Video Share Interoperability Specification
1.3
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Security Classification Category (see next page)	
Not Restricted	

Security Classification - UNRESTRICTED

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Document History

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Feedback

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1 INTRODUCTION

This document specifies the terminal interoperable Real-Time Live Video Share service. The intended audience of this document is terminal or software vendors who wish to implement an inter-operable Video Share service. Mobile operators, seeking to implement the Video Share service, can also refer to this document for a more technical understanding of the service. The GSMA has also produced a *Video Share Service Definition* document (GSMA SE.41) which describes the Video Share service to aid GSMA Working Groups in their work on Video Share. This can also be referred to for a more general introduction on Video Share.

Terminal interoperable Real-Time Live Video Share service allows users to share live video between them over PS connection in real time simultaneously with ongoing CS call, thus enhancing and enriching end-users voice communication. Video Share is a one-to-one combinational service utilizing 3GPP compliant IMS core system and 3GPP CSI (TS 24.279) based solution, session is set up using SIP and video is transferred using RTP. Video Share uses P2P model, i.e. applications are built in terminals thus a separate Application Server in network is not needed.

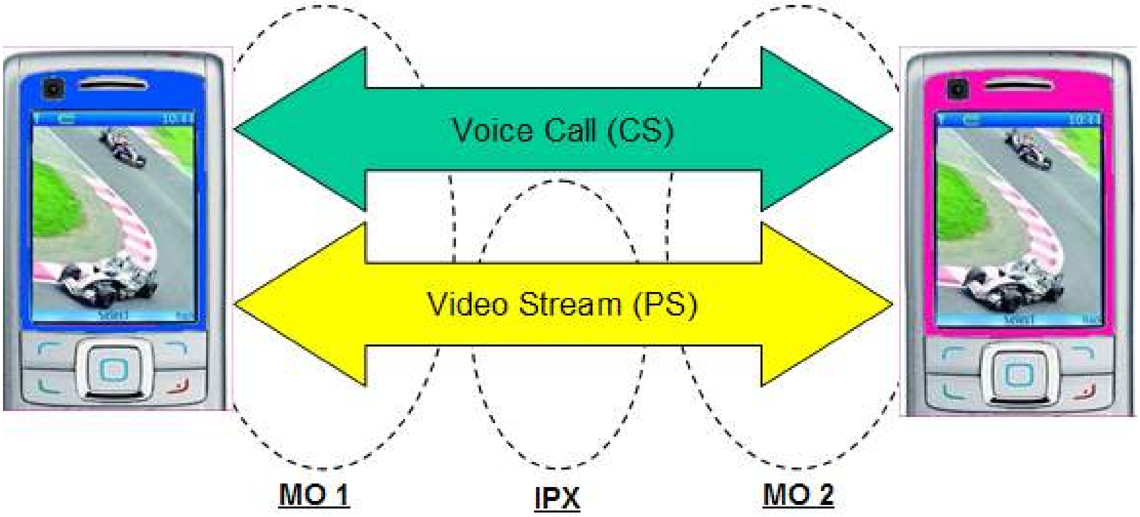

Figure 1: High-Level Figure of Video Share Connection

Video Share service is a vendor independent application, i.e. interoperable between different terminals, as well as between terminals and different IMS core systems.

The first deployment of Video Share service was performed in GSMA SIP Trial TCVS01 Test Campaign.

Note: The term "*P2P*" in this context means "Peer-to-Peer".

2 SCOPE

2.1 In Scope

The aim of this document is to present the technical principles for terminal interoperable Real-Time Live Video Share service.

2.2 Out of Scope

Out of scope for this particular document are general issues not directly related to Video Share service itself. For example 3GPP compliant IMS core systems are prerequisite for Video Share, but they are not detailed in this document.

Conformance testing/certification in general are out of scope for this document.

Also out of scope for this release of document are:

- Other services/applications
- PSTN related issues
- Commercial issues
- Back-office functions (e.g. O&M)
- Load-balancing, high availability etc

3 VIDEO SHARE

3.1 General

Basically Video Share session consists of the following steps:

1. CS call setup
2. Capability query
3. Invitation procedure (SIP)
4. Video transmission (RTP)
5. Teardown of video session
6. Teardown of CS call

The following figure illustrates general flows used in Video Share. Note that figure is simplified for clarity reasons, for example network elements between UEs are not shown.

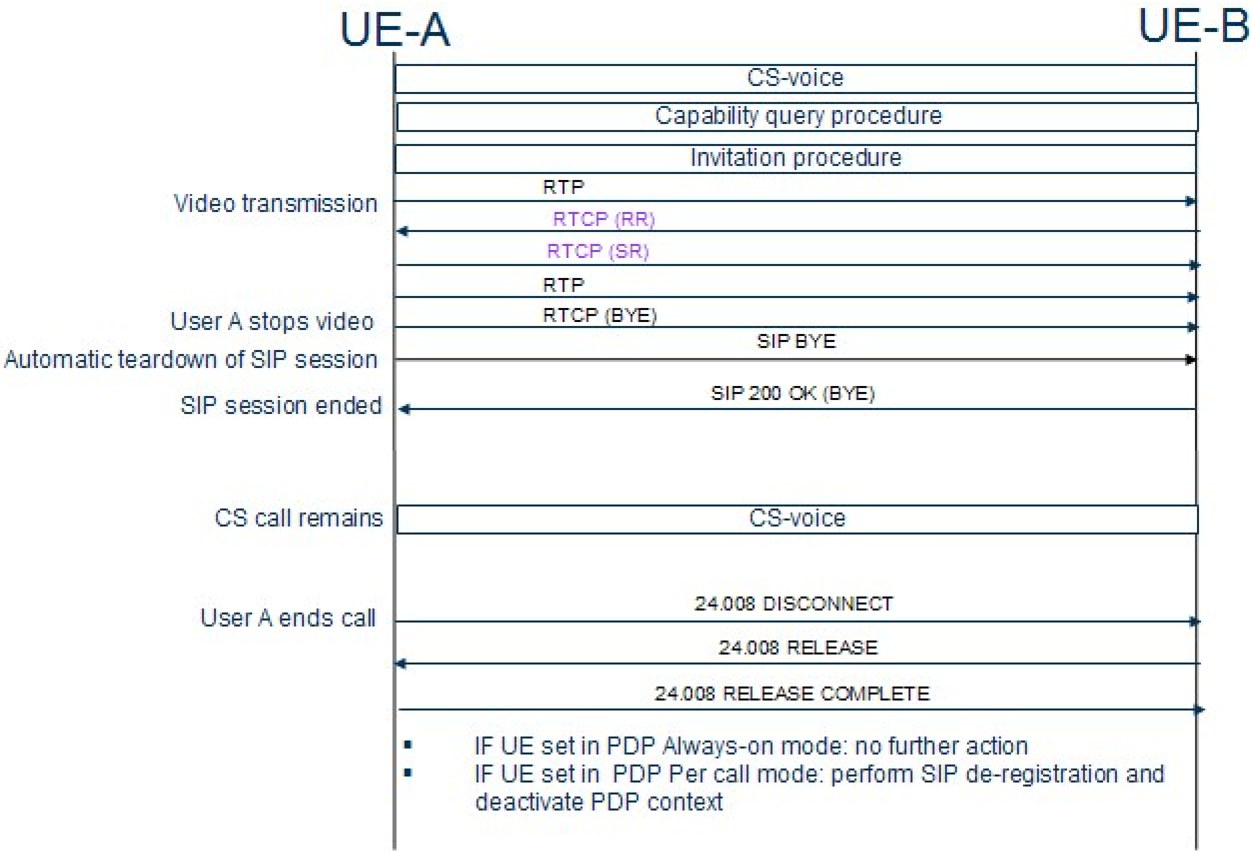

Figure 2: General View of Flows in Video Share Service

Besides the case in figure 2 above, the SIP session is torn down by the terminal (A or B) party that is not receiving any RTCP reports anymore from the other terminal e.g. due to the other terminal has made a handover to a non-DTM 2G access during the video transmission phase. See RFC 3550 for further details of this timeout functionality.

3.2 Service Identification

Feature tag (based on RFC3840 & 3841) indicates that terminal is capable of supporting certain media features. Feature tag structure used for Video Share: *+g.3gpp.cs-voice*. This implies that terminal supports normal CS Voice call used as a part of Video Share service.

Following SIP messages carry this Feature tag:

- INVITE (in Accept-Contact and Contact header)
  - 200 OK (in Contact header)
- OPTIONS (in Accept-Contact and optionally in Contact header)
  - 200 OK (in Contact header)
- REGISTER (in Contact header, handling of Feature tag in REGISTER method in the network is optional)
  - 200 OK (in Contact header)

Usage of Feature tag is mandatory for “new terminals”, but also “legacy terminals” (as described in Chapter 4) not using it will be supported.

Note: other possible Feature tags (such as for PoC service) are not excluded. For example, a terminal may optionally in the SIP messages above set an additional feature tag(s) besides *+g.3gpp.cs-voice*. The sending terminal shall not require the receiving terminal to understand the additional feature tag(s). Hence, the receiving terminal may ignore the additional feature tag(s) and handle only the *+g.3gpp.cs-voice* feature tag. A terminal that chooses to utilize the *+g.3gpp.cs-voice* feature tag and additional feature tag(s) when sending SIP messages, shall be able to receive SIP messages (INVITE, OPTIONS, 200 OK) that do not include any additional feature tag(s), and still invoke video share.

Preferred way for GSMA is to let Feature tag(s) be set in every SIP message that can carry a feature tag. I.e. in every SIP message that has a contact header and/or an accept-contact header, both requests and responses.

In addition to the Feature tag mechanism described above, it is possible to utilize OMA Presence Simple enabler for exchanging the capability information. Value "*org.gsma.videoshare"* of the field <service-id> inside the Presence data element <service-description> indicates support for Video Share service. For further information refer to OMNA Presence <service-description> Registry:
http://www.openmobilealliance.org/Technical/omna/omna-prs-PidfSvcDesc-registry.aspx

3.3 Capability Query

Video Share session begins with CS call between UE A and UE B. After call is set up, capability of the other terminal can be queried, i.e. is recipient capable of supporting Video Share session or not. This is performed with SIP OPTIONS method. Positive response to query is sent using 200 OK. Both UEs can perform this query. Additionally it is possible to utilize OMA Presence Simple enabler for exchanging capabilities. This is performed using the Presence <service-description> element as illustrated in the previous Chapter.

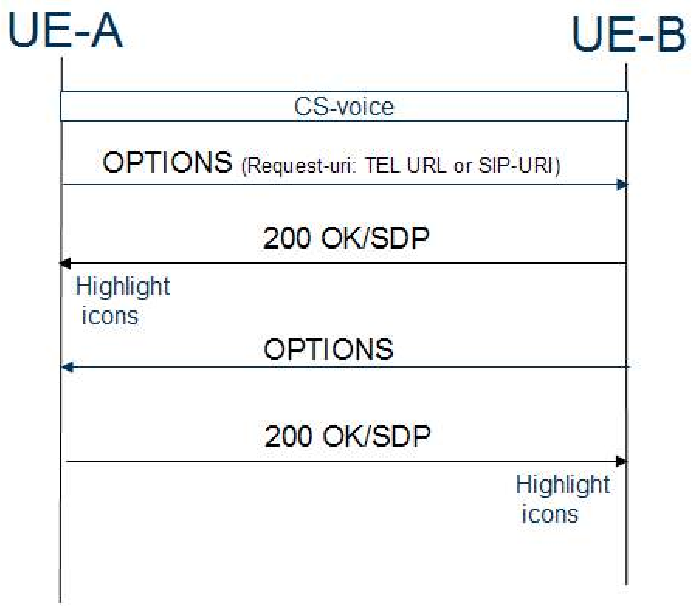

Figure 3: Capability Query using SIP OPTIONS

Icons can be used in terminal UI to show user that Video Share session towards this particular recipient can be set up, in case recipient indicates supports for Video Share service.

To have SDP in OPTIONS message is optional and it is not encouraged behavior to insert it to this message. Terminal receiving OPTIONS must be capable to handle OPTIONS request with or without SDP in the message. SDP in 200 OK for OPTIONS is mandatory and shall contain relevant m=video and a=rtpmap lines. Terminal receiving 200 OK for OPTIONS should make decision about other terminals VS capability based on existence of feature tag and content of m= and a= lines.

The contents of the OPTIONS response shall contain sufficient information for the terminals to negotiate a peer to peer streaming connection. The response includes:

- The coder/streaming capability of the terminal
- A feature tag indicating that the mobile can support the combination of voice and IMS services (specifically: +g.3gpp.cs-voice)

SDP information shall contain the following kind of information for Video Share (UE B responses with 200 OK to OPTIONS query):

m=video 0 RTP/AVP 96
a=rtpmap:96 H263-2000/90000
a=application:com.*company.service* (OPTIONAL)

If optional codecs, such as MPEG4 and H.264/AVC, are supported then the SDP attributes in the SIP response will contain parameters associated with all supported codecs.

Note that in some cases only UE B uses OPTIONS query, for example if UE A has capability information of UE B already available (cached from earlier query) or UE A doesn’t support OPTIONS method.

According to 3GPP specifications it is possible to use either parallel or sequential method of capability query. UE B can send OPTIONS immediately (in parallel to UE A) or UE B can send OPTIONS only after being queried first by UE A.

Instead of 200 OK it is possible to receive an error message, such as 4xx (except 480), 5xx (except 501) or 6xx SIP error. *501 Not Implemented* implies that UE B doesn’t support OPTIONS, but may still support the Video Share service. If 480 or 408 is received with a Retry-After header with a value of X seconds, UE may retry once after X seconds. Otherwise the UE shall retry after 10 seconds. If this fails it will retry again after a further 20 seconds. If it fails after that point, no more retry attempts are made for that particular Video Share session. Note: error message can also come from the network instead of UE.

It is mandatory for a terminal to respond to an OPTIONS request. In scenarios of Multi-Party Call, Call Hold and Call Forwarding, the terminal's response to the OPTIONS request (service discovery) should indicate VS incapable.

If the SIP URI was not obtained during the options procedure, then the MSISDN (tel URI) of the terminating subscriber shall be used to initiate the SIP transaction by the originating subscriber.

3.4 Session Setup

After capabilities of both ends are known, the next step is the actual Video Share session setup. SIP session setup for Video Share can be performed using either IETF mode or IMS mode of SIP signaling (both modes shall be supported with the ability provided to the operator to configure one mode as ‘per default’ mode, subject to its own operator policies and preferences):

- IETF mode: IETF RFC 3261 standard flow applies
- IMS mode: 3GPP Rel-5 pre-condition flows applies

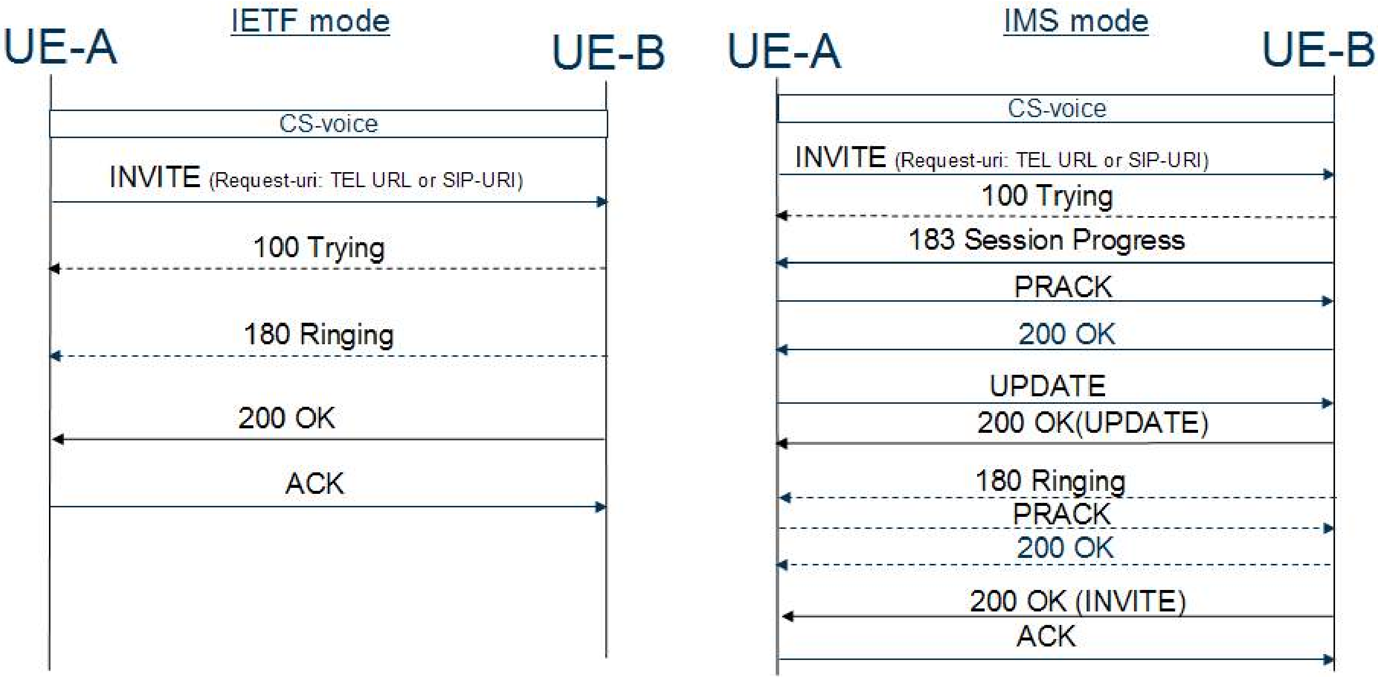

Figure 4: Session Setup in Video Share

Used IETF/3GPP IMS session set-up model: UE A chooses the mode, UE B shall follow (“legacy terminals” as described in Chapter 4 excluded). Optimal location for performing the fallback is in the terminal that receives the INVITE. There will be no additional signalling in the network and there is no additional set-up delay. For example:

- If incoming INVITE indicates require:precondition receiving terminal will behave according to IMS mode even if it would be configured to be in IETF mode.
- If incoming INVITE doesn't indicate support:precondition received terminal will behave according to IETF mode even if it would be configured to IMS mode.

It is mandatory to support fallback between 3GPP and IETF modes, due to different operator preferences and “legacy terminals”, for instance.

To initiate a video share session, an INVITE shall be sent to the tel URI or SIP URI of the other device. Following example shows what kind of information is carried in invitation procedure:

INVITE	**200 OK**
(Accept-Contact: +g.3gpp.cs-voice;	*(Contact: +g.3gpp.cs-voice)*

explicit) m=video *portUE-A* RTP/AVP 96 a=sendonly a=rtpmap:96 H263-2000/90000 a=framesize:96 176-144 a=framerate:8 a=fmtp:96 profile=0; level=45	m=video *portUE-B* RTP/AVP 96 b=AS:54 a=recvonly a=rtpmap:96 H263-2000/90000 a=framesize:96 176-144 a=framerate:8 a=fmtp:96 profile=0; level=45

Note that sections within brackets are Feature tags. Other lines present video codec & other features used for this Video Share session.

INVITE message can contain more standard headers than the ones explicitly mentioned above. One of them is the P-asserted-Identity header. If it is included and it contains the tel URI of the sender, the recipient UE can use this value to check whether the incoming SIP request matches the user in the CS call. Even if the user chooses to hide their CLID for voice calls, the “From:” address of all SIP messages must not be anonymous.

Replies to the incoming INVITE message shall be 486 or 603 Busy when the recipient does not accept the Video Share invitation.

In IMS mode use of preconditions shall be implemented according to 3GPP TS 24.229 and in accordance with RFC 3312 Integration of Resource Management and Session Initiation Protocol (SIP).

The SIP INVITE for Video Share shall include the SDP information with Media type, codec supported and RTP Port Information.

Feature tag shall be included in INVITE. SDP attribute a=application is not needed, since usage of Feature tag fulfills the same requirement. Note however that implementation from terminal vendors with "legacy terminals" that don’t support Feature tag shall use a=application attribute in SDP such as a=application:com.nokia.rtvs and that these vendors may continue to set their a=application line also in their new terminals supporting feature tag. This a=application line may appear before the m-line or after the m-line.

SDP attribute optionality is described below:
- a=framesize (Optional)
- a=framerate (Optional)
- a=rtpmap (Mandatory)
- a=fmtp (Mandatory by terminating side)
- a=recvonly (Mandatory by session terminating side)
- a=sendonly (Mandatory by session originating side)
- b-line (Mandatory by session terminating side)
- m-line (Mandatory)

Either user shall be able to cancel the call at any point during the call setup.

3.5 Other Features

Radio access for Video Share is WCDMA*
- 64kbps mandatory
- 128kbps optional
- Interactive class used, streaming class is optional

Video codec H.263-2000 profile 0 level 45 is mandatory. When using the H.263 video codec, only QCIF resolution video shall be supported for Video Share.
- MPEG4 Visual Simple Profile 0b optional, using the RTP format in RFC 3016

- H.264/AVC Baseline Profile Level 1b is optional, using the RTP format in RFC 3984

Note: The terminal must remain operational when encountering codec offers for optional codecs other than H.263 Profile 0 Level 45.

RTCP Sender and Receiver Reports: RTCP sender and RTCP receiver reports shall be sent by the Video Share application.

RTP and RTCP Ports: Source and destination ports are set in UDP protocol header.

RTP/RTCP Rate Adaptation: Video sharing application shall have the ability to adapt to changing network conditions. Sending UE adjusts the media details to the changing network conditions using, for example, RTCP receiver reports and knowledge about the available network. Receiving UE must be able to adapt to changing bit rates within the limits negotiated in SIP/SDP.

TCP/IP Support: Header compression support is not required for Video Share.

An IMS authentication scheme must be supported (no special authentication of Video Share service as such). The authentication used should be independent of the set-up profile.

SigComp can be used, but it is not mandatory

Both PDP Always-on and PDP Per Call modes can be used. The Always-on method is preferred over the Per call method due to it decreasing the risk of SIP registration racing conditions and causing less radio access traffic load

- If PDP Per Call mode is used, SIP registration and PDP activation performed upon CS call

Both tel URI and SIP URI addressing schemes can be used

- Networks, terminals and Video Share application need to support tel URI addressing
- Terminals need to support tel URI / MSISDN carried within SIP OPTIONS
- It is assumed that UE B receives the Calling Line presentation in E.164 format

IPv4 will be used for Video Share (terminals might support also IPv6)

Video Share Session Termination: Either subscriber shall have the option to end the Video Share session and maintain the voice call. Upon termination of the host voice call, the Video Share session shall be terminated. The SIP:BYE message must be sent if a Video Share is dropped. Users should be able to share video, stop sharing, and restart sharing all within a single CS call. The re-initiation must include the whole setup procedure, including the SIP:INVITE and SIP:BYE.

*) GSMA Video Share over EDGE/DTM networks is not excluded. Implementations may use DTM class 5 or 9 or 11, and interactive or streaming class bearer (requires 2nd PDP context) for the video stream. DTM Class 11 gives higher video throughput

than DTM class 5 (around 60 kbps vs 30 kbps in average). Streaming class is recommended for the sake of QoS.

3.6 User Interface – Interaction with Other Services

Interaction with Call Waiting
When another CS call comes in to the Video Share sender or receiver, the user should have the ability to accept the call. If the user accepts the second call, the Video Share session should drop but both voice calls should remain up.

Three Way Voice Call Initiation
Video Share is only supported on 2-person calls. If a user places the call on hold or accepts another incoming call while sharing video, the voice call will remain but the Video Share session must be dropped.

Interaction with Multiparty Calling
No Video Share options shall be presented when a call is on hold, or while in a multi-party call. Replies to incoming INVITE messages shall be 486 or 603 Busy.

Call Hold Signaling
After a successful OPTIONS exchange, if the UE receives a Facility message that the other party has put the call on hold, the UI shall indicate that Video Share is not currently available. Once the UE receives a Facility message that the other party has resumed the call, UI functionality for Video Share shall be restored.

Transition from UMTS to GSM
When either device goes from UMTS or EDGE/DTM to GSM, the Video Share session shall be dropped.

Video Share When Outside UMTS Coverage

The Video Share capability negotiation shall be disabled when the device is on a GSM network. The user shall not be presented with an option to initiate or receive a Video Share call when on GSM. When the other mobile transitions to GSM, the Video Share connection shall drop and the voice connection shall be maintained.

4 INTERWORKING WITH LEGACY TERMINALS

Interworking with Video Share terminals based on legacy specifications (*Video Sharing 1.0, TurboCall, weShare 2.0*) should be supported. These terminals don't support all of the features described within the document, but anyway they support the actual Video Share service. When using these terminals the following restrictions can apply:

- No support for OPTIONS method (i.e. capability query not performed)
- No fallback support for IETF/3GPP mode (i.e. "legacy terminal" dictates the mode to be used and other terminal works according to that mode)
- No support for Feature Tag
- SDP sent also with PRACK & 200 OK
- SDP special application attribute, e.g. a=X-application:com.nokia.rtvs or a=application:com.nokia.rtvs, is set by these terminals

Therefore it should be taken into account that some of the steps shown above are optional, i.e. might not be always performed, depending on the used terminals. Note that restrictions apply specifically only to the already deployed legacy terminals, i.e. new terminals should not be designed using restrictions listed above.

5 DATA PARAMETERS

The following table shows parameters that are used by UE for Video Share.

Parameter	Purpose	Value
IMS APN		*<company specific>*
Streaming QOS Support	Indicate whether secondary PDP Context should be used	Optional
RTP Destination Port		Operator Specific
RTCP Destination Port		Operator Specific
SIP Destination Port		Operator Specific
Video Sharing Feature Tag	Indicates CS Voice Call + IMS Capability	+g.3gpp.cs-voice
P-CSCF	Domain name for P-CSCF address	*<company specific>*

6 SIP OPTIONS—MESSAGE CONTENT EXAMPLES

The following tables give examples of the SIP OPTIONS & 200 OK message contents as used with Video Share.

SIP OPTIONS (UE to Network)

Item	Field	Description
1	Accept	
2	Accept-contact	Include tag +g.3gpp.cs-voice
3	Accept-encoding	
4	Accept-language	
5	Call-ID	
6	Contact	Include tag +g.3gpp.cs-voice
7	Content-type	Application/SDP
8	Content-length	
9	Cseq	
10	From	
11	P-Preferred Identity	Public URI
12	Route	
13	Supported	QoS Tag
14	To	Tel URI or SIP URI of B party
15	User-agent	Could include version information
16	Via	

SIP OPTIONS (Network to UE)

Item	Field	Description
1	Accept	
2	Accept-contact	Include tag +g.3gpp.cs-voice
3	Accept-encoding	
4	Accept-language	
5	Allow	
6	Call-ID	
7	Contact	Include tag +g.3gpp.cs-voice
8	Content-deposition	
9	Content-encoding	
10	Content-language	
11	Content-type	Application/SDP
12	Content-length	
13	Cseq	
14	Date	
15	From	
16	MIME-Version	
17	P-Asserted Identity	Public URI
18	P-Asserted Identity	Tel URI or SIP URI
19	Supported	QoS Tag
20	Timestamp	
21	To	Tel URI or SIP URI of B party
22	User-agent	Could include version information
23	Via	

SIP—200 OK (UE to Network)

Item	Field	Description
1	Accept	
2	Allow	
3	Call-ID	
4	Contact	Include tag +g.3gpp.cs-voice
5	Content-type	Application/SDP
6	Content-length	
7	Cseq	
8	From	
9	Require	
10	Supported	QoS tag
11	To	Tel URI or SIP URI of A party
12	Via	

SIP—200 OK (Network to UE)

Item	Field	Description
1	Accept	
2	Allow	
3	Call-ID	
4	Contact	Include tag +g.3gpp.cs-voice
5	Content-type	
6	Content-length	
7	Cseq	
8	From	
9	Require	
10	Supported	QoS tag
11	To	Tel URI of A party
12	Via	

7 REFERENCES

- 3GPP TS 24.008: Mobile radio interface Layer 3 specification; Core network protocols; Stage 3
- 3GPP TS 24.229: Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3
- 3GPP TS 24.279: Combining Circuit Switched (CS) and IP Multimedia Subsystem (IMS) services; Stage 3
- 3GPP TS 26.141: IP Multimedia System (IMS) Messaging and Presence; Media formats and codecs
- IETF RFC 3312: Integration of Resource Management and Session Initiation Protocol (SIP)
- IETF RFC 3320: Signaling Compression (SigComp)
- IETF RFC 3550: RTP: A Transport Protocol for Real-Time Applications
- OMA Presence Simple enabler