOS}_{\text{LQO}} & ) \\ \text{SpQ} & - & \text{C(received} & & \text{B - side)} & = & \text{f(MOS}_{\text{LQO}} & ) \end{matrix}$$

Optionally it might be useful to aggregate both speech quality values into one. In this case the worst of both shall be used. This aggregated speech quality value shall be called SpQ (min).

**Trigger points:**

Begin: Start of the audio and video output at both sides

End: Release of connection

NOTE(s): The acoustic behaviour of terminals is not part of this speech quality measurement.

For wideband (7 kHz) applications no standardized algorithm is available yet.

**3.6.7 VT Speech Quality on sample basis [MOS-LQO]**

**Abstract definition**

Indicator representing the quantification of the end-to-end speech transmission quality of the Video Telephony Service. This parameter computes the speech quality on a sample basis.

**Computation**

The validation of the end-to-end quality is made using the MOS scale. This scale describes the opinion of customers with voice transmission and it's troubles (noise, robot voice, echo, dropouts etc). The speech quality measurement is taken per sample. An aggregation for measurement campaigns or parts of it should be made on speech sample basis.

Reference: ITU-T P.862 (PESQ Algorithm)[1] in conjunction with ITU-T Rec. P.862.1.

Abstract formula:

$SpQ_{A}$	- S(received	A - side)	= f(MOS_	$LQO_{A}$	)
$SpQ_{B}$	- S(received	B - side)	= f(MOS_	$LQO_{B}$	)

Optionally it might be useful to aggregate both speech quality values into one. In this case the worst of both shall be used. This aggregated speech quality value shall be called SpQ (min).

**Trigger points:**

Begin: Start of the audio and video output at both sides

End: Release of connection

NOTE(s): The acoustic behaviour of terminals is not part of this speech quality measurement.

For wideband (7 kHz) applications no standardized algorithm is available yet.

**3.6.8 VT Video Quality [ ]**

**Abstract definition**

End-to-end quality of the video signal as perceived by the end user during a VT call.

Remark(s): This parameter is not calculated unless the VT audio/video set-up attempt is successful.

**Computation**

Not available

Abstract formula:

Not available

**Trigger Points:**

Event (from equation)	Trigger Point (from customer's point of view)
Successful Audio/Video Set-up Attempt	Start of the audio and video output at both sides
Release of connection	Release of connection

**3.6.9 VT End-To-End Mean One-Way Transmission Time [s]**

**Abstract definition**

Delay time from input of the signal at MS (MO/MT) (mic / cam) to output of the signal at MS (MT/MO) (loudspeaker / display).

Remark(s): This parameter is not calculated unless the VT audio/video set-up attempt is successful.

**Computation**

Abstract formula:

Time from input of the signal at MS (MO/MT) to output at MS (MT/MO)

Aggregation Algorithm: ((Transmission Time MO ->MT) + (Transmission Time MT->MO)) / 2

Remark(s): This parameter is not calculated unless the VT audio/video set-up attempt is successful.

**Trigger Points:**

Event (from equation)	Trigger Point (from customer's point of view)
	Signal in MS (MO/MT) side
	The same signal in MS (MT/MO) side

**3.6.10 VT Audio/Video Synchronisation [%]**

**Abstract definition**

Percentage of times that the time differences of the audio and video signal at the user side exceeds a predefined threshold.

Remark(s):

This parameter is not calculated unless the VT audio/video set-up attempt is successful. Only if audio and video use different bearers this indicator would reflect the behaviour of the network and the mobiles.

**Computation**

Not available

Abstract formula:

Not available

**Trigger Points:**

Event (from equation)	Trigger Point (from customer's point of view)
Successful Audio/Video Set-up Attempt	Start of the audio and video output at both sides
Release of connection	Release of connection

Other Intra-Network issues

### **In-band Commands and indications**

The 3G-324M standards define a number of commands and indications that can be used during Video Telephony calls, these are well defined and do not elicit response messages from the other terminal but they do force an action (command) or provide information (indication). This behaviour should not introduce interoperability issues if they are correctly identified and declared as part of the TCS as described previously. The most common commands/indications that are observed to cause issues related to interoperability are:

### **Fast Update**

videoFastUpdatePicture commands the video encoder to enter the fast-update mode at its earliest opportunity. This is typically used when the decoder is having difficulty decoding the incoming bit-stream and error resilience tools are not sufficient to recover from a corrupted sequence of frames. In the wireless environment where the round trip delays can be significant, the use of Fast Update is not the most efficient method of recovery but should be understood by all devices. Instances of no reaction to Fast Update have been seen in terminals, which have a negative impact on the end user visual experience.

### **DTMF & User Input Indication (UII)**

This is not typically required in mobile to mobile usage scenario's but a method by which terminal keypad presses can be conveyed to the other end-point is important for mobile to network use scenario's e.g. VideoMail servers, Video Content Portals, Multimedia Gateways etc. The H.245 standard specifies a number of methods to achieve DTMF to UserInput Indication. Interoperability issues can occur if there is a mismatch in UII capabilities or error in the declared capability.

It is recommended that terminals use the alphanumeric indication to convey DTMF user input unless the other terminal has indicated another preference. For the purposes of interoperability, it's also recommended that the terminal declares its support for UserInputIndication (via userInputSupportIndication) in the TCS and that the IA5String character set is supported.

### **Temporal Spatial Trade-off**

The videoTemporalSpatialTradeOff command instructs the far-end video encoder to change its trade-off between temporal and spatial resolution.

A value of 0 commands a high spatial resolution and a value of 31 commands a high frame rate.

A value should be in the range of 0 to 31, and a value of 0 means the highest spatial resolution and a value of 31 means the highest frame rate.

It is recommended that terminals support the TemporalSpatialTradeOff capability as it presents the ability for the encoder to vary its trade-off between "quality" and "frame-rate" which is a desirable function for some usage scenarios.

### **Audio-Video Synchronisation or Lip-Sync**

The lack of synchronisation between media streams is an issue in Video Telephony, its inherently a terminal specific issue and is heavily dependent on the device architecture and CODEC implementation. Typically the video capture, encoding and multiplexing takes longer than the same steps for audio so synchronisation loss usually manifests itself as the late arrival of video.

The issue is compounded by the variation in the synchronisation achieved by devices, which can lead to very undesirable effects in interoperability scenarios. The recommended solution

is for terminal vendors to implement their solution so that the video delay is minimised and therefore the impact is reduced without any need for additional steps to improve the user experience.

The 3G-324M specifications have a H.223 Skew Indication that can be used to indicate to the far-end terminal the average amount of time skew between two logical channels in milliseconds. However, it must be understood that this indication is only of value where the skew value is relatively low. If it is used to indicate a large loss of synchronisation and the other party reacts by delaying the audio, the user can end up with an intolerable end-to-end delay, which gives the impression of a half-duplex call. This is widely considered to provide a worse user experience than an “out of sync” call.

Therefore, it is recommended that Skew Indication is not used to compensate for poor terminal implementations but is used by terminals wishing to provide an indicative value of their skew in milliseconds. It is also recommended that receiving terminals can interpret the skew indication message.

## 4 INTER-NETWORK IMPLEMENTATION GUIDELINES FOR CS VT CALLING BETWEEN 3G NETWORKS

### 4.1 Purpose and scope

The purpose of this section is to provide implementation guidelines to 3G operators for inter operator (national or international) 64 Kbps Video Telephony calls.

The scope of this section is limited to 3GPP R99 implementations of CS VT calls. This includes those requirements that a VT partner of a mobile operator needs to meet for interconnection. Also in scope are:

1. Seamless access to Videomail service when roaming
2. Billing of inter-network VT calls.

### 4.2 Reference documents

1. ITU Q.767 - Application of the ISDN User Part of CCITT signalling system No. 7 for international ISDN interconnections (1991).
2. ITU-T Q.850 – Usage of cause and location in the Digital Subscriber Signalling System N°.1 and the Signalling System N°.7 ISDN User Part
3. 3GPP TS 24.008 V3.19.0 (2004-06)
4. 3GPP TS 22.002 V3.6.0 (2001-03)
5. 3GPP TS 27.001 V3.15.0 (2004-06)
6. 3GPP TS 29.007 V3.15.0 (2004-06)
7. 3GPP TS 23.008 V3.8.0 (2003-06)
8. 3GPP TS 23.018 V3.12.0 (2003-03)
9. 3GPP TR 29.007 Change Request 089 v5.7.0 - Backward signalling of service information between VMSC and GMSC for MTC
10. ITU-T Q.50 - Signalling between Circuit Multiplication Equipment (CME) and International Switching Centres (ISC)
11. ITU-T Q.115.1

### 4.3 Requirements towards VT partner

This section identifies the different key requirements that the VT partners shall fulfil for inter operator VT calls.

#### 4.3.1 Handset requirement

LLC IE is mandatory for an outgoing VT call in UDI mode (3GPP 27.001 section B.2.1.1). The VT partner mobile operator shall ensure that Handsets / Mobile Equipment do conform to this specification to guarantee that LLC is present in IAM message/ATP field. The aim is to overcome some limitations known with ISUPv1 (Q.767).

#### 4.3.2 Outgoing traffic from 3G partner

VT partner shall guarantee that Video Telephony bids (IAM messages) originated on its own PLMN and presented to the national or international outgoing interconnections will be formatted like this:

1. Transmission Medium Requirement (TMR): '64 kbit/s unrestricted'.

2. User Service Information (USI): this field is the same as the ISDN Bearer capability information element from ITU-T Q.931. This information element shall be coded as follows:

8	7	6	5	4	3	2	1	Octet	
0	Bearer capability information element identifier						0	0	1 = 04
Length of the bearer capability contents								2 = 03	
ext. 1	Coding standard ITU		Information transfer capability (ITC) Unrestricted digital information					3 = 88	
ext. 1	Transfer mode Circuit Mode		Information transfer rate 64 kbit/s					4 = 90	
ext. 1	Layer 1 ident. 0 1		User information layer 1 protocol (ULIP) H.223 & H.245					5 = A6	

Table 1: coding of ISDN Bearer Capability

Access Transport Parameter (ATP) (this field allows the transport of several Q.931 Information Elements (IE) such as Low Layer Compatibility (LLC IE)):

The Low layer compatibility information element (LLC IE) shall be present inside ATP and coded as follows:

8	7	6	5	4	3	2	1	Octet	
0	Low layer compatibility information element identifier						1	0	1=7C
Length of the low layer compatibility contents								2=03	
ext. 1	Coding standard ITU-T		Information transfer capability (ITC) unrestricted digital information					3=88	
ext. 1	Transfer mode circuit mode		Information transfer rate 64 kbit/s					4=90	
ext. 1	Layer 1 ident. 0 1		User information layer 1 protocol (ULIP) H.223 and H.245					5=A6	

Table 2: coding of ISDN Low Layer Compatibility

ISDN User Part indicator in the IAM field (from ITU-T Q.767): "ISDN User Part used all the way".

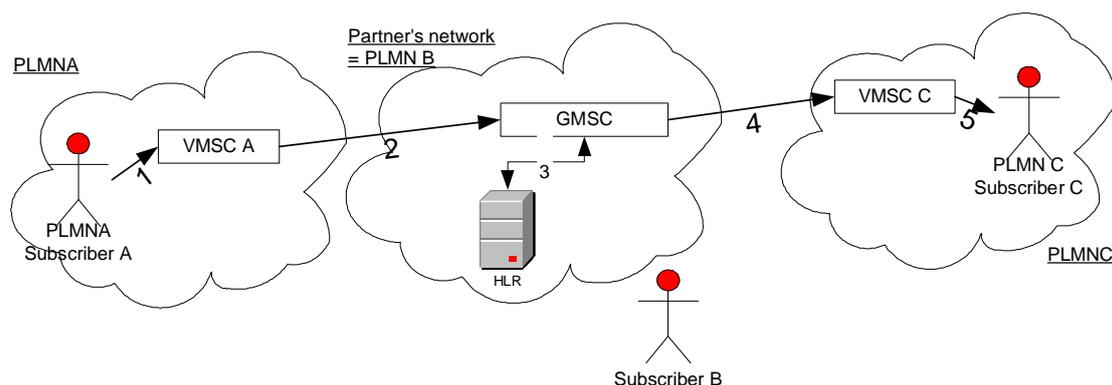
### 4.3.3 External traffic rerouted from VT partner

These are calls, which are not originated on the 3G partner's PLMN but are coming from an external network and are then rerouted to your own PLMN:

- Call to a partner's subscriber then diverted to your own PLMN (see Use Case 10):

The 3G partners shall guarantee that the ISUP IAM message of the call-diverted leg will contain Bearer Capability in USI and LLC in ATP if they were received in the signalling of the incoming call.

The drawing below shows the case where the call is diverted at the GMSC.



(2) incoming call.

The ISUP IAM message contains the VT information (ISDN BC and LLC in USI and ATP ISUP parameters).

(4) diverted call.

The GMSC shall insert the VT information received from the originating network. ISUP IAM message of the diverted call shall contain the VT information.

**Figure 2: VT call diversion between networks**

#### 4.3.4 Error cases

##### 4.3.4.1 Call failure management

In case that the VT call cannot be successful then a Release message (with a relevant cause value) shall be generated by the 3G partner's PLMN backward to the originating PLMN. Currently a Permanent Reference Document on error release code is under negotiation between IREG members and should be agreed in Q2 2005. This PRD will be the reference document for implementation guidelines. See IREG document IR.69.

It will specify the recommended error to use for various scenarios including:

1. the called 3G partner's subscriber has no subscription for VT calls,
2. the called subscriber does have a subscription for VT calls but is not under coverage of a 3G MSC,
3. the called subscriber does have a subscription for VT calls but is currently using a non VT handset,
4. Variants of the above cases with call forwarding.

4.3.4.2 Method for aligning information in Bearer Capability and Low Layer Compatibility

It is recommended that the 3G partners (originating and terminating 3G operator) prepare error recovery functionality as follows.

This functionality is intended to make international Video Telephony robust during early stages of implementation, and all involving parties, such as 3G partners (originating and terminating 3G operators) and international carriers, should implement the requirements stated in this document.

1. Incoming traffic to the 3G partner (terminating 3G operator)

(Interconnection from international/national carrier)

- o In the case that within the IAM message the UIL1P = 'H.223 & H.245' parameter is missing in the USI field but it is presented in the ATP/LLC field, the 3G partner (terminating 3G operator) may determine that the call is a video call.
- o If above, 3G partner (terminating 3G operator) may fill in the UIL1P = 'H.223 & H.245' parameter in the USI field /ISDN BC.
- o Or alternatively, inter-work to UMTS BC, setting the UIL1P = 'H.223 & H.245' parameter in the 5a octet.

2. Outgoing traffic from the 3G partner (originating 3G operator)

(Receiving SET-UP from handsets)

- o In the case that, within the IAM message from the handsets, the LLC is missing but it is presented in the BC, the 3G partner (originating 3G operator) may determine that the call is a video call.

If above, 3G partner (originating 3G operator) may fill the LLC in the ATP field of the IAM message as stated in section 4.4.2 of this document.

Compatibility

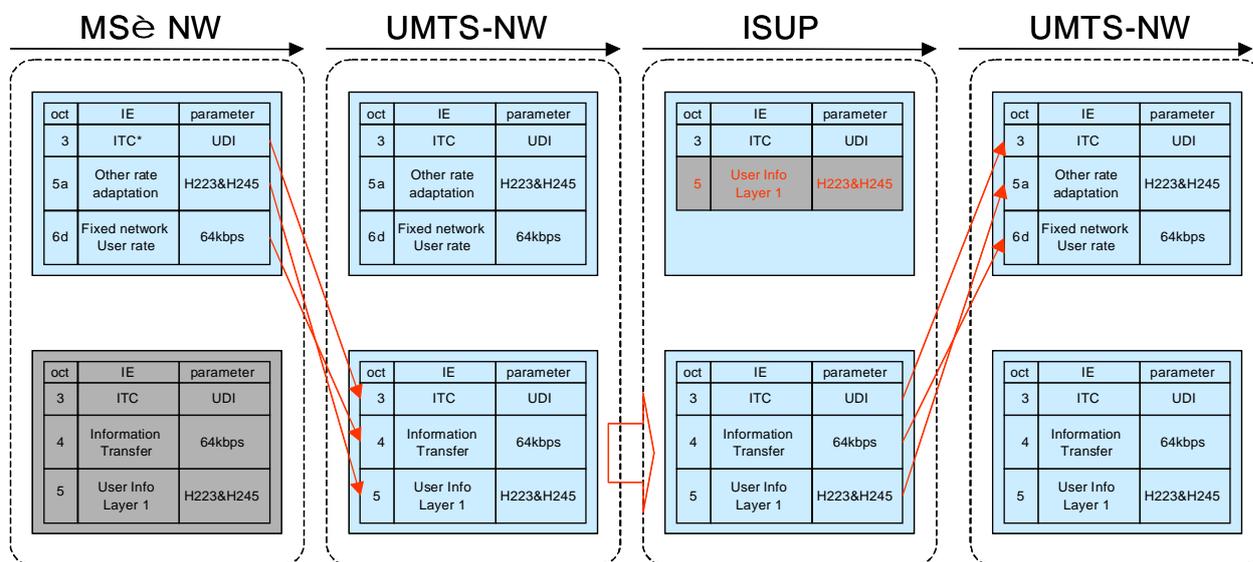


Figure 3: Method for aligning information in Bearer Capability and Low Layer

4.3.5 Requirements towards Transit Carriers

4.3.5.1 Responsibilities

It is the responsibility of the originating mobile network operator when entering into commercial agreements with carriers for the conveyance of VT traffic, to fulfil the requirements presented hereafter.

It is the responsibility a carrier when acting as a transit carrier for the conveyance of VT traffic on behalf of an originating mobile network operator to fulfil the requirements presented hereafter. This includes ensuring that these requirements are also met by any other carrier used to convey VT traffic between the originating mobile network operators and terminating mobile network operator.

It is the responsibility of the terminating mobile network operator to ensure that its VT termination service fulfils the requirements presented hereafter.

#### *4.3.5.2 Traffic channel requirement*

A VT call is an ISDN one therefore 64 Kbps clear channels are compulsory for Video Telephony. The transmission medium used will comply with the Transmission Medium Requirement = '64 kbit/s unrestricted'.

It shall be guaranteed by the carriers from the originating mobile network and the terminating one.

#### **Usage of Digital Circuit Multiplication Equipment (DCME) by carriers:**

In particular, if carriers use DCME, switches and DCME shall support 64 Kbps unrestricted connection type negotiation as described in 10. However, even if procedure described in 10 is supported, usage of DCME for carry VT calls may result in VT failure because of 64 Kbps channel being unavailable.

#### **Usage of Echo Control Devices (ECDs) by carriers:**

If ECD is used, according to 11, no ECD shall be inserted if the bearer capability is 64 Kbps unrestricted. If the ECD is permanently associated, these ECDs have to be disabled and provide bit transparency.

#### *4.3.5.3 Signalling requirement*

ISUPv2 should be used to guarantee VT calls because it transports transparently the signalling information necessary for VT.

1. TMR,
2. USI (which transports the ISDN Bearer Capability),
3. ATP (which transports the ISDN Low Layer Capability).

If ISUPv2 can not be guaranteed, Transit carriers shall guarantee that Video Telephony bids (IAM messages with Transmission Medium Requirement set to '64 kbit/s unrestricted', User Service Information with: Transfer mode = 'circuit mode', Information transfer rate = '64 kbit/s', User information layer 1 protocol = 'H.223 & H.245') will be handled like this:

1. The IAM fields hereafter will be passed unchanged to the next carrier or to the final destination:
2. Transmission Medium Requirement (TMR),
3. User Service Information (USI),
4. Access Transport Parameter (ATP).

**Note:** Blue Book ISUP v1 (Q.767) doesn't guarantee the coding and handling of the value 'H.223 & H.245' for the User information layer 1 protocol information Element of USI. The carrier shall be able to route and terminate correctly a VT call. To ensure the routing it is necessary to set the ISDN User Part and the ISDN User Part Preference indicators as follows.

The ISDN-UP and ISDN-UP preference indicator are set according to the bearer service, Teleservice and supplementary service(s) requested.

The ISDN user part indicator possible values are:

- |   |                                      |
|---|--------------------------------------|
| 0 | ISDN user part not used all the way. |
| 1 | ISDN user part used all the way.     |

The expected behaviour is to have for video calls the ISDN User Part Indicator set to “used all the way”

The ISDN user part preference indicator possible values are:

- |     |  |
|-----|--|
| 0 0 | ISDN user part preferred all the way.    |
| 0 1 | ISDN user part not required all the way. |
| 1 0 | ISDN user part required all the way.     |
| 1 1 | spare.                                   |

The expected behaviour is to have for video calls the ISDN User Part Preference Indicator set to "ISUP required all the way".

For any other value the carrier network could route the VT call to an encoded channel (e.g. VoIP trunks) blocking any H.245 signalling with the risk to drop the call.

#### *4.3.5.4 Billing requirement*

Mobile network operators may wish to charge VT termination rates differentially from, for example, voice termination. Any carrier conveying VT traffic shall be able to identify and bill VT traffic separately. This requirement also enables carriers to charge different transit costs for the conveyance of VT traffic.

See 4.6.1 for issue and implementation proposal.

#### *4.3.5.5 Routing requirement*

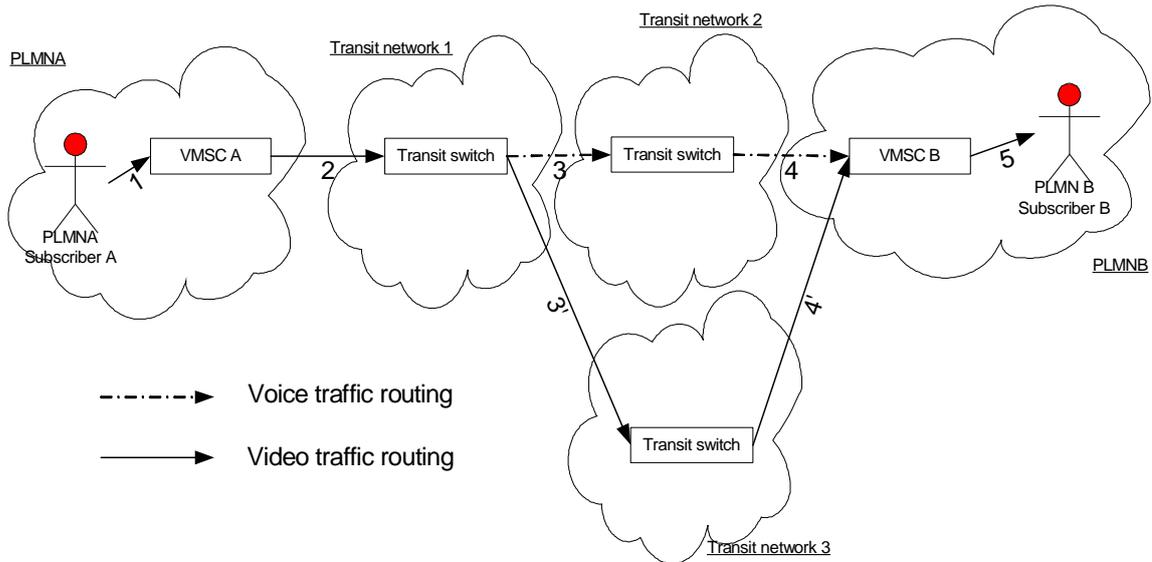
Because some carriers do not guarantee proper carrying of VT calls, it could be necessary to route differently the VT traffic for the same destination.

The carrier shall be able to route differently the UDI traffic to the same destination based on TMR values.

- TMR = “Speech”,
- TMR = “3.1kHz audio”,
- TMR = “64 Kbit/s Unrestricted Digital Information”.

If TMR = “64 kbit/s Unrestricted Digital Information”, then the carrier shall be able to route the call through a specific route supporting inter network VT calls.

Furthermore, there is an opportunity for carriers to differently route the VT traffic from the other UDI one based on UIL1P parameter of USI.



Transit network 2 does not support proper VT transit but is cheaper than Transit network 3 which does support VT.  
 Transit network 1 can still route voice calls to PLMN B through Transit network 2 while routing the VT traffic to the same destination through Transit network 3.

**Figure 4: VT transit routing requirement**

#### 4.4 Filtering recommendation

There is the possibility that Video telephony customers could experience video SPAM, video advertisements or other nuisance calls that originate from other networks. There are a number of ways to deal with this issue. A combination of these methods would be most effective:

1. Bilateral agreements with the originating network to suspend subscribers that generate these calls.
2. Development and Deployment of anti-spamming technology within the terminating network.
3. Commercial agreements with the transit carrier to implement video telephony anti spamming measures. These include blocking incoming VT calls from particular originating networks.

#### 4.5 Seamless access to Videomail service when roaming

Operators are currently considering different methods for video-mail deposit, access and retrieval. There are issues with synchronisation on deposit that affect cross-network video-mail retrieval.

The problem that 3G network operators are experiencing at the moment concerns the different audio/video skew times given by different handsets when depositing Videomail. So far, operators have tried to characterise the skew from different handsets, and have compiled tables of results that can be used to provide the correct skew when replaying the Videomail. However, this is only possible for handsets that operators can correctly identify. With more players now entering the 3G markets around the world, there will be many new handsets that we have not characterised. Also, the standard method for identifying VT handsets is to use the H.245 Vendor ID string. This Vendor ID is still not supported on all handsets (for example the Sony Ericsson Z1010).

To make matters worse, some handsets vary their skew instantaneously, depending upon their internal architecture (e.g. single processor or multi-processor), Operating System, and other tasks currently running, such as performing a handover between cells or receiving an SMS during the VT call.

If the issue can be easily handled by operator with their own vendors, there are issues reported when the deposit is done by a subscriber from another network and no skew value is available for his handset or vendor ID is not implemented. As a consequence, when your customer retrieves the video mail, the video mail equipment can't correct the skew and the customer will experience unsynchronised message leading to a very bad service quality.

##### **Requirements:**

Handsets shall support the H.245 Vendor ID string.

##### **Recommendations:**

GSMA and/or other bodies could set up a register of typical audio/video skew times that all operators would contribute to (and have access to). This would be based on a standard test that we could define.

GSMA Devices Group ensures that all future VT handsets have audio/video skew set to a certain value - or at least within certain bounds.

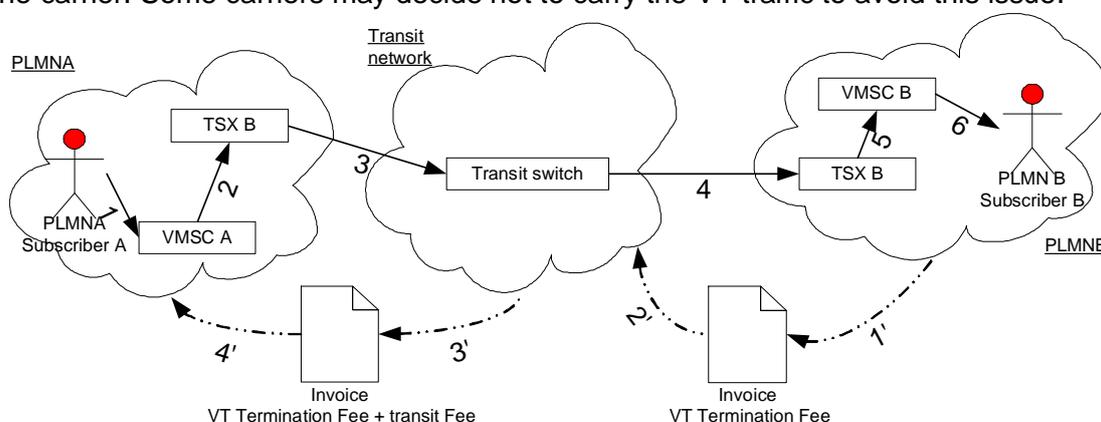
Architectures of multi-threaded handset operating systems are defined such that VT coding of the audio and video streams is NOT affected by other parallel tasks. From a longer-term research perspective, the audio and video codecs should continually send (along with their usual data) a real-time instantaneous measurement of audio/video skew or send a synchronization pattern. This could be used by any playback platform to adjust skew on the fly.

## 4.6 VT transit traffic billing

### 4.6.1 Normal case

It is likely that Mobile Operators will charge different Video Telephony termination fee from other ISDN calls. As cascade billing is applied in most cases, it is compulsory for transit carriers to be able to differentiate VT and other ISDN traffic so they can charge the originating network or another transit carrier.

If the carrier is not able to differentiate traffic, this could lead to an important loss of revenue for the carrier. Some carriers may decide not to carry the VT traffic to avoid this issue.



(1') PLMN B charges the transit network for a VT terminated call. The VT termination fee is higher than other ISDN ones (data or voice).

(3') The transit network should be able to charge PLMN A for VT transit call.

**Figure 5: cascade billing of inter network VT**

As a consequence, it is necessary to differentiate VT traffic in CDRs and in billing applications.

The CDR Bearer Service parameter shall be filled according to the content of ISDN parameters of the ISUP IAM message.

The implementation can be done thanks to the TMR (Transmission Medium Requirement) field which provides the following bearer:

- Bearer Service = "Speech"
- Bearer Service = "3.1kHz audio"
- Bearer Service = "64 Kbit/s Unrestricted Digital Information"

If TMR = "64 kbit/s unrestricted", an additional analysis is done to distinguish VT bearer from other ISDN traffic. In this case, if UIL1P ( User Information Layer 1 Protocol ) in USI (User Service Information) parameter equals 'H.223 & H.245' or 'H.221 & H.242', the Bearer Service field of the Transit CDR will be set to "64 Kbit/s Video Telephony", else it will remain as "64 kbits UDI".

In case UIL1P is absent from USI parameter or USI is absent, the same check of UIL1P is performed on the LLC if present (Lower Layer Capabilities) in the Access Transport parameter.

**Note** that the method described above only differentiates between 64 Kbps UDI VT from other ISDN traffic. It does not allow the ability to differentially charge between 64 Kbps VT and sub-64 Kbps VT rates defined by 3GPP. It does not allow differentiation between 64 Kbps Transparent Mode (BTM), 64 Kbps Frame Tunnelling Mode (FTM), legacy 9.6 Kbps, and HSCSD.

#### **4.6.2 Billing issue due to service change or incomplete information**

Change Request 089 (9) defines the mechanisms to support service change by the called UE and how the information on used services for MTCs is transported back from the VMSC after negotiation with the UE, to the GMSC.

When receiving a Set-up message, the UE may negotiate parameters with the MSC according to the rules defined in 3GPP TS 27.001.

This negotiation takes place by means of the UE reflecting back to the MSC a complete bearer capability information element in the call confirmed message, with the relevant parameters changed. The VMSC may map the received PLMN BC into an ISDN BC. This ISDN BC can be transported together with possibly available LLC and HLC in the access transport parameter of the Answer message (ANM) according to ITU-T Q.763.

To match the actual specifications of the call, the GMSC as well as the Originating MSC and all the transit switches should be able to insert the negotiated information into the CDRs.

Currently, PLMNs and transit carriers are unable to guarantee support of service change especially when a change of the channel is requested (UDI to 3.1 kHz for example). However it should be possible to support service change from a 64 kbits CSD transparent channel to a VT call without any impact on the transmission medium requirement and therefore on the transit networks. 3GPP 27.001 already allows for the ability to negotiate 64k BTM to 64k VT. See table B.4f in 27.001. In this case, the bearer of transit network as well as the PLMN ones could support a VT call as H.324 is an inband protocol. If the technicalities are supported then the billing implications are complex and will require further investigations.

## 5 ROAMING IMPLEMENTATION GUIDELINES FOR CS VT

### 5.1 Purpose and scope

The principal purpose of this section is to establish the optimum method by which a VT call can be identified under this roaming scenario, in order to support appropriate charging for it and to support the correct call set-up. We assume that the appropriate bearers are already available to roaming subscribers

Most VT roaming issues are related to interconnect issues that are addressed in previous chapters of this document. The scope of this section is therefore limited to those VT requirements that are specific to roaming:

- Application Context management to support VT roaming.
- Billing requirements.
- VT call to a ported number when roaming out.

### 5.2 Reference documents

1. 3GPP TR 29.002
2. PRD TD.58 v1.0 – TAP3 Implementation Handbook
3. PRD TD.57 v3.10.1 - Transferred Account Procedure Data Record Format Specification Version Number 3

### 5.3 Application context management

This sub-section identifies which are the application context version to set to support VT in roaming conditions.

Application context are used to specify the context of a MAP operation used. For example, Insert Subscriber data can be used in two contexts:

1. When a subscriber registers on a new VLR and no subscription information is available at this VLR.
2. When subscription information has been modified at the HLR.

Through the different MAP releases, new versions of MAP operations have been specified, and the need for differentiating them arose.

When roaming in different PLMNs, subscription, authentication and other subscriber's related information need to be exchanged between the VPLMN and the HPLMN. For each VPLMN and each Application Context, the maximum version supported shall be specified.

### 5.4 Application context version

We recommend that in accordance with 3GPP standards, that MAPv3 R.99 is mandatory to support the exchange of VT subscription information from the inbound roamer's HLR to the VLR.

The VT information is carried into the basic service parameter (into the bearer service list) of the MAP Insert subscriber data message.

Value 1F ("general-dataCDS") is not specified in MAP releases before v3.

Therefore if MAPv2 or v1 is used to send the subscription data to the VLR, the subscriber will not be able to make or receive a VT call.

Two application contexts are using the MAP insert subscriber operation and shall be configured in version 3:

#### **NetworkLocUpContext**

NetworkLocUp is the context used for Update Location procedure and will set the MAP version of ISD and UpLoc messages when used in this context.

#### **SubscriberDataMngtContext**

SubscriberDataMngtContext is the context used when subscription data has been modified into the HLR by the subscriber's operator.

### **5.5 Potential inter action issues**

Before introduction of 3G roaming agreements, CAMEL information exchange with a given VPLMN could be easily inhibited by limiting the above AC to version 2 (where no CAMEL agreement exists).

However, when the 3G roaming case described below is encountered, this method can no longer be used:

- The 3G roaming partner network has implemented CAMEL with other operators but not with your network.
- The 3G roaming partner network does not inhibit CAMEL information exchange or triggering on its VLR.

In this case, if you don't implement a method yourself to inhibit the exchange of CAMEL information, then updating above AC to version 3 will automatically trigger sending of them when a CAMEL subscriber locates on the 3G roaming on the network.

As no CAMEL testing has been done with the 3G partner, it is likely that CAMEL services will not work.

**Recommendation:** When implementing VT roaming with a 3G partner, then CAMEL inhibition per VPLMN shall be implemented at the same time.

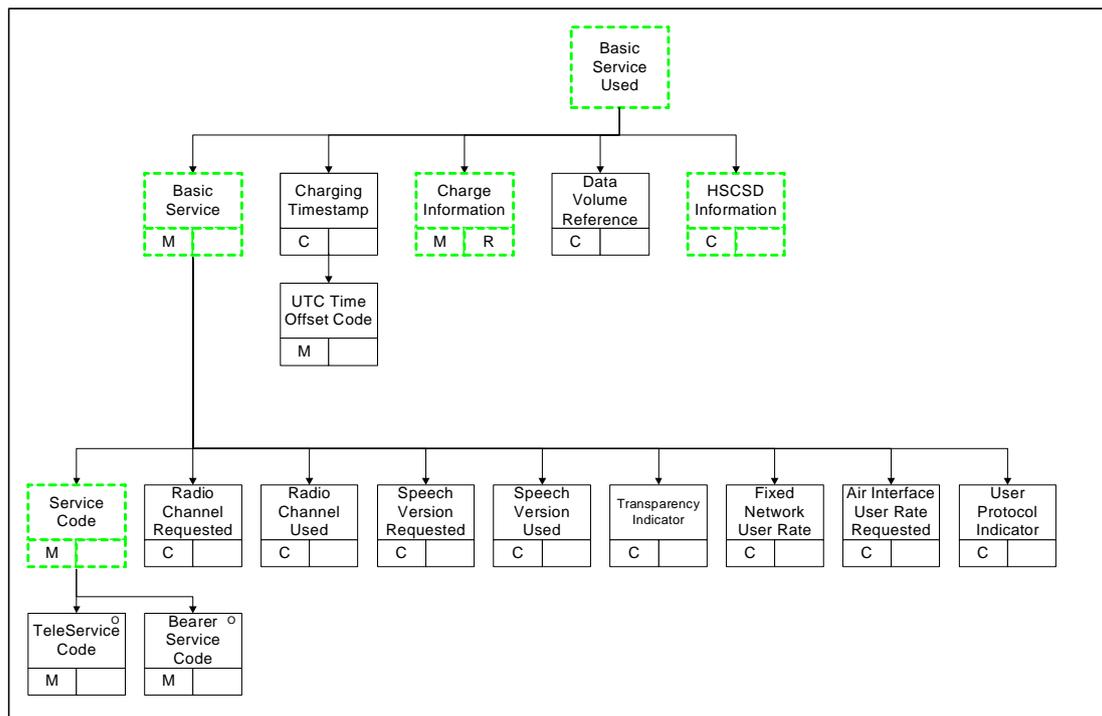
### **5.6 VT roaming billing**

#### **5.6.1 TAP version**

TAP v3.10 and upward support proper information for VT roaming billing.

#### **5.6.2 TAP element values**

The information is extracted from document 2. The information related to VT is inside the Basic Service Used element:



**Table 3: Basic Service Used Logical Structure from 2**

Fixed Network User Rate, Air Interface User rate Requested and User Protocol Indicator are new TAP 3.10 elements.

For a VT call (64 kbits CS, transparent & UDI), they shall be all filled as follow:

Parameter	Value for VT
Bearer Service Code <sup>1</sup>	GBS30 = 37 <sub>H</sub>
Transparency Indicator	Transparent = 00 <sub>H</sub>
Fixed Network User Rate	64 kbits/s = 08 <sub>H</sub>
User Protocol Indicator <sup>2</sup>	H.223 & H.245 = 04 <sub>H</sub>

**Table 4 : TAP Basic Service element values for 64 Kbits UDI VT**

## 5.7 VT call to a ported number when roaming out

### 5.7.1 Description of the issue

MNP has been now introduced in many countries around the world. When calls to a ported number come from outside the country, the call is always routed to the range owner network or to a national transit carrier able to determine the current subscription network.

When no VT inter network agreement exists between the number range owner and the VPLMN, then there is a risk that a call from an inbound roamer to a ported in subscriber of the same operator will fail. In this case, the VPLMN may route the call through an

<sup>1</sup> Some operators fill Bearer Service Code with Value 30<sub>H</sub>. This is not standard and shall be fixed as soon as possible.

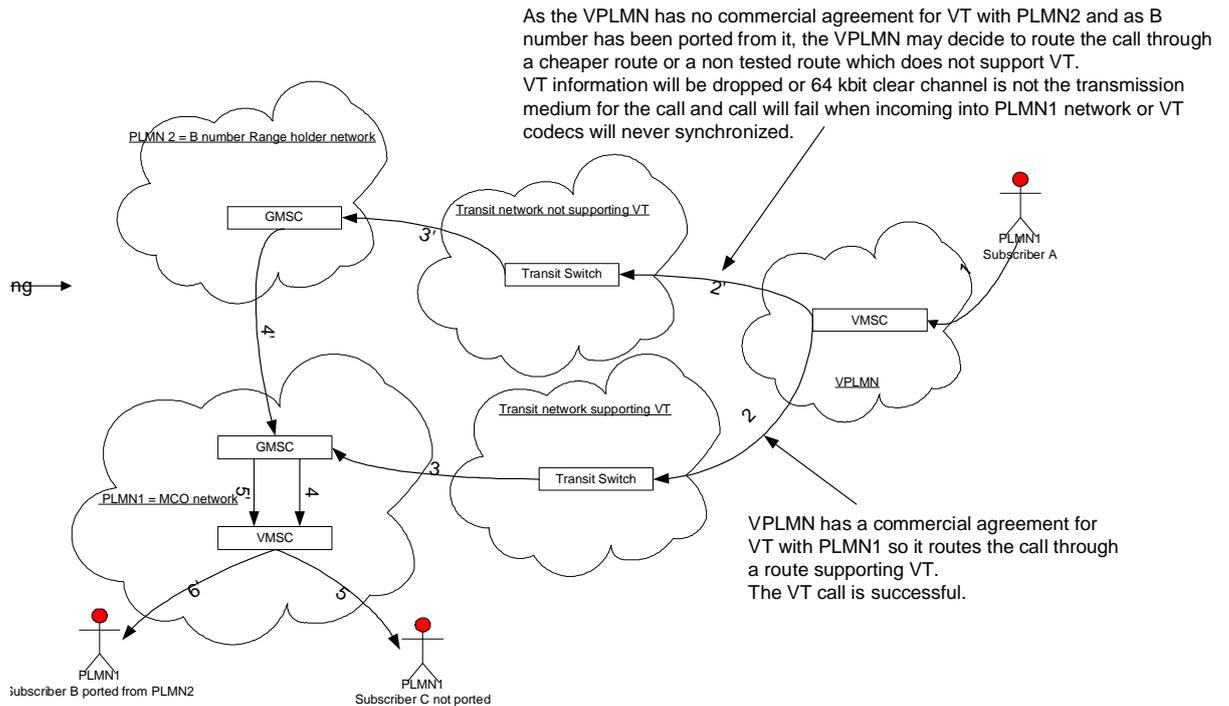
<sup>2</sup> This is the most important parameter which allows MNOs to differentiate VT from other services using GBS30.

international carrier that does not satisfy the transit interoperability requirements made in this document because they could be cheaper than the carriers supporting VT.

The figure below shows the potential issue in this case.

**5.7.2 Recommendation**

Each 3G roaming partner shall route UDI calls through a carrier satisfying requirements of Objective 1.3 when the called party is a number of a country where MNP applies even though it has no agreement with the range owner network.



## **6 SUMMARY**

This document has provided implementation guidelines and/or recommended best practice for 3G operators implementing CS VT services intra-network, inter-network and for roaming conditions, as an addition to GSMA PRD SE.34.

Basic sets of use cases are given that are sufficient for a new 3G operator to implement VT services. An examination of more complex uses cases and exceptions could be developed in further work.

The section on CS VT intra-network implementation provides general recommended parameters for:

- terminals
- radio bearers
- codecs
- multiplexing protocols
- core network requirements
- billing requirements

The performance of the resulting VT service should be measured by the KPIs that are also detailed.

The section on inter-network CS VT provides the requirements for the interconnecting transit carrier network(s) and significant details of the bearer capability fields required by the mobile network operators, as well as the transit billing arrangements.

The section on roaming gives a recommended method by which a VT call can be identified under a roaming scenario, in order to support appropriate charging, and to support the correct call set-up. A recommendation is also made for a method to route UDI calls in order to support MNP for roaming scenarios involving CS VT calling.

## 7 ANNEX A USE CASE FORMAT

### 7.1 Format

This is the format for Use Cases:

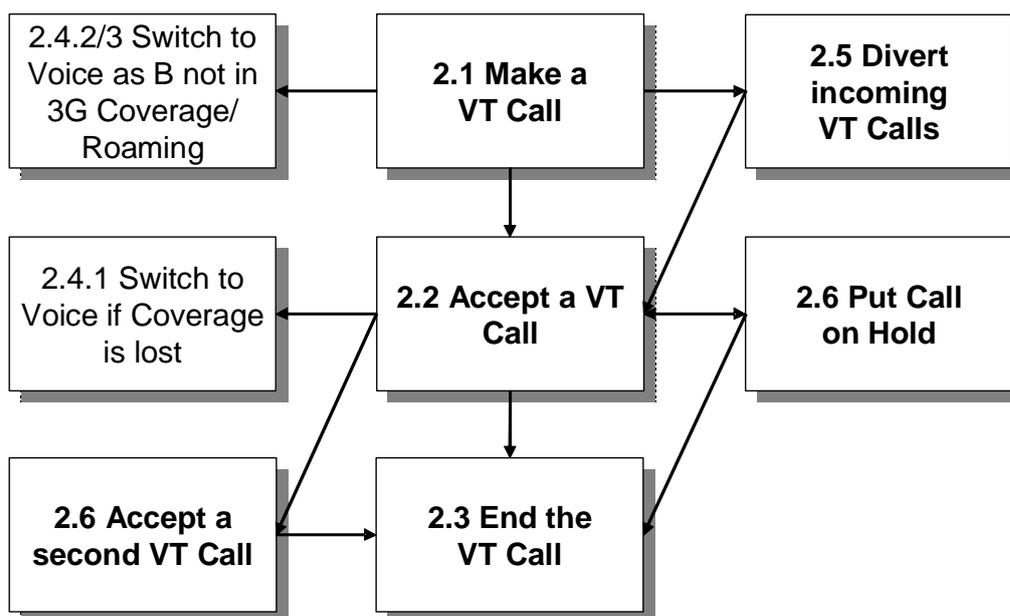
Use Case Name	Verb-noun phrase that states actor's intent and not their action
ID Number	Unique reference number for User Case (U##)
Preamble	
Preconditions – What must be true before Use Case begins	
Use Case Body	
Narrative, Scenario or Conversation	<p>Narrative:                      Free-form text in paragraph format                      Describes the intent of the user in performing the use case                      Describes high-level actions of the user during the use case                      Refers to key concepts from the problem domains that are involved in the use case.                      Good for describing high-level overview, but may hide detail from other groups. Use this first to generate Use Cases – identify those Use Cases where Scenario or Conversation may help clarify the Use Case</p> <p>OR</p> <p>Scenario:                      One particular path through a use case written from the actor's point of view                      Describes a sequence of events or list of steps to accomplish. A step may be optional.                      Each step is a simple statement                      May describe:                      (a) Actors' intentions (what they accomplish, not the minutiae of their gestures or individual keystrokes)                      (b) System responsibilities and actions</p> <p>OR</p> <p>Conversation:                      A dialog between the actor and the system that emphasizes interactions and shows cause and effect                      Can show optional and repeated actions                      Each action can be described by one or more sub-steps                      May describe:                      (a) Actor intentions and actions                      (b) System responsibilities and actions</p>
Supplementary Details and Constraints	
Exceptions— something expected that does not fulfils the user's goal during the execution of a use case e.g. system variations, MT preferences, error	Exceptions are deviations from the typical case that happen during the normal course of events – They should be handled, not ignored – How to resolve them can be open to debate

	<ul style="list-style-type: none"> <li>• What if a user mistypes her password?</li> <li>• What if an order can't be fulfilled?</li> <li>• What if a connection to a web browser is dropped?</li> </ul>
KPIs	Which KPIs are relevant to this Use Case
Post-conditions— what must be true about the system after a use case completes	
Constraints — constraints this use case must conform to	
Technical Comment	Response from technical groups on how this Use Case would be implemented or whether it raises any technical issues.

## 7.2 Inter-relationship of use cases

This section sets out the format to be used for Use Cases to be used by the VTIOP project. The Use Cases here do not necessarily replace Requirements. The use cases were written from a product perspective.

The diagram below outlines the relationship of the Use Cases with each other.



*Video Telephony Use Cases discussed in this document*

## 7.3 Post Conditions

The following conditions apply to all the use cases after execution of the use case. The post conditions are usually part of the individual use cases but the following were felt to be generic post conditions applicable to all use cases.

- System must be able to free resources for next calls
- Update “Calls Made” on handset
- System generates and stores CDR for call
- Inter-network CDRs are generated if appropriate

- Handset generates "Favourite numbers"
- Other Handset functions can be called – add to address book
- Pre-paid active credit management will decrease
- Common codecs