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**Reference**

**Document History**

Version	Date	Brief Description
3.0.1	24 <sup>th</sup> January 2003	This document describes operator requirements for Video Telephony services using a 64Kbps Circuit Switched Bearer.
3.1.0	5 <sup>th</sup> of December 2003	Additional requirements added to section 7 to support fall back from a video call to a UDI voice call.

**Changes Since Last Version**

Additional requirements added to section 7 to support fall back from a video call to a UDI voice call.

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## 1. Introduction

Mobile Video Telephony provides real time synchronised video- and audio communication between customers of this service. The customer is able to access the service via a mobile handset, which needs to contain a Video Telephony 3G-324M client for this purpose. In 3GPP Release 99 context usually a 64kbps bearer is deployed for the service. Video Telephony uses a GBS30 Multimedia call data bearer for synchronous data transmission [9]. Because this bearer is not available within 2G mobile telephony networks, video telephony can be considered as a service characteristic of 3G.

Video Telephony as defined by 3GPP is not a Teleservice but a conversational service that uses a data bearer for communication. As such Supplementary Services are only available as defined for data services.

## 2. Scope

Scope of this document is to insure a user friendly video telephony service that can be used across mobile networks, e.g. usage while roaming, set up of international calls etc. Therefore the document compiles all relevant 3G Video Telephony Service requirements, which are necessary to give a seamless and consistent user experience. This includes:

- Generic terminal requirements in order to give a consistent user experience when using different terminal types.
- Definition of the interworking and roaming requirements in order to insure a seamless user experience while roaming or making international Video Telephony calls.
- New requirements for the Video Telephony service in order to improve the usability of the service e. g. "Push to Video". These requirements shall be taken into account in future technical specification work.

It is not intended to define the service in detail so allowing operators to produce services that can be tailored to their particular markets and which ensures that operators will be able to compete on the services offered to customers.

## 3. References

- [1] 3GPP Release 99 TS 26.110 (V3.0.1). Codecs for Circuit Switched Multimedia Telephony Service; General Description.
- [2] ITU-T Recommendation H.324 (02/98). Audiovisual and Multimedia Systems. Infrastructure of audiovisual services — Systems and terminal equipment for audiovisual services.
- [3] 3GPP Release 99 TS 23.972 (V3.0.0). Circuit Switched Multimedia Telephony.
- [4] 3GPP Release 99 26.111 (V3.3.0). Codec for Circuit Switched Multimedia Telephony Service; Modifications to H.324.
- [5] 3GPP Release 99 TS 24.008. Core Network Protocols – Stage 3.
- [6] 3GPP Release 99 TS 26.911 (V3.2.1). Codecs for Circuit Switched Multimedia Telephony Service; Terminal Implementor's Guide.
- [7] ITU-T Recommendation V.8. Procedures for starting sessions of data transmission over the PSTN.
- [8] ITU-T Study Group 16, Question 15. H.263 Annex X Profiles and Levels Definition. Revision 3.
- [9] 3GPP Release 99 TS 22.002 (V3.6.0) Circuit Bearer Services (BS) supported by a Public Land Mobile Network (PLMN).

## 4. Video Telephony Use case Examples

Video Telephony enriches the communication between people, because the image component complements and adds value to the voice communication. This applies for situations in private live, e.g. while on vacation. Users of the service enjoy not only seeing the other person's face, but also sharing with them the scenery of a nice travel spot. In business live Video Telephony can be used as a powerful tool to increase efficiency. A typical example use case is the monitoring of work in progress on a site of ongoing construction.

The following three use case examples are presented in order to give a better understanding of the user experience as well as the implications to network and handsets.

### 4.1. Monitoring work in progress

An architect wants to check if the excavation is in a stage to start foundation work. The foreman who is on the construction side has a video-enabled handset, which the architect calls from his PC. The PC is equipped with a web cam. The reason for not using his mobile videophone is that the architect wants to store images from the construction side in order to include them in a report.

#### Implications to network and handsets

- Connectivity to the PC via PSTN, ISDN or Internet is required.
- 3G coverage is required on the construction side.
- The video handset should be designed in a way that the foreman can see the image he is transmitting in his own display. In addition he needs to communicate with the architect via voice so he is able to send the images of those parts, which are of most interest.

### Business traveller

A business traveller is calling his family from abroad. He chooses a video call instead of a voice call so he is able to see his wife and children and visa versa.

#### Implications to network and handsets

- Because the business traveller is abroad, roaming and the ability to establish an international video telephony call are required.
- Good indoor coverage is required because the business traveller is calling from inside his hotel room and the location of the other end of the call is the family house.
- The wife of the business traveller may not have accepted the call on her mobile videophone without the information that her husband is calling (CLI) because otherwise she may not have felt comfortable accepting the call.

### Watching the house

A homeowner checks his premises using a video phone placed in his house and a phone, which he carries with him.

#### Implications to network and handsets

- Because the house owner is on vacation overseas, roaming and the ability to establish an international video telephony call is also required to enable this scenario to work.
- The terminal placed within his home needs to have an auto answer feature because it should only transmit images when the owner of the house is calling.

## 5. Call Scenarios

At launch, UMTS services will be confined to islands of coverage, limiting the availability or likely success of mobile originated/terminated Video Telephony calls. As a consequence, GSMA SerG

has to promote interoperability between mobile and fixed networks to widen the VT service area. The potential scope of the generic Video Telephony architecture is illustrated in Figure 1 and is based on the 3GPP standard 3G-324M, [1].

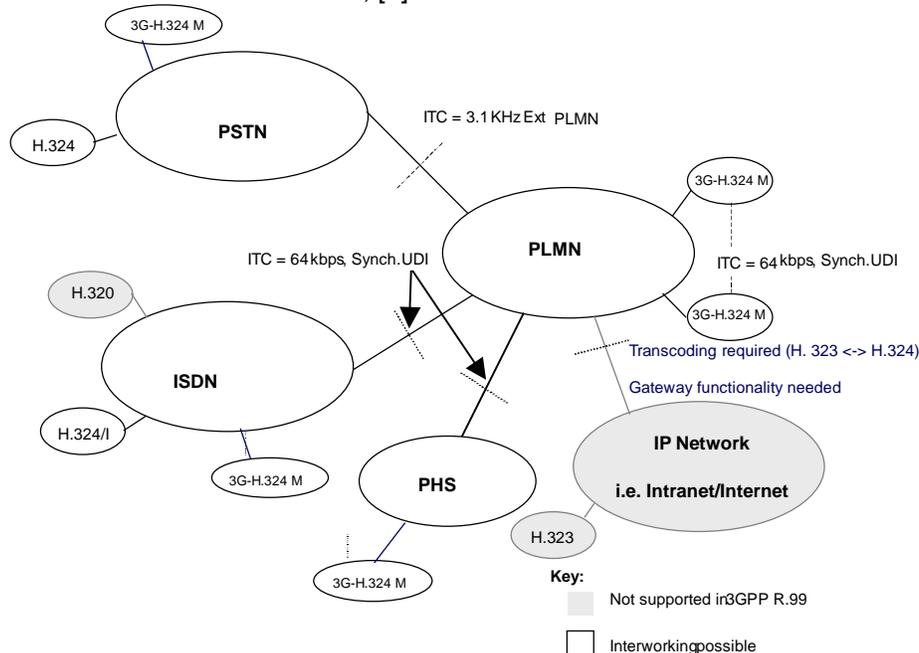


Figure 1. Generic Video Telephony Architecture

### 5.1. PLMN <> PLMN

This level of connectivity is fundamental and requires a synchronous, transparent, circuit switched bearer at either 32/64 kbps UDI for digital connections.

### 5.2. PLMN <> PSTN

This level of connectivity is characterised by a lower bandwidth (synchronous, transparent, 3.1 KHz audio CS bearer at 28.8 kbps) than the pure PLMN case, resulting in a degraded call quality (audio and video).

### 5.3. PLMN <> ISDN

In the CS domain, business video conferencing systems are based on H.324/I (Annex D) or the legacy H.320 standard. H.320 interworking requires a Gateway. In this instance connectivity is supported via 64 kbps UDI. H.324/I may require some transcoding to be performed e.g. via a transcoding gateway.

Terminals incorporating a client based on the 3G-324M standard can also be deployed within ISDN networks. This simplifies the interconnection scenario and hence reduces the need for transcoding.

### 5.4. PLMN <> IP Network

This scenario offers mobile access to IP network based video systems. As the 3GPP H.324M is used in 3G PLMNs and H.323 is used within the IP Network, a Gateway is needed to establish such a Video call. Note that there are several different levels of connectivity to connect to the IP Networks. These levels of connectivity may have to be considered within the gateway functionality.

### 5.5. PLMN<>PHS

This interconnection is similar to that for the PLMN<>ISDN scenario. It is supported within the Japanese market with a 3G-324M based client. Connectivity is supported via 64kbps UDI.

### 5.6. Interworking Options

Table 1 summarises the possible interworking options, which do not require any service-specific gateway (G/W) functions between the PLMN, PSTN and ISDN.

Interworking Options	3G-324M
<b>3G-324M</b>	✓
H.324	✓ (G723/AMR)
H.324/I	✓ (G723/AMR)
H.320	✗ (G/W)
H.323	✗ (G/W)
H.324M on PSTN	✓
H.324M on ISDN	✓

Table 1. 3G-324M Interworking options.

Note: The codecs defined in the specification ITU-T H.324 for H.324 (PSTN) and H.324/I (ISDN) are H.263 and H.261 for video and G.723.1 for audio. There may be interoperability problem with 3G-324M since the codecs defined as mandatory are H.263 for video and AMR for audio and as optional G.723.1. Therefore transcoding may be needed for the audio codecs.

## 6. Terminal and Network Requirements

This section outlines the requirements for Video Telephony capable terminals and networks. Based upon industry comments received regarding the Video Telephony service as defined in this document, the following levels are used to express the priority and importance of particular requirements requested by operators.

- Mandatory (M): the feature is extremely important and a top priority to operators.
- Recommended (R): the feature is important.
- Optional (O): the feature has been considered as an option, but is not required on the products. This may also mean that the feature will not be supported or deployed by many operators.

### 6.1. High Level Terminal Requirements

#### 6.1.1. Colour Display

VT 6.1.1.1	Minimum display dimensions: QCIF (176x144).	M
VT 6.1.1.2	During a Video Telephony call it shall be possible to show application icons, transmitted video and received video or mute options selected.	M
VT 6.1.1.3	Image re-sizing (fit-to-screen) for viewing, where the captured, stored or received image resolution is greater than the screen resolution.	R
VT 6.1.1.4	Display properties such as Resolution, Contrast, Brightness, Colour, White-balance, Zoom should be set by the handset automatically. There should be an option to switch to a manual mode for setting of these properties.	R
VT 6.1.1.5	Image shall be automatically reversed when using a camera rotation.	R

### 6.1.2. Camera

VT 6.1.2.1	Camera; to be built-in (preferred), a camera accessory is acceptable <sup>1</sup> .	M
VT 6.1.2.2	Low light operation < 1 Lux.	R
VT 6.1.2.3	Focal length: 20cm < FL < Infinity. (e.g. to read letters filmed on a paper or a photo)	R
VT 6.1.2.4	Image stabilisation.	R
VT 6.1.2.5	Video capture resolution = QCIF.	M
VT 6.1.2.6	180 degrees or over camera rotation, mechanically or otherwise. If available, the default setting shall be to automatically rotate the image all	R
<b>VT 6.1.2.7</b>	<b>Adjustable camera properties:</b>	
VT 6.1.2.7.1	All Adjustable camera properties such as Resolution, Contrast, Brightness, Colour on/off, White-balance, Exposure control, Focus, Zoom to be automatic.	O
VT 6.1.2.7.2	The user shall be able to manually adjust the camera properties.	O

### 6.1.3. Audio

VT 6.1.3.1	Support for built-in mono speaker and hands free microphone.	R
VT 6.1.3.2	Headphone and Microphone plug.	M

### 1.1.1 Basic 3G-324M Client Requirements

VT 6.1.4.1	The VT client shall comply with all requirements set out in 3GPP[1], [3], [4], [5],[6],[9].	M
VT 6.1.4.2	The VT service shall be indicated via a suitable display icon.	M
VT 6.1.4.3	Other services shall not be restricted during a VT call e.g. MMS and Browsing.	R

### 6.1.4. Audio Codecs

VT client audio codec requirements, [4], [6]:

VT 6.1.5.1	AMR	M
VT 6.1.5.2	G.723	O

To ensure interworking with the fixed network it is essential that no audio codec transcoding be performed at the network edge (QoS problem). Either the fixed Video Telephony Systems (Hardware and Software) have to support AMR audio codecs or the terminals have to support G.723 audio codec.

### 6.1.5. Video Codecs

VT client video codec requirements:

VT 6.1.6.1	H.263, Profile 0, encoding and decoding, [8].	M
VT 6.1.6.2	MPEG-4, Simple Visual Profile, Level 0, encoding and decoding <sup>2</sup> , [4], [6].	R
VT 6.1.6.3	H.263, Profile 3, for encoding and decoding, [8].	O

<sup>1</sup> If the camera is an accessory and not attached to the phone, incoming calls shall not be rejected.

<sup>2</sup> H.263, Profile 0, is a base component of MPEG4 hence its application also satisfies requirement VT 6.2.3.1

Due to the pre-existence of 3GSM MPEG4 compliant terminals and its wide application within the multimedia/internet community, MPEG 4, Simple Visual Profile, Level 0, encoding and decoding is a recommended requirement.

Note: Within the Release 99 3GPP specifications H.263, profile 0 (also known H.263 baseline) is the only mandatory video codec. As it's name suggests this codec only provides basic capabilities and hence a basic quality for the video component of video telephony. By the application of one of the video codecs stated as optional within the 3GPP specifications (i.e. MPEG4) the quality of video telephony calls can be improved considerably<sup>3</sup> (via improvement in performance characteristics such as error resilience, and compression efficiency etc...).

Note: H.263 Profile 3 comprises Advanced Intra Coding (Annex I), Deblocking Filter (Annex J) and Modified Quantisation (Annex T), to improve compression efficiency, and Sliced Structure Mode (Annex K) to improve error resilience.

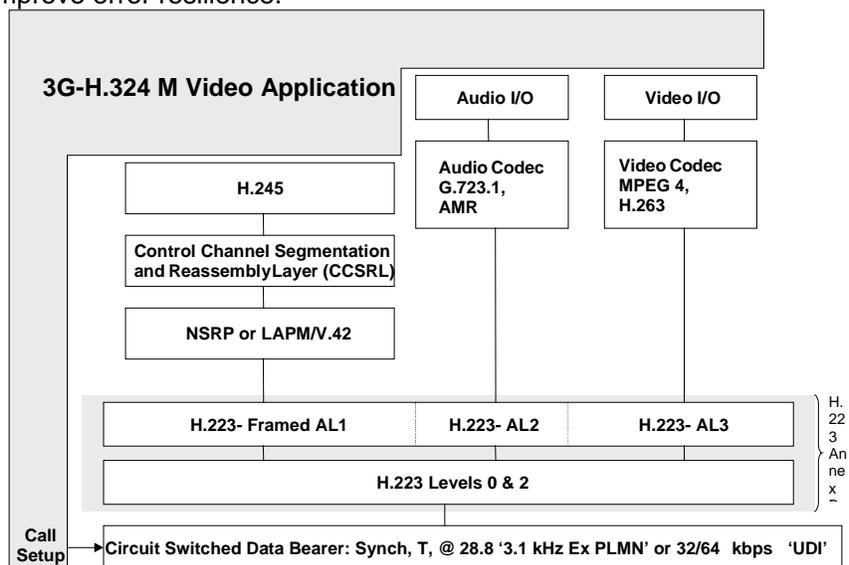


Figure 2. 3G-324M Video Telephony Client Architecture.

## 6.2. Call Set-up Requirements

VT 6.2.1.1	The multimedia call set-up requirements shall comply with the requirements defined by 3GPP, [5].	M
VT 6.2.1.2	It shall be possible to set up a Video Telephony call using a single numbering scheme.	M
VT 6.2.1.3	Each party shall be able to mute the video stream at call set-up or at any other time during the call.	M
VT 6.2.1.4	Each party shall be able to mute the audio stream at call set-up or at any other time during the call.	R
VT 6.2.1.5	Each party shall be able to mute both, the audio stream and the video stream at the same time. i.e. a call hold type functionality.	R
VT 6.2.1.6	The availability of a 3G network (i.e. network coverage) shall be indicated.	M
VT 6.2.1.7	A busy network indication should be given.	M
VT 6.2.1.8	At call set up and during the call, a Video Telephony icon shall always be displayed indicating the usage of the video telephony service, irrespective of the	M

<sup>3</sup> This quality improvement is only possible if the same optional codec is available end-to-end.

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	mute options selected.	
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### 6.3. PSTN to Mobile Calls

To support this call-case, Figure 3, the following functions are assumed at the MSC/IWF, [1], [5].

- For PSTN originated and terminated calls, the IWF modem-type shall be V.34 supporting V.8 in-band signalling, [8].
- The network has to support multi-numbering for PSTN originated data call cases, and the MSC shall signal both Multimedia and Speech BCIEs in the SETUP message sent to the mobile.

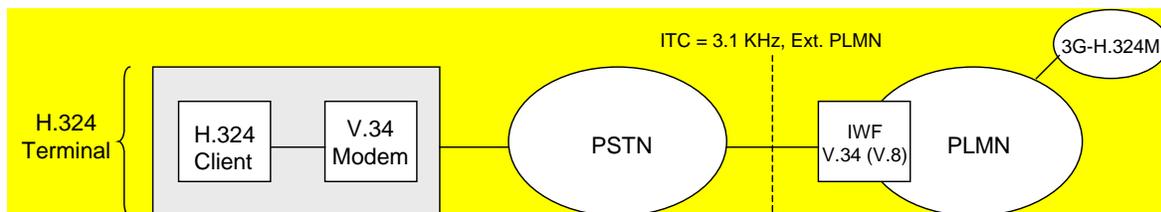


Figure 3. PSTN Interworking Call Case.

In the case that the mobile operator supports this call scenario, the following requirements shall apply:

VT 6.3.1.1	A mobile receiving both Speech and Multimedia BCIEs in a SETUP message shall accept both BC types in the corresponding CALL CONFIRMED to the network.	M
VT 6.3.1.2	Where Speech and Multimedia are accepted in the SETUP, the mobile shall be prepared to invoke In Call Modification procedures to fall back to speech (in the event of modem handshaking failure).	M

The network shall fulfil the following requirements.

VT 6.3.1.3	IWF modem-type shall be V.34 supporting V.8 in-band signalling.	M
VT 6.3.1.4	Support of multi-numbering for PSTN originated data calls.	M

Note, the default time-out for V.8 in-band signalling failure is 3s.

Where ITC = '3.1 kHz Ex PLMN' (i.e. PSTN > PLMN), fallback to speech is possible (subject to V.8bis network support) during the call set-up phase in 3GPP R.99. This feature is not supported for MO or UDI calls.

### 6.4. ISDN to Mobile Calls

For calls set up with a FNUR of 64Kbits/s, an IWF is not necessary for ISDN to Mobile calls.

In the case that the mobile operator supports this call scenario, the following requirements shall apply:

VT 6.4.1.1	Requirements for Mobile to/from ISDN calls shall comply with 3GPP, [5] [3].	M
VT 6.4.1.2	24.008 call control signalling should interwork with the corresponding Q.931 (ISDN D-Channel) signalling.	M

#### 6.4.1. H.324/I Client within the ISDN Domain

This call scenario is a basic feature for 3GPP-R99. H.324/I terminals have to be enhanced in order to be able to interwork with 3G-H324M terminals. The appropriate annexes, which allow multimedia services in a wireless network, need to be implemented.

The codecs supported by H.324/I are different from the codecs 3G-324M. A transcoding gateway may be needed. To ensure interworking with fixed networks it is essential that no audio codec transcoding is performed at the network edge (QoS problem). Either the H.324/I Client (Hardware and Software) has to support AMR audio codecs or the terminals have to support G.723 audio codec.

The interworking function (IWF) is not necessary for H.324/I to mobile calls set up at 64Kbits/s.

Note: A V.34 modem is not used for H.324/I. The specification H.324 quotes that the H.324/I terminals shall use an I.400-Series ISDN user-network interface in place of the V.34 modem. All references to the "V.34 modem" in the Recommendation for H.324/I shall be replaced with "I.400-Series ISDN user-network interface".

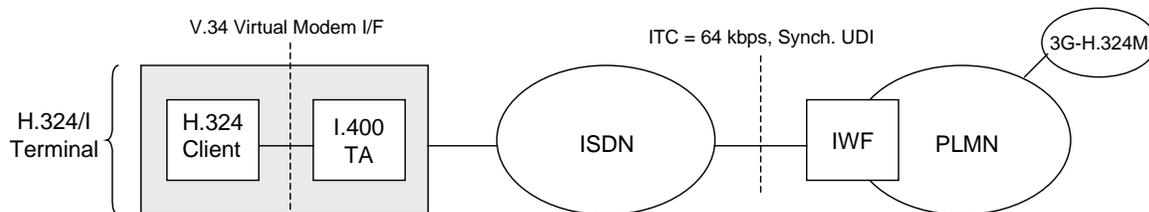


Figure 4. The ISDN(H.324/I) Interworking Call Case.

#### 6.4.2. 3G- 324/M Client within the ISDN Domain

An alternative ISDN Interworking scenario is shown below in figure 5. In this scenario a 3G-324M client is provided within the ISDN domain. Calls are possible over a UDI bearer at a data rate of either 64 or 32Kbps.

As the Client in the ISDN domain is similar to that present in the 3G mobile network (i.e. both are compliant to the 3GPP standard 3G-324M Codec for Multimedia Telephony) the need for transcoding within the PLMN is significantly reduced. This can lead to improved call quality and reduce implementation costs for the mobile operator.

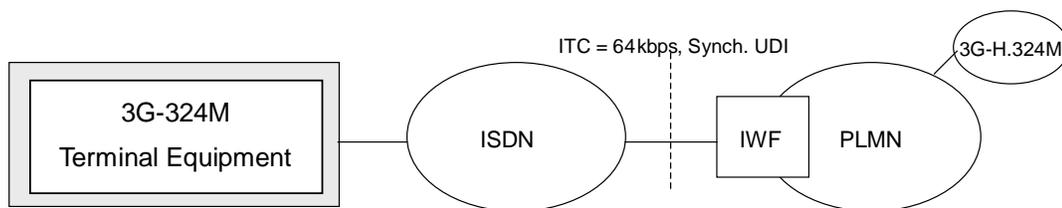


Figure 5. ISDN (3G-324M) Interworking Call Scenario

#### 6.5. Mobile Originated Calls

The VT client architecture, Figure 2-, indicates several bearer options that need to be defined at call set-up:

<b>Information Transfer Capability</b>	UDI or '3.1 kHz Ex PLMN', Note 1
<b>Duplex Mode</b>	Full duplex
<b>Sync/Async</b>	Synchronous
<b>Connection Element</b>	Transparent
<b>Fixed Network User Rate</b>	64, 32 kbps or 33.6, 28.8 kbps, Note 1, 2
<b>Other Modem Type</b>	V.34, Note 3
<b>User Information Layer 1</b>	H.223/H.245 (standardised in 24.008)

Note 1, this field shall only indicate one value.  
 Note 2, although 3GPP, [5], only defines FNUR > 32 kbps, support of 28.8 kbps is needed. This rate is a mandatory V.34 symmetrical rate.  
 Note 3, only applies where ITC = '3.1 kHz Ex PLMN'

Table 2. MS→MSC SETUP BCIE.

### 6.5.1. Automated retry schemes

In case of setting up a video call to a phone, which does not support the bearer on which the call is set up, or Video Telephony at all, the call will fail. The exception to this is the set-up of an analogue video call, which would automatically fall back to voice in case the data connection necessary for Video Telephony cannot be established.

In order to achieve a higher overall call success rate, the terminals should provide the functionality to automatically try the set up of a new call of a different type (i.e. normal voice call) to the same destination. The customer shall be able to enable and disable the feature as well as to configure it according to his needs (i.e. which bearers to be retried in which order).

The most likely case is probably that the terminal of the opposite party does not support Video Telephony or is out of 3G coverage. In this case it's sensible that the terminal of the calling party tries to set up a voice call to the same destination after the failure to successfully set up a video call. In the case that a multi numbering scheme is used, the terminal may need to take this into account.

VT 6.5.1.1	Terminals shall support automated retry schemes.	R
VT 6.5.1.2	In the case that automated retry scheme are supported the user shall be able to enable and disable the feature.	M
VT 6.5.1.3	In the case that automated retry schemes are supported, the user shall be able to configure the retry scheme (e.g. which bearers are to be retried in which order, a manual mode configured by user).	R

## 6.6. Mobile Terminated Calls

VT 6.6.1.1	Requirements for Mobile Terminated Calls shall comply with 3GPP, [5] [3].	M
VT 6.6.1.2	An incoming video telephony call shall be indicated clearly, e.g. with a suitable icon or message.	M
VT 6.6.1.3	It shall be possible for the recipient to reject an incoming video telephony call.	M

## 6.7. Auto Answer Features

An auto-answer feature is required for remote monitoring use cases e.g. a call between a 3G terminal and a VT terminal connected to the PSTN/ISDN/PLMN. For security reasons, any auto-answer feature must be conditional on authentication of the calling party. This may be provided by the use of a CLI *White List*, populated by the user. An alternatively authentication procedure may be the use of a PIN number/username and password or a combination of the two. However, the exact method of authentication is FFS.

VT 6.7.1.1	The VT client should support an auto answer user option when CLI information is available.	R
VT 6.7.1.2	If Auto Answer Features are supported invocation shall be conditional on authentication of the calling party e.g. by use of a CLI <i>White List</i> (The exact method of authentication is for FFS).	M
VT 6.7.1.3	If Auto Answer Features are supported in cases where CLI is not available, an alternative authentication of the A-Party shall be provided, e.g. using a PIN number, Username & Password etc.	M

## 7. Future Requirements

### 7.1. Possible future call scenarios

#### 7.1.1. PLMN<>TV cable

This scenario is quite different but cable telephony could be a way to increase the market. This service should be provided with a specific equipment to connect camera on TV and a gateway.

### 7.2. Switch from video to voice Coverage triggered

Coverage for services provided over Mobile Networks is dependent upon an operator's deployment policy. However, physical constraints can result in the coverage for higher bit rate services, such as Video Telephony, to be reduced compared to that for voice.

Consequently, the availability of a mechanism to enable switching from UMTS video call to UMTS voice call is required

Such mechanism will have to guarantee satisfactory user experience preventing the Video Telephony service to be dropped when user is heading out of a cell. In this case, a video call shall be switched to voice in a seamless manner without any action from the user.

In case of a user recovering video telephony coverage, it should be possible to switch back from voice service toward video.

In the early days of UMTS, this is an especially likely scenario and bad user experience could jeopardize adoption of 3G services. As such investigation of suitable technical realizations is necessary within 3GPP standards.

The following requirements are applicable to this functionality:

VT 7.2.1.1	Video -> UMTS voice switch coverage triggered	M
VT 7.2.1.2	UMTS voice -> Video switch coverage triggered	R

### 7.3. Handover between UMTS and GSM

Inter-system handover between UMTS and GSM is not supported for 64 kbps UDI calls. In the case of a user moving out of 3G coverage, the video call will be released. The user has to establish a voice call either manually or the handset provides a voice call set up automatically. In the early days of UMTS this is an especially likely scenario. Therefore, the handover from 3G to 2G linked with a downgrade from video to voice is a desirable functionality.

The investigation of suitable technical realisations is necessary and is therefore recommended. It is further recommended for discussion in future 3GPP specification work.

VT 7.3.1.1	Video -> voice downgrading handover between UMTS and GSM.	R
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### 7.4. Handover between UMTS and WLAN and Bluetooth

Inter-system handover between UMTS and GSM is not supported for 64 kbps UDI calls. In the case of a user moving out of 3G coverage, the video call will be released. VT should be available via Bluetooth or WLAN if the network is available and if the terminal has this functionality

The investigation of suitable technical realisations is necessary and is therefore recommended.

VT 7.4.1.1	Handover between UMTS and Bluetooth/WLAN.	R
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### 7.5. In-call modification

In call modification allows the customer to easily switch between a voice and a video call. In an ideal environment the customer who is in a voice call can convert the call into a video call with one keystroke seamlessly and visa versa. From the user point of view, the feature should be available from both, voice or VT call originated to switch to the other mode during a call one or many times at each moment.

The voice charge should be applied when the voice call is on and the video charging when the video is on.

During the modification, the appropriate bearer should be selected to permit the right service with full transparency for user. During modification the previous service should stay active in voice mode at least without stopping the call and, the new service selected should be negotiated in background. After acceptance by the other party, the new service should be active and the first one should be shut down in background. Duration of the modification should be as short as possible and transparent from the user point of view.

In-call modification of the underlying bearer for voice and video modes is currently outside the scope of the 3GPP Release 99 and H.324 specifications. Because the functionality is considered to be important, the standardisation of in-call modification would be highly desirable.

In the case that this functionality is supported, by the mobile operator, the following requirements shall apply.

VT 7.5.1.1	In-call modification of the underlying bearer for voice and video modes.	R
VT 7.5.1.2	In-call VT features should be supported when the terminal is in 3G coverage.	M
VT 7.5.1.3	In-call VT features shall not be available when outside 3G coverage.	M
VT 7.5.1.4	When In-call modification is available; differentiation of call components for billing purposes.	M
VT 7.5.1.5	When In call modification is available; an incoming VT call should be differentiated from in call modification switching.	R
VT 7.5.1.6	During the modification, the appropriate bearer should be selected to permit the correct service with full transparency for the user.	R
VT 7.5.1.7	During modification the previous service should stay active in voice mode at least without stopping the call.	R
VT 7.5.1.8	After acceptance by the other party, the new service should be activated and the previous one should be shut down in background.	R
VT 7.5.1.9	Modification duration should be as short as possible and transparent from the user point of view.	R

In this case both call components, video and voice have to be differentiated in order to enable charging on a different rate for voice and video.

### 7.6. Fall back to speech for UDI calls

There are several reasons why the set up of video call can fail, for example the called terminal does not support video telephony or is currently out of 3G network coverage. From the customers point of view this experience is rather unsatisfactory. Functionalities like automated retry schemes help to improve this situation but are not ideal because of the additional waiting time for the customer until the new call is set-up. Therefore, the availability of this feature is highly desired and support of this feature is recommended.

Fall back to speech for UDI calls is not available in 3GPP Release 99 or Release 4 Video Telephony specifications. However such feature would significantly improve the experience of user and is therefore recommended for further standardisation work.

VT 7.6.1.1	Fall back to speech for UDI calls.	R
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### 7.7. Emergency calls

Especially in emergency situations, the Video Telephony service can be of great aide. Through the visual information it is much easier for the emergency bodies to understand the appropriate level of help needed in the particular situations like a car crash or a fire.

In the following a short consideration of future terminal and network requirements suitable for this application is given. However the realisation of Video Telephony emergency calls is considered to be highly dependent on national laws and regulations. It is included in the document as stimulation for further development.

- In case of a VT emergency call, the call could be accepted in VT mode by the emergency services without any call acceptance defined for VT.
- If the calls is rejected by the emergency services because they are not equipped for VT calls or due to networks error, the handset automatically redials the emergency number in voice mode independently of the call retry configuration to increase the call success rate. Call retry should be possible in case of fall back to another service (speech, handover to GSM network).

If the emergency services are equipped with VT, when receiving an emergency call (in both voice and VT modes), it would be beneficial for the emergency call centre to be able to activate the camera via remote control.

### 7.8. Video card

Video card is a service to introduce yourself to other party by transmitting personal information associated with a photo presented in real time during the call set up.

Video card should be based on evolution of CLIP service to promote use of picture/video for VT calls for business and individual uses (e.g. call centre calling a consumer and sending the logo and name of society in picture plus the text name and photo of caller).

Video card at call set-up should be available for all VT calls. In case of incoming calls, it should be possible for Video cards to be presented even from non-PLMN networks.

User should create a personalized card including multimedia content (e.g. picture, audio or video) and personal information.

Video card should be sent manually or automatically. The end user should be able to store caller cards within his address book.

## 8. Supplementary Services for Video Telephony

Currently only those supplementary services for data calls apply because Video Telephony calls are handled as such. Those are:

- Call waiting
- Call forwarding on:
  - Busy
  - Not reachable
  - No reply
  - All calls
- Call barring
- CLIP/CLIR
- Call completion to busy subscriber

However, in addition "Cold hold" would be a desirable supplementary service to toggle between two video calls.

8.1.1.1	Cold hold	R
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## 9. Charging principles

Charging principles for the Video Telephony service is subject to the GSMA document BA 27.

## 10. International calls and roaming

As Video Telephony depends on the 64kbps UDI bearer, for national as well as international and roaming calls, such a bearer must be provided by all carriers used for this specific connection. Presently many long distance carriers do not guarantee this functionality.

If the 64kbit/s UDI bearer is not provided end to end, this may lead to SS7 signalling issues. In this case it might not be possible to use a primary MSISDN for mobile terminating VT calls and a secondary MSISDN will have to be used to separate VT calls.

VT 10.1.1.1	Support of 64kbit/s UDI bearer -end to end- for Interconnection	M
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## 11. Glossary

3GPP	3rd Generation Partnership Project
AMR	Adaptive Multi-Rate
BC	Bearer Capability
BCIE	Bearer Capability Information Element
CBS	Cell Broadcast Service
CDR	Call Detail Record
CLI	Connected Line Identity
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
CoLP	Connected Line Presentation
CSD	Circuit Switched Data
FACH	Forward Access Channel
FNUR	Fixed Network User Rate
FTM	Frame Tunnelling Mode
GBS30	General Bearer Service (number) 30
ISDN	Integrated Services Digital Network
ISG	IMT 2000 Steering Group
ITC	Information Transfer Capability
IWF	Interworking Function
MMS	Multi Media Messaging
MO	Mobile Originated
MPEG	Motion Pictures Experts Group
MSISDN	Mobile Station ISDN Number
MT	Mobile Terminated
NT	Non Transparent
PCH	Packet Channel
PLMN	Public Land Mobile Network
PPP	Point to Point Protocol
PS	Packet Switched
PSTN	Public Switched Telephone Network
QCIF	Quarter Common Intermediate Format
QoS	Quality of Service
RAB	Radio Access Bearer
RB	Radio Bearer
RNC	Radio Network Controller
SS7	Signalling System 7

T	Transparent
TAP	Transfer Account Procedure
UDI	Unrestricted Digital Information
UE	User Equipment
UMTS	Universal Mobile Telecommunications System
VT	Video Telephony
WAP	Wireless Application Protocol