Video Share Phase 2 Interoperability Specification
Version 3.0
16 January 2015

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

Security Classification: Non-confidential

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

Copyright Notice

Disclaimer

The GSM Association ("Association") makes no representation, warranty or undertaking (express or implied) with respect to and does not accept any responsibility for, and hereby disclaims liability for the accuracy or completeness or timeliness of the information contained in this document. The information contained in this document may be subject to change without prior notice.

Antitrust Notice

The information contain herein is in full compliance with the GSM Association's antitrust compliance policy.

Table of Contents

1   Introduction

1.1   Overview

This document is the Phase 2 Service Specification for 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 Phase 2 Service Definition* document which describes the Video Share service to aid GSMA Working Groups in their work on Video Share Phase 2. This document provides a more general introduction on Video Share Phase 2.

Work on Phase 2 of Video Share has built on the work completed in Phase 1. The Phase 1 Service Definition (SE.41) and Service Specification (IR.74) are referred to and built upon throughout this document. Phase 2 of Video Share is intended to be backwards compatible to Phase 1 (unless explicitly stated otherwise).

A number of aspects of the Video Share service have changed or been enhanced between Phase 1 and Phase 2. The two key changes are:

- Phase 1 was based on a Terminal interoperable Real-Time Live Video Share service allowing users to share live video between them over PS connection in real time simultaneously with ongoing CS call. In phase 2 the coupling between the Video Share session and the CS call has been broken, allowing Video Share to be initiated as a stand-alone service. Phase 2 Video Share sessions may still be launched in association with an on-going CS call, but the CS call is no longer essential.
- In Phase 1 Video Share uses P2P model, that is, applications are built in terminals thus a separate Application Server in network is not needed. In Phase 2 however, an VS-AS is used to enable enhancements to the service, such as Multiparty sessions, and to allow Video Sessions that cannot be established directly towards the terminating user to result in content being stored within the network and retrieved at a later date, using other mechanisms such as MMS or web-based download. In Phase 2, when Phase 1 Use Cases are being activated, the VS-AS is transparent to the signaling and media flow, but has to be included in the signaling path so that the enhancements described above can be invoked when required.

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

The Figure 1 gives the network architecture for Video share Phase2. It is based on the reference IMS Architecture described in 3GPP TS 23.228[21].

The VS-AS, that is, Video share Application Server and Presence Server are IMS Application Servers as defined in 3GPP TS 23.228[21].

Web-Portal and VS-Storage are not IMS Entities and are respectively described in Section 2.9.1 and Section 2.9.2 below.

IPX can take on the responsibilities to interconnect with other operator IMS networks (refer Section 4.3.3 in Video Share Phase2 Service Definition [1]).

The IMS Interfaces (Gm, Mb, Mp, ISC, Mx) shown in the Figure 1 are as specified in 3GPP TS 23.002[33] & 3GPP TS 23.228[21]. It is an Operator's choice to use IPv4 or IPv6.

The interfaces Wb1, Wb2, & Wb3 (for example, HTTP, RTSP based) are explained in Section 2.9.

Detailed specification of interfaces between Web-Portal, Video Storage and VS-AS are left outside of the scope of this document.
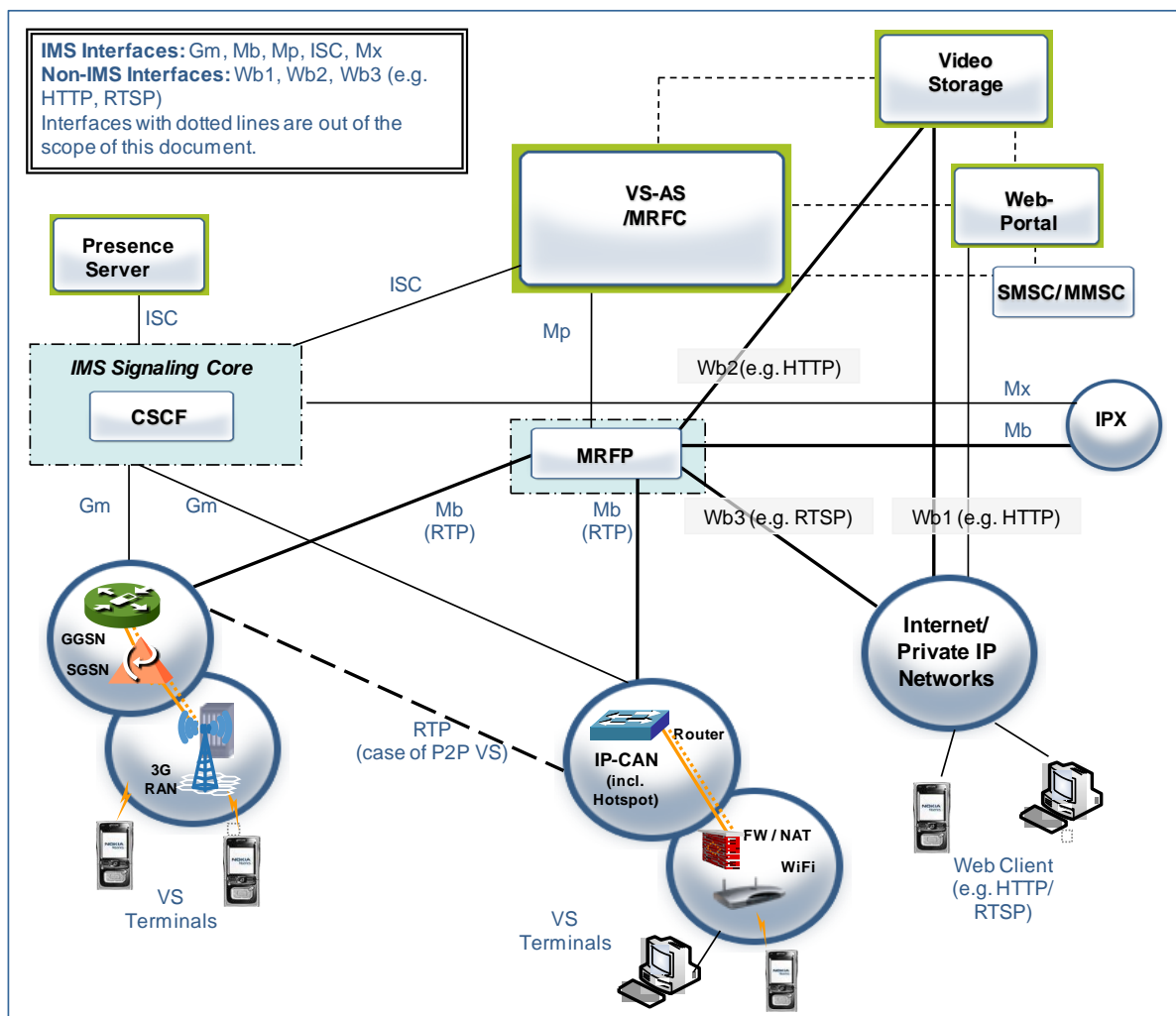

Figure 1: Network Architecture for Video Share Phase 2

In this figure, IPX is only represented as an illustration of a technical solution (among others) regarding interconnection of operator networks.

1.2   Scope

1.2.1   In Scope

The aim of this document is to present the technical principles for terminal interoperable Real-Time Live Video Share service for Phase 2, and describe backwards compatibility with the technical principles within Phase 1 as described in IR.74.

1.2.2   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, particularly

  - The implementation detail of the IM service used to launch Video Share sessions
  - The implementation of the Presence service used in Presence-based Capability Exchange

- PSTN related issues
- Commercial issues
- Back-office functions (for example, O&M)
- High availability

1.3   Definition of Terms

Term	Description
UE	User Equipment as defined in 3GPP TS 21.905
VS-AS	Video Share Application Server is an IMS Application Server
VS Device	Video Share Device is any device which contains an IMS Client which support Video Share IMS Application Service and can connect to an IMS network. It encompasses both UE's and terminals such as a PC or laptop connected to the IMS network via an IP-CAN
VS-Storage-UC-URI:	It is a provisioned SIP URI to be used in the Request-URI of the SIP INVITE while initiating the VS Storage Use case
Conference-Factory-URI	It is a provisioned SIP URI to be used in the Request-URI of the SIP INVITE while initiating Point to Multipoint VS Session
CS	Circuit Switch
CDR	Call Data Record
MRF	Media Resource Function
MRFP	Media Resource Function processor
MRFC	Media Resource Function Controller
CSI	Combination of CS & IMS Services
ICSI	IMS Communication Service Identifier
IARI	IMS Application Reference Identifier
PS	Packet Switch
QoS	Quality of Service
IP-CAN	IP Connectivity Access Network: It represents any network that can

Term	Description
	provide access to the IMS (for example: GPRS, I-WLAN, and so on). Details on some IP-CANs can be found in 3GPP TS 24.229

1.4   Document Cross-References

Document	Name
[1] GSMA PRD SE.50 v1.1	Video Share Phase 2 Service Definition
[2] GSMA PRD SE.41 v2.0	Video Share Service Definition
[3] GSMA PRD IR.74 v1.0	Video Share Interoperability Specification
[4] RFC 3312	Integration of Resource Management and Session Initiation Protocol (SIP)
[5] RFC 4032	Update to the Session Initiation Protocol (SIP) Preconditions Framework
[6] RFC 3311	The Session Initiation Protocol (SIP) UPDATE Method
[7] draft-ietf-sip-uri-list-conferencing-02	Conference Establishment Using Request-Contained Lists in the Session Initiation Protocol (SIP)
[8] draft-ietf-sip-multiple-refer-02	Referring to Multiple Resources in the Session Initiation Protocol (SIP)
[9] RFC 4579	SIP Call Control – Conferencing for User Agents
[10] RFC 3265	(SIP)-Specific Event Notification
[11] RFC 4575	Session Initiation Protocol (SIP) Event Package for Conference State
[12] RFC 3515	The Session Initiation Protocol (SIP) Refer Method
[13] RFC 3551	RTP Profile for Audio and Video Conferences with Minimal Control
[14] RFC 4566	SDP: Session Description Protocol
[15] RFC 4855	Media Type Registration of RTP Payload Formats
[16] RFC 4867	RTP Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs
[17] RFC 3555	MIME Type Registration of RTP Payload Formats
[18] RFC 3550	RTP: A Transport Protocol for Real-Time Applications
[19] RFC 3261	Session Initiation Protocol
[20] 3GPP TS 24.229	Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP)
[21] 3GPP TS 23.228	IP Multimedia Subsystem (IMS)
[22] 3GPP TS 24.147	Conferencing using the IP Multimedia (IM) Core Network (CN) subsystem
[23] 3GPP TS 32.260	Telecommunication management; Charging management; IP Multimedia Subsystem (IMS) charging
[24] 3GPP TS 32.275	Telecommunication management; Charging management; MultiMedia Telephony (MMTel) charging
[25] 3GPP TS 32.297	Telecommunication management; Charging management; Charging Data Record (CDR) file format and transfer
[26] 3GPP TS 32.298	Telecommunication management; Charging management; Charging Data Record (CDR) parameter description
[27] 3GPP TS 24.410	Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); NGN Signalling Control Protocol; Communication HOLD (HOLD) PSTN/ISDN simulation services

Document	Name
[28] 3GPP TS 24.008	Mobile radio interface Layer 3 specification
[29] 3GPP TS 20.060	GPRS Tunnelling Protocol (GTP) across the Gn and Gp interface
[30] 3GPP TS 29.207	Policy control over Go interface
[31] 3GPP TS 29.208	End-to-end Quality of Service (QoS) signalling flows
[32] 3GPP TS 23.107	Quality of Service (QoS) concept and architecture
[33] 3GPP TS 23.002	IMS Network Architecture
[34] 3GPP TS 22.279	Combined Circuit Switched (CS) and IP Multimedia Subsystem (IMS) sessions
[35] 3GPP TS 24.279	Combining Circuit Switched (CS) and IP Multimedia Subsystem (IMS) services
[36] 3GPP TS 24.173	IMS multimedia telephony communication service and supplementary services
[37] OMA-TS-Presence_SIMPLE-V1_0	OMA Presence SIMPLE Specification
[38] RFC 3267	RTP Payload Format for AMR and AMR-WB
[39] RFC 3264	An Offer/Answer Model Session Description Protocol
[40] RFC 4629	RTP Payload Format for ITU-T Rec. H.263 Video
[41] RFC 3016	RTP Payload Format for MPEG-4 Audio/Visual Streams
[42] RFC 3984	RTP Payload Format for H.264 Video

2 Video Share Phase 2 – Technical Detail

2.1 Video Share Phase 1 - RECAP

It is important in defining the Phase 2 Service Specification to be aware of the key technical points from IR.74 and Phase 1 in general. These are summarized below.

- Feature tag *+g.3gpp.cs-voice* used within the Accept-Contact and Contact fields of SIP methods. This is the only identifier used in the service – IMS domain uses content of SIP and SDP messages to determine that the service is Video Share.
- IMS is not applying any Service specific logic. IMS involvement is limited to Authentication of the subscriber, routing of SIP signaling and CDR generation.
- Capability exchange is achieved using SIP OPTIONS method and 200 OK.
- Two session establishment mechanisms (to be both mandatory supported from VS Device side) are defined – 'IETF mode' based on RFC3261, and 'IMS mode' based on the use of Pre-conditions as defined in 3GPP TS 24.229[20].
- H.263 Option 0 Level 45 was only mandatory codec.
- Use of both 'Always On' and 'On Demand' PDP contexts for transmission of the media plane were permitted, with preference for Always On.
- Minimum bandwidth of 64kbps for the RAB
- Backwards compatibility with 'pre-Phase 1' vendor specific terminal implementations was supported using specified SDP.
- Tel URI and SIP URI addressing schemes are supported
- A Video Share session consisted of the following steps:
- CS call setup
- Capability query
- Invitation procedure (SIP)
- Video transmission (RTP)

- Teardown of video session
- Teardown of CS call

For further detail on the technical aspects of these points, refer to IR.74 [3]

2.2   General

Whilst in Phase 1, the Video Share session was tied to a specific, on-going CS call, in Phase 2 that association is broken. As a result, the formulaic sequence that a session goes through from CS call establishment to termination of Video Share and CS call is no longer applicable in Phase 2. Sessions that are established within as 'additional enhancements' to an ongoing CS call do still follow this sequence, but the ability to initiate a Video Share session as a standalone service or from some other service is added.

The Video Share service has always been technically independent of any other service from within which it is launched. Integration of the services is a matter for the GUI on the device, but the service initiation for Video Share remains equivalent to the establishment of a new session completely independent of the associated service. Thus in Phase 2, where Video Share can be launched from an IM session, the IM session remains independent from the Video Share session apart from in the User Device. When the IM session is terminated, if the Video Share session is still active, this is also terminated in the same way as for a Video Share session associated with a CS call, but the device must trigger the two terminations separately as well.

In Phase1, a Video Share Session is always associated with a CS Call hence the service is always a CSI Combinational Call as defined in 3GPP TS 22.279[34].This is still possible in Phase2 as well. But in Phase2, it is also possible to perform a Video Share session without existence of a CS call. Therefore, in this context it is important to note here that if there is already an ongoing Video Share Session and subsequently a CS call is setup between the same two end-points then the CS call would exist independently and no association is done with the ongoing IMS Video Share Session. In other words, it is not considered as a Combinational Session (refer to definition in 3GPP TS 22.279[34]) in the scope of VS Phase2 Service.

In Phase2, a CS voice call followed by a Video Share session is a CSI, but vice versa is not. This means that when the Client is in a state where both the CS voice call and Video Share session are active, and the CS call is subsequently disconnected, the Client behaviour will be different based on the order in which the CS voice and VS session are set up. If the CS call is established first, followed by a VS session, then the entire session will be disconnected when the CS call is disconnected. However, if the VS session is set up prior to CS call being established, the VS session will remain intact if the CS call is disconnected (and vice versa).

Besides the case in Figure 2 , 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, for example, due to the other terminal has made a handover to a non-DTM 2G access during the video transmission phase. See RFC 3550[18] for further details of this timeout functionality.
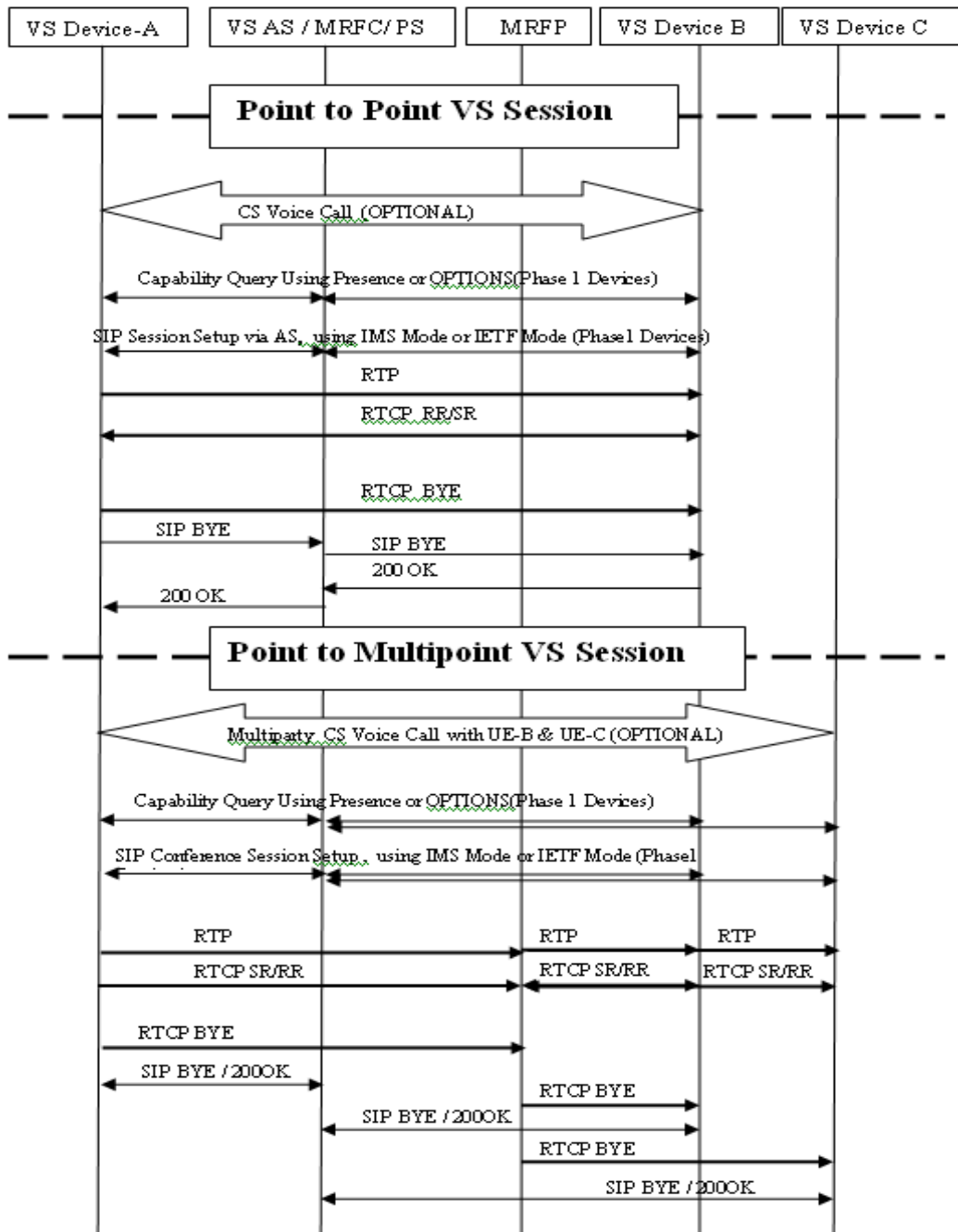
Figure 2: General Flows in Video Share Phase 2

2.3   Service Identification

2.3.1   Feature Tag

In the IR.74 [3] section 3.2, the use of CSI Feature Tag for identification of the Phase 1
Video Share service was explained and described. This usage remains unaltered as the
mechanism to identify Phase 1 compliancy for the Video Share service. The Feature Tag
used is g.3gpp.cs-*voice.* The g.3gpp.cs-voice feature tag is defined in 3GPP TS 24.279[35]
and used by a terminal that supports voice in a circuit switched environment within the
context of combining a circuit switched voice call with an IMS session.

Phase 2 VS Devices must also continue to use Feature Tag as described in IR.74 [3] to allow Phase 1 Use cases to be supported when initiated or terminated by Phase 1 terminals.

2.3.2   3GPP IMS Service and Application Identifiers

Between the development of Phase 1 Video Share and Phase 2, 3GPP has defined Service and Application Identifiers to be used for IMS services. These are the IMS Communication Service Identifier (ICSI) and IMS Application Reference Identifier (IARI) respectively. The definition of the ICSI and IARI is found in 3GPP TS 24.229[20] in sub clause 7.2A.8 and 7.2A.9 respectively. These identifiers can be used by both Terminal and Network. Terminals can use the identifiers to indicate their capability to the network. The identifiers can also be included in CDRs and inter-operator agreements as a part of service based agreement and charging framework across operators.

The Combinational Services Feature Tag alone is not sufficient to identify the different Use cases defined in the Service definition document. Hence, the ICSI and IARI must be used to distinctly identify the Phase2 Use cases. (Note that the Feature Tag for Combinational Services is also still used both to indicate that aspect of the overall service and for backwards compatibility with Phase 1)

The Videoshare ICSI is

urn-7:3gpp-service:ims.icsi.gsma.videoshare

Video Share also must use an IARI to explicitly identify what application is being used. The IARI to be used for Video Share Phase 2 must be *urn-7:3gpp-application.ims.iari.gsma-vs.*

The mechanism to be used for to identify various Video Share Phase2 Use cases are as follows

- VS with CS Voice Call

The set of ICSI, IARI & CSI Feature tag is used for identification.

          - ICSI to be used → urn-7:3gpp-service:ims.icsi.gsma.videoshare
          - IARI to be used → urn-7:3gpp-application.ims.iari.gsma-vs
          - CSI Feature Tag → g.3gpp.cs-voice
[This Feature tag is to be used in Conjunction with the ICSI & IARI mentioned above to indicate Combinational Service. It also serves for Interoperability with Phase1 terminals]

- Point to Multipoint VS with CS Voice Call (U2.1 in [1])
- The set of ICSI, IARI & CSI Feature tag mentioned for "VS with CS Voice call" above is to be used for Identification. In addition to this, usage of a Conference Factory URI in Request-URI SIP header must indicate request for a CSI Point to Multipoint VS. The Conference-Factory-URI is provisioned by the Operator.
- VS without CS Voice Call (U3.1 in [1])

     The set of ICSI and IARI is used for identification.
          - ICSI → urn-7:3gpp-service:ims.icsi.gsma.videoshare
          - IARI → urn-7:3gpp-application.ims.iari.gsma-vs
- Point to Multipoint VS without CS Voice Call (U2.2 in [1])

- The Combination of ICSI, IARI for "VS without CS Voice call" above is to be used for Identification. In addition to this, usage of a Conference-Factory-URI in Request-URI SIP header must indicate request for a Point to Multipoint VS. The Conference-Factory-URI is provisioned by the Operator.
- P2P VS with Video Sourced from Video Storage

This use case is to be identified using the following:

- Existence of combination of ICSI, IARI & CSI Feature Tag for "VS without CS Voice call" OR "VS with CS Voice call", as applicable
- Existence of the media level SDP attribute X-vrsrc-id
- Directionality of media to be recvonly i.e a=recvonly
- Existence of remote VS Device's URI in the Request-URI SIP header

- VS with Storage Locker (U3.2 in [1])

The set of ICSI, IARI mentioned for "VS without CS Voice call" above is to be used for Identification. A special SIP URI, the VS-Storage-UC-URI is to be used in the request-URI of the INVITE, which is used to identify this Usecase. This URI is provisioned by the Operator.

Optionally the session-level SDP attribute *a= X-vrsrc-id*' (see Section 2.6.1.1) will carry the suggested/preferred file name.

ICSI and IARI are coded as URNs and included into the Accept-Contact and Contact SIP headers by using the two 3GPP defined media feature tags: *g.3gpp.icsi-ref and g.3gpp.iari-ref, respectively*. Note that any colons in URN are replaced by "%3A" when included in the media feature tags. For example:

+g.3gpp.iari-ref="urn%3Aurn-7%3A3gpp-application.ims.iari.gsma-vs"

+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.gsma.videoshare"

The following SIP messages carry these service identifiers:

- INVITE (in Accept-Contact and Contact header) and subsequent 200 OK (in Contact header)
- REGISTER (in Contact header, handling of service identifier in REGISTER method in the network is optional) and subsequent 200 OK (in Contact header)
- OPTIONS (in Contact header of 200OK)

For the purpose of service identification towards the network, the ICSI must also be included in the P-Preferred-Service Header in SIP methods sent by the VS Device towards the IMS network and that are related to the Video Share Phase 2 service.

The network must also include the ICSI in the P-Asserted-Service header, either by checking the contents of the P-Preferred-Service (if present) against the subscribed services and then transposing the content of P-Preferred-Service into P-Asserted-Service, or if P-Preferred-Service is not included in the method, by asserting the ICSI for the session and including it in P-Asserted-Service. P-Asserted-Service is stripped before the SIP session hop to the

terminating party. The terminating VS Device will only receive ICSI (as well as IARI and CSI Feature Tag) in Accept-Contact and/or Contact, if present.

2.3.3   <service-description> for Presence-based Capability Exchange

As will be described in Section 2.4, in Phase 2, realization of Capability Query using the OMA Presence based mechanisms is introduced. The OMA-TS-Presence_SIMPLE-V1_0 specification [37] specifies a number of different Presence-based use cases, including that for Capability Exchange. Within the OMA reference document this feature is referred to as 'Application-specific Availability'.

Presence information within OMA-TS-Presence_SIMPLE-V1_0[37] is catalogued in a 'document' within which a field - <service-description> - is defined. <service-description> is made up of <service-id> and <version> fields, and valid <service-description> values can be found at http://www.openmobilealliance.org/tech/omna/omna-prs-PidfSvcDesc-registry.htm

It is proposed that Video Share be allocated a <service-description> value for use in Presence-based Capability Exchange (although once the value is defined it can be used also in any of the other Presence services described by OMA that include <service-description> within their semantic). Video Share Capability must be indicated by the inclusion of <service-description> with <service-id> set to "*org.gsma.videoshare"* and <version> set to *2.0*. Note that these values can only be used to indicate support of Video Share Phase 2. It would be reasonable to allocate a value to Video Share Phase 1 capability, but this would be useless since Presence-based Capability Exchange is not defined for Phase 1 and so the value would never be used. Video Share Phase 1 support is indicated by;-

- non-support of Presence-based Capability Exchange; and
- OPTIONS-based capability exchange without inclusion of ICSI and/or IARI as described in 4.2.2 above.
- Existence of *g.3gpp.cs-voice* CSI Feature Tag(3GPP TS 24.279[35]) in Accept-Contact & Contact headers

For more detail, see Section 2.5 below.

2.4   Ability to trigger/invoke Video Share Application Server (as defined in IR.84) from Video Share Phase 1 capable devices

An operator can be interested in including a Video Share Application Server in the path even in the case of Phase 1 VS Devices (that is IR.74 compliant devices). As an example, the Video Share Application Server could be used to interwork Videoshare terminals (either Phase 1 or Phase 2) towards legacy terminals.

For those cases, Video Share Application Server shall be able to accept messages from Video Share phase 1 capable devices (IR.74).

2.5   Capability Query

2.5.1   Capability Exchange based on OMA Presence

The OMA Presence Framework as specified in OMA-TS-Presence_SIMPLE-V1_0[37] is recommended to be used for Capability Query operation by Phase 2 VS Devices. However,

Phase2 VS Devices can optionally fall back to using OPTION exchange mechanism as explained in Section 2.5.2.

The Presence Framework facilitates Capability Query as follows:

1. A VS Phase 2 VS Device plays the role of Presence Source and publishes its Presence Information (PIDF). In the PIDF, Video Share Capability must be indicated by the inclusion of <service-description> with <service-id> set to "*org.gsma.videoshare"* and <version> set to 2.0.
2. The Presence AS receives and stores the Presence information and serves the information on request.
3. A VS Phase 2 VS Device plays the role of a Watcher (either fetcher or subscriber) to fetch the Presence information from the Presence AS in order to obtain the VS Capability of the remote.

It is necessary for the VS Phase 2 VS Device to support SUBSCRIBE, NOTIFY and PUBLISH SIP methods to achieve (a),(b) and (c) above as specified in OMA-TS-Presence_SIMPLE-V1_0[37].

The call flow in Figure 3 depicts (a) & (b) and Figure 4 depicts (c).

It is imperative that a Phase 2 VS Device will have done a PUBLISH of its Presence information for Capability Query operation by another Phase 2 VS Device to be successful. It is recommended that the VS Device performs a PUBLISH when

- IMS Registration succeeded, indicating its availability
- it is no longer available/willing(for example already engaged in a VS Session), indicate its non-availability
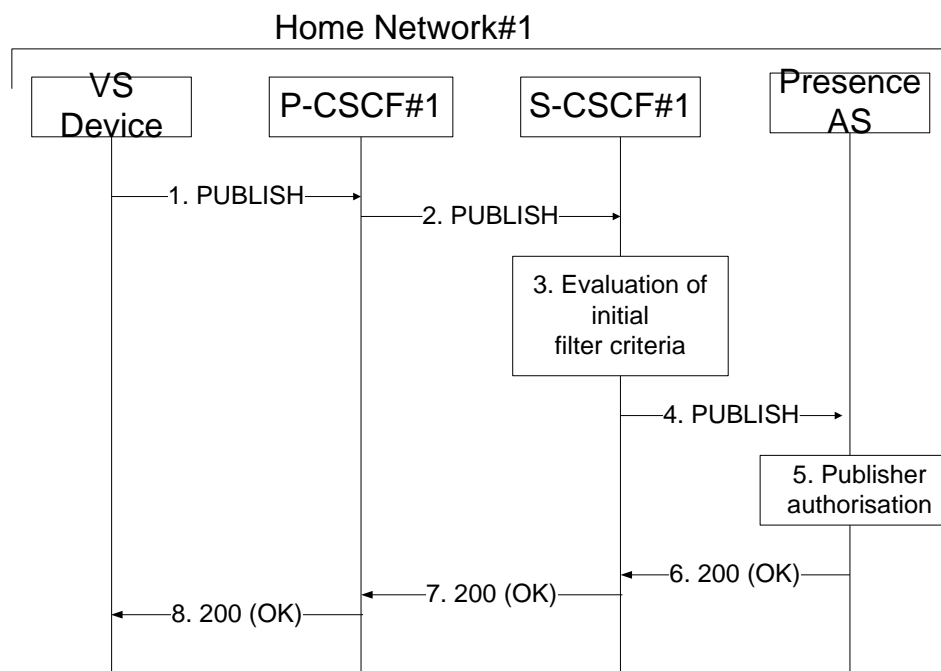- it becomes available/willing , indicating its availability/willingness

Figure 3: VS Device publishing Presence information including VS Capability info

The VS Phase 2 VS Device must support the role of a FETCHER. It can optionally support the role of SUBSCRIBER. A FETCHER simply requests the current value of the Presence Information whereas a SUBSCRIBER requests notification from the Presence Server whenever there is a change in the Presence Information. Naturally, the latter introduces more traffic than the former.

Since VS Phase 2 allows for VS in absence of CS call also, it is not required that the SUBSCRIBE is done on CS Call setup, it can be done any time.

The implementation of the Presence service within the network is out of scope of this document.
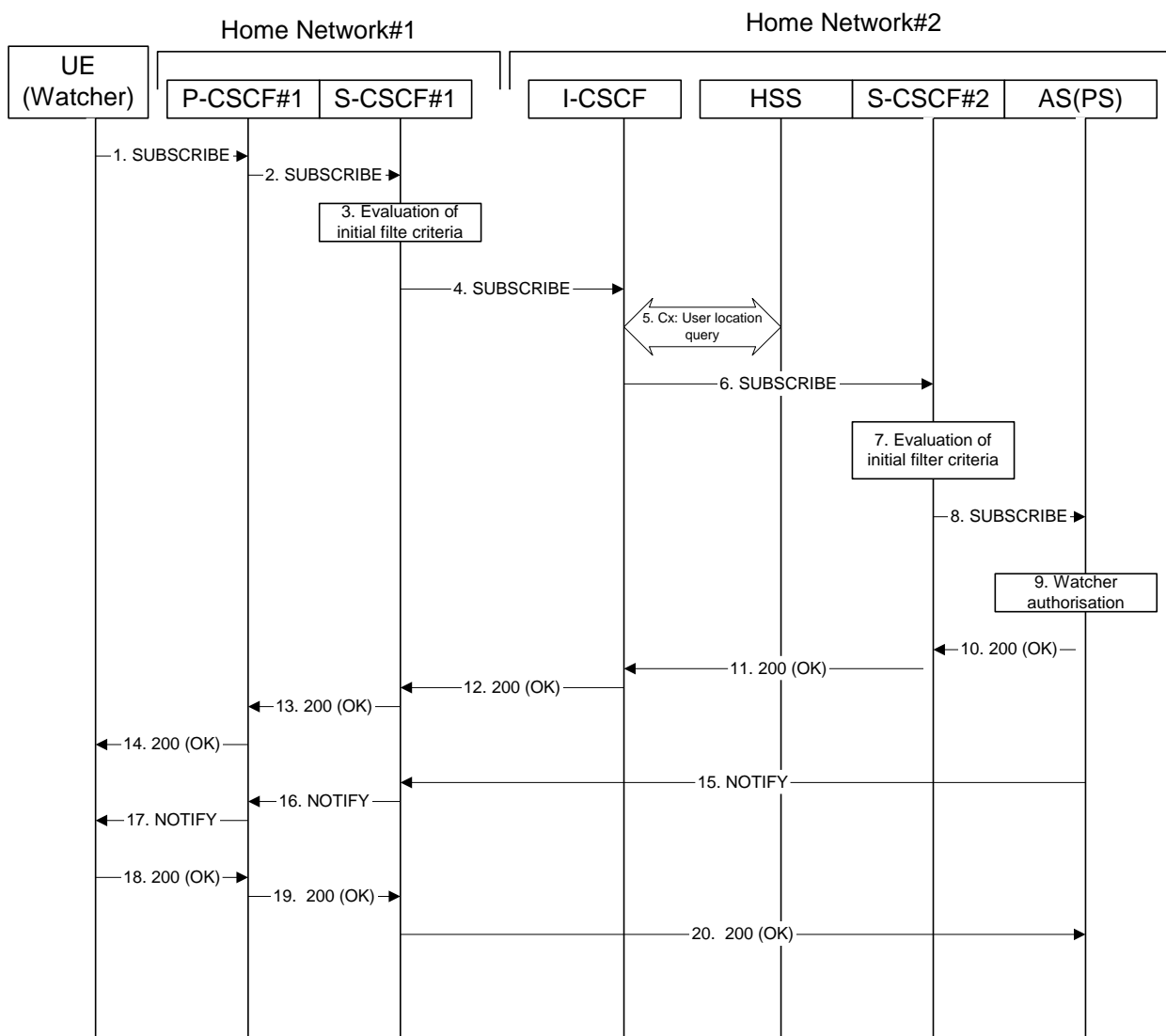
Figure 4: VS Device subscribing & fetching Presence information containing VS Capability info

2.5.2   Fall back to OPTIONS based Capability Query

The Capability Query operation between Phase 2 VS Devices is recommended to be performed using Presence as explained in Section 2.5.1. However Phase2 VS Devices must fall back to using OPTION exchange mechanism specified in Section 2.5.3 below, in case the SUBSCRIBE request fails. This can happen when the IMS Core does not support Presence or in case where remote VS Device does not support Presence.(for example, Phase1 VS Device).

Following are the scenarios wherein fall back to OPTIONS exchange occurs:

1. Phase1 VS Device initiating CSI VS towards Phase2 VS Device.The Capability Query must be same as that specified in Section 2.5.3 below

2. Phase2 VS Device initiating Capability Query VS towards Phase1 or Phase2 VS Device that does not support Presence. The Phase 2 VS Device initiates SUBSCRIBE as described in Section 2.5.1. The <service-description> for VS will be

absent in the Presence Information received. As a result the Phase 2 VS Device must fallback to OPTIONS exchange mechanism specified in Section 2.5.3 and initiate OPTIONS.

3. Phase2 VS Device is in a IMS Network that doesn't support Presence and initiates Capability Query. The Phase 2 VS Device initiates SUBSCRIBE as described in Section 2.5.1. The IMS Core would return a failure response since there is no Presence Server. As a result the Phase 2 VS Device must fallback to OPTIONS exchange mechanism specified in Section 2.5.3 and initiate OPTIONS.

In the scenario wherein a Phase2 VS Device initiates a Capability Query, without engaging in CS call, towards a Phase1 VS Device, the <service-description> for VS will be absent in the Presence Info received by Phase2 VS Device since the remote being a Phase1 VS Device would not have Published corresponding Service Description.

As a result the Phase 2 VS Device must fallback to OPTIONS exchange mechanism specified in Section 2.4.3 and initiate OPTIONS. But since remote is a Phase1 VS Device it must respond with a failure response since CS call does not exist.

Consequently Phase 2 VS Device deduces that remote is a Phase1 VS Device and does not proceed further since Phase1 VS Devices do not support VS without CS call.

2.5.3   OPTIONS Exchange

The OPTIONS exchange mechanism described in specified in IR.74 [3] Section 3.3 is to be used with changes specified here in this section.

- The Contact header in the 200OK (OPTIONS) must contain the two feature tags g.3gpp.cs-voice and g.3gpp.iari-ref. The value of g.3gpp.iari-ref must be the IARI urn-7:3gpp-application.ims.iari.gsma-vs indicating the support of Video Share Phase2. The existence of only g.3gpp.cs-voice can imply that only Video Share Phase1 is supported.
- The 200OK (OPTIONS) response can optionally contain m-lines describing the video stream supported.
- The usage of a=application attribute specified in IR.74 is deprecated.

2.6   Session Setup

2.6.1   P2P VS Session

The VS AS plays the role of a B2BUA during setup of SIP Session between VS Devices. The signaling happens always via the VS-AS for Session Setup for both, VS Session between Phase2 VS Devices and between Phase1, Phase2 VS Devices. However the Media Path must be directly between the VS Devices and not via the VS-AS.

The Phase 2 VS Device's and the VS-AS must support, both, the IETF and the IMS Mode signaling, for SIP Session setup. However the Session setup between Phase2 VS Devices (via VS-AS) must follow IMS Mode. The Session setup between Phase1 and Phase2 VS Devices via VS-AS) can use either of the modes depending on what the Phase1 VS Device supports.

The Figure 5 below shows Session Setup between two Phase2 or Phase1 VS Devices supporting IMS mode signalling.
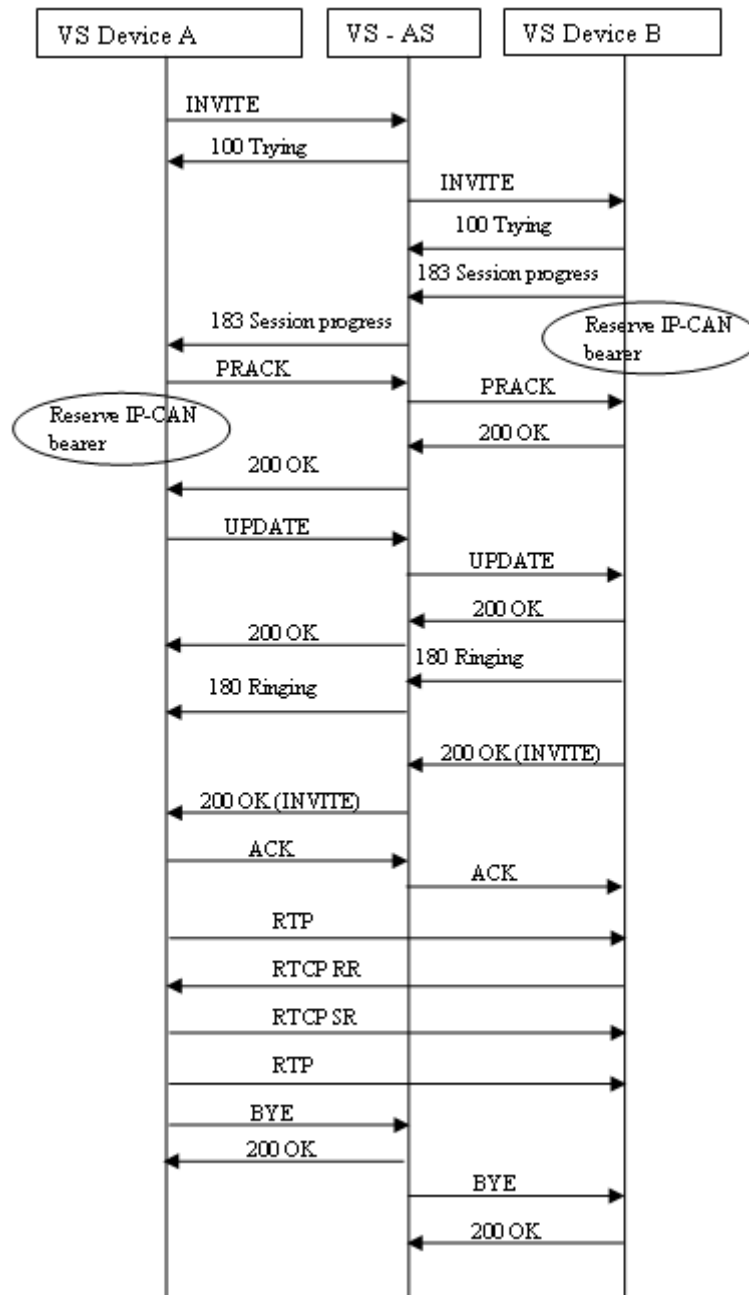
Figure 5: P2P VS Session Signaling : IMS mode

A VS Phase2 VS Device indicates the mode of signaling as IMS mode of signaling by indicating that it supports precondition and that it supports reliable provisional responses. This done by including "precondition" and "100Rel" option-tags in the *Supported* header of the SIP INVITE. The usage of precondition and reliable provisional responses during session setup is as described in 3GPP TS 24.229 [20] Section 5.1.4 & 5.1.3. Support for pre-condition mechanism implicitly mandates support for SIP UPDATE as defined in RFC 3311 [6].

The inclusion of QoS pre-condition and bandwidth modifier ("b=AS") in SDP must be done as specified in 3GPP TS 24.229 [20] Section 6.1.1 & 6.1.2. An example SDP is described further below in this section.

Typically, the INVITE must contain the offer and 183 response must carry the answer. In this offer/answer exchange it is indicated that the media is inactive since resources may not be available yet.

The PRACK and corresponding 200OK need not carry SDP as the final codec decision is already made as part of the initial offer/answer exchange. No behavior is described here if it exists.

Using the UPDATE/200OK exchange, the peers indicate each other that it can send/ receive media as the necessary resources are available.

The 180(Ringing) does not have a SDP therefore need not be sent reliably that is PRACK/200OK is not required for this provisional response.

The 200OK (INVITE) starts off the media flow for the Session.
If VS DEVICE-B receives QoS Pre-Condition SDP attributes in INVITE indicating that VS DEVICE-A has already QoS met at its end (for example: a=curr:qos local send) and if VS DEVICE-B is Phase2 or Phase1 terminal supporting Pre-Condition mechanism, then the signaling will be same as in the Figure 6 except that 200OK(INVITE) will carry pre-condition SDP attributes as part of VS DEVICE-B's answer. It must be noted that the 180 Ringing must carry the pre-condition option tag in this case.

If a Phase1 VS Device supports IMS mode of signalling, the SIP session setup sequence is same as illustrated in Figure 5(or optimized as Figure 6 as mentioned above). Otherwise the Phase2 VS Device must fallback to IETF mode of signalling as shown in Figure 6. If a Phase2 VS Device has initiated the INVITE then it must fallback if it receives a provisional response with precondition and 100rel option tags missing in the Supported header. If the Phase1 VS Device has initiated the INVITE, then Phase2 learns that IETF mode of signalling to be done by noticing the missing option-tags in the Supported header of the received INVITE.
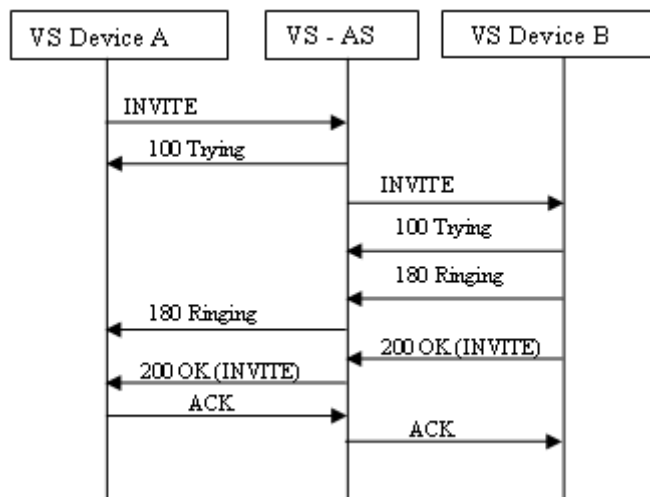
Figure 6: P2P VS Session Signalling : IETF mode

Because IMS Authentication is used regardless of whether IETF-mode or IMS-mode for INVITE is being employed, VS DEVICE's operating in either mode must include *Security-Verify according to the security mechanism used over the IP-CAN (for example: ipsec-3gpp* to indicate the IPSec Security Association between VS DEVICE and P-CSCF, for an access over 3g-RAN).

The change of mode from IMS mode to IETF mode must be the only mode change allowed (it is not possible for a VS Phase2 VS Device to respond to an IETF mode INVITE with an IMS mode 100 Trying followed by a 183 Session Progress).

The Phase2 Service Definition has Use cases wherein Audio is also carried over PS. Suitable Media Description for Audio stream is to be included in SDP thus causing setup of RTP streams to carry the Audio. An example media description in SDP is illustrated in Section 2.6.1.1 below.

Following table illustrates the possible cases wherein Audio also is carried over RTP:

	Type of Video	
	Live Video Share	**Video Clip with Audio**
Video Share with CS call	Ambient Audio sent over CS (no Audio over PS)	Ambient Audio is sent over CS Audio in Video Clip is sent over PS along with Video (separate RTP Session) The Receiver has the option to decline the PS Audio Stream, as part of SDP Offer/Answer signaling, and receive only the Video stream
Video share without CS call	Ambient Audio sent over PS along with Video (separate RTP Session)	Audio Clip sent over PS along with Video (separate RTP Session)

Table 1: Cases wherein Audio also is carried over RTP

The PS Audio and Video streams are transmitted over separate RTP Sessions and synchronized using the NT and RTP Timestamps as specified in RFC 3550 [18].

2.6.1.1   SIP Headers and SDP

Accept-Contact, Contact and SDP in Session Setup:

The values to be carried in the Accept-Contact and Contact SIP headers for VS Phase2 is as follows:

*Accept-Contact: * ;+g.3gpp.cs-voice; explicit*

*Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A   3gpp-service:ims.icsi.gsma.videoshare"*

*Accept-Contact: *;+g.3gpp.iari-ref="urn%3Aurn-7%3A 3gpp-application.ims.iari.gsma-vs"*

Contact: <contact addr>;+g.3gpp.cs-voice;+g.3gpp.icsi-ref="urn%3Aurn-7%3A 3gpp-service:ims.icsi.gsma.videoshare; +g.3gpp.iari-ref="urn%3Aurn-7%3A 3gpp-application.ims.iari.gsma-vs"

In case of IMS mode signaling Supported and Allow header as follows:

Supported: precondition, 100Rel

Allow: UPDATE

The INVITE message can contain more standard headers than the ones explicitly

mentioned here. 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.

Following table details the SDP attributes used in the Offer/Answer's during Session setup:

Attribute	Offer SDP	Answer SDP
m-line	Mandatory	Mandatory
b-line	Mandatory	Mandatory
a=framesize	Optional	Optional
a=framerate	Optional (multiple allowed)	Optional (multiple allowed)
a=rtpmap	Mandatory (multiple allowed)	Mandatory (multiple allowed)
a=fmtp	Optional (multiple allowed)	Mandatory (multiple allowed)
a=recvonly	Not Applicable	Mandatory
a=sendonly	Mandatory	Not Applicable
a=X-vsrc-type	Optional	Optional
a=X-vrsrc-id	Mandatory only for VS Session with video sourced from Video Storage	Mandatory only for VS Session with Video Storage and VS session with non-VS capable terminals
a=maxptime	Optional	Mandatory
a=ptime	Optional	Optional
a=curr	Mandatory (applicable only for IMS Mode)	Mandatory (applicable only for IMS Mode)
a=des	Mandatory (applicable only for IMS Mode)	Mandatory (applicable only for IMS Mode)
a=conf	Not Applicable	Mandatory

Table 2: SDP Attributes for VS Phase2

The SDP attributes related to Audio/Video encoding and the RTP packetization are discussed in Section 2.7 and Section 2.8 below.

Note that the examples of SDP discussed below are informative while the Table -2 content specifying the various SDP attributes is normative.

An (informative) example of SDP sent from a Phase 2 terminal as an offer is given below;-

> **Informative Example: 1**
> *m=video portUE-A RTP/AVP 98 96*
> *a=sendonly*
> *a=X-vsrc-type:videolive*
> *a=rtpmap:98 H264/90000*
> *a=fmtp:98 profile-level-id=42C00D; packetization-mode=0*
> *a=rtpmap:96 H263-2000/90000*
> *a=framesize:96 176-144*
> *a=framerate:8*
> *a=fmtp:96 profile=0; level=45*
> *b=AS:54*
> *a=curr:qos local none*
> *a=curr:qos remote none*
> *a=des:qos mandatory local send*
> *a=des:qos none remote send*

The *a=curr:qos and a=des:qos* attributes are related to QoS preconditions. This line will be absent in case of IETF mode of signaling.

In this SDP, support of H.263 Profile 0 Level 45 is indicated. As mentioned this implies fall back support also for H.263 Profile 0 Level 10.

In this SDP, support and preference for H.264 baseline profile level 1.3 is indicated. See also Section 2.7.

An example of SDP answer is given below:

> **Informative Example : 2**
> *m=video portUE-B RTP/AVP 98 96*
> *a=recvonly*
> *a=X-vsrc-type:videolive*
> *a=rtpmap:96 H263-2000/90000*
> *a=framesize:96 176-144*
> *a=framerate:8*
> *a=fmtp:96 profile=0; level=45*
> *a=rtpmap:98 H264/90000*
> *a=fmtp:98 profile-level-id=42C00D; packetization-mode=0*
> *b=AS:800*
> *a=curr:qos local none*

> *a=curr:qos remote none*
> *a=des:qos mandatory remote recv*
> *a=des:qos mandatory local recv*
> *a=conf:qos remote recv*

The attribute *a=curr:qos* is related to QoS preconditions that are mandatory in 3GPP; *a=conf:qos* remote *recv* indicates a willingness to get confirmation of resource reservation. (Confirmation is done by SIP UPDATE).The b=AS:54 indicates the bandwidth at which the transmission will occur (here being 54kbps). These lines will be absent in case of IETF mode of signaling.

It is to be noted that the media can be set inactive by inclusion of "a=inactive" in the offer-answer during in INVITE and 183(Session progress).

If the VS Device sending the offer also supports, for example, MPEG4, the SDP in the offer would contain additional "a=" lines as follows:

> **Informative Example: 3**
> *m=video portUE-A RTP/AVP 96 97*
> *a=sendonly*
> *a=rtpmap:96 H263-2000/90000*
> *a=framesize:96 176-144*
> *a=framerate:8*
> *a=fmtp:96 profile=0; level=45*
> *a=rtpmap:97 MP4V-ES*
> *a=framesize:97 176-144*

A Phase 1 terminal, VS Device B, supporting only Profile 0 Level 10 of H.263 would send an answer to either of these offers with SDP as below:

> **Informative Example: 4**
> *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=10*

The a=fmtp line indicates the downgraded H.263 support, whilst b=AS:54 indicates the bandwidth at which the transmission will occur (here being 54kbps).

The SDP answer from a Phase 2 VS Device supporting MPEG4 might look as follows:

> **Informative Example: 5**
> *m=video portUE-B RTP/AVP 96 97*
> *b=AS:116*
> *a=recvonly*
> *a=rtpmap:96 H263-2000/90000*
> *a=framesize:96 176-144*

a=framerate:8

a=fmtp:96 profile=0; level=45

a=rtpmap:97 MP4V-ES

a=framesize:97 176-144

Support is indicated for H.263 Profile 0 Level 45 and for MPEG4, with a data rate of 116kbps to be used.

In Video Share Sessions where Audio is also carried over PS, the SDP must carry media descriptions for the Audio stream as well in the SDP Offer and Answer similar to the Video Stream. An example SDP Offer with descriptions of both the Audio and Video Stream is as follows: (the SDP for Audio/video encoding and RTP packetization are discussed in Section 2.7 & 2.8 below)

Informative Example: 6

m=video portUE-A1 RTP/AVP 96

a=sendonly

a=X-vsrc-type:videolive

a=rtpmap:96 H263-2000/90000

a=framesize:96 176-144

a=framerate:8

a=fmtp:96 profile=0; level=45

b=AS:54

a=curr:qos local none

a=curr:qos remote none

a=des:qos mandatory local send

a=des:qos none remote send

m=audio portUE-A2 RTP/AVP 97

a=sendonly

a=rtpmap:97 AMR/8000/1

a=fmtp:97 mode-set=7;octet-align=1

a=ptime=20

a=maxptime=240

b=AS:8

a=curr:qos local none

a=curr:qos remote none

a=des:qos mandatory local send

a=des:qos none remote send

The a=X-vsrc-type SDP attribute:

The Line a=X-type:videolive specified in IR.74 [3] is deprecated in this specification. Instead the attribute *a=X-vsrc-type* may be used to indicate whether the Video is Live or whether the video is from a recorded clip. The possible values are 'videolive' and 'videoclip'. The absence of this attribute in the SDP implies 'videolive'.

The a=X-vrsrc-id SDP attribute:

A media level attribute used to carry Video Stream identifier and optionally a associated password. The identifier may be a filename(can include relative path) whose usage is

described in Section 2.9.3 and Section 2.6.5 or a string identifying a Video Stream on the VS-AS whose usage is described in Section 2.8.5.

The password string may optionally exist following the Stream-identifier and a space.

Example: a=X-vrsrc-id: myVideo.h263

               a=X-vrsrc-id: *bob123/abc19871.3GP*

               a=X-vrsrc-id: vstream123 opensesame

2.6.2   Point to Multipoint Session

In Point to Multipoint Sessions, a Phase2 VS Device transmits Video to 'n' other Phase1 or Phase2 VS Devices. It may or may not be in combination of a CS Voice call.

This is achieved by IMS Conferencing mechanism described in 3GPP TS 24.147 [22].

The VS-AS must facilitate the Conferencing by supporting the capabilities of the MRFC. The specification 3GPP TS 24.147 [22] describes procedures for a combined conferencing VS-AS and MRFC. That is the VS-AS & MRFC may either be collocated or interoperate using a proprietary protocol and proprietary functional split. The MRFP and MRFC as described in 3GPP TS 24.147 [22].

Figure 7 shows the setup of a Point to Multipoint Video Share Session.
A Phase2 VS Device can create a VS Conference and invite other Phase1 or Phase2 VS Devices into the newly created Conference by sending an INVITE request to a Conference-Factory-URI, which is *to Operator specific,* along with the list of SIP or Tel URI's of the VS devices (henceforth referred to as participants) to be invited in the body of the INVITE.

The VS Device initiating the conference, henceforth referred to as the Controlling Participant, must receive a Conference URI generated by the VS-AS/MRFC via the provisional/final response to the INVITE. The Conference URI is subsequently used for adding new participants to the conference and for subscribing to Conference Event Notifications, which will be further described in Sections 2.6.2.2 & 2.6.2.3 below.

The RTP path is setup between the Participants and the MRFP. The MRFP receives the Video from controlling participant (over the respective RTP session) and sends a duplicate of it to the every other participant over their respective RTP sessions. It is to be noted that this is the only direction of Video Transmission possible in the Conference.

The Controlling Participant must subscribe (refer Section 2.6.2.3) to the Conference Event Notifications after creating the Conference. Subsequently it must receive notifications whenever a participant joins or leaves.

Though the setup of RTP path between the Controlling participant and the MRFP may be complete, it does not start transmission of Video until all the willing Participants have joined the conference.

The SIP headers and SDP details to be used specified in Section 2.6.1.1 applies here as well. In addition the Contact header in the 183 provisional responses from the VS-AS will contain the Conference URI and the "isfocus" feature tag.

The Section 2.6.2.1 describes the details about the URI List to be carried in the body of INVITE.
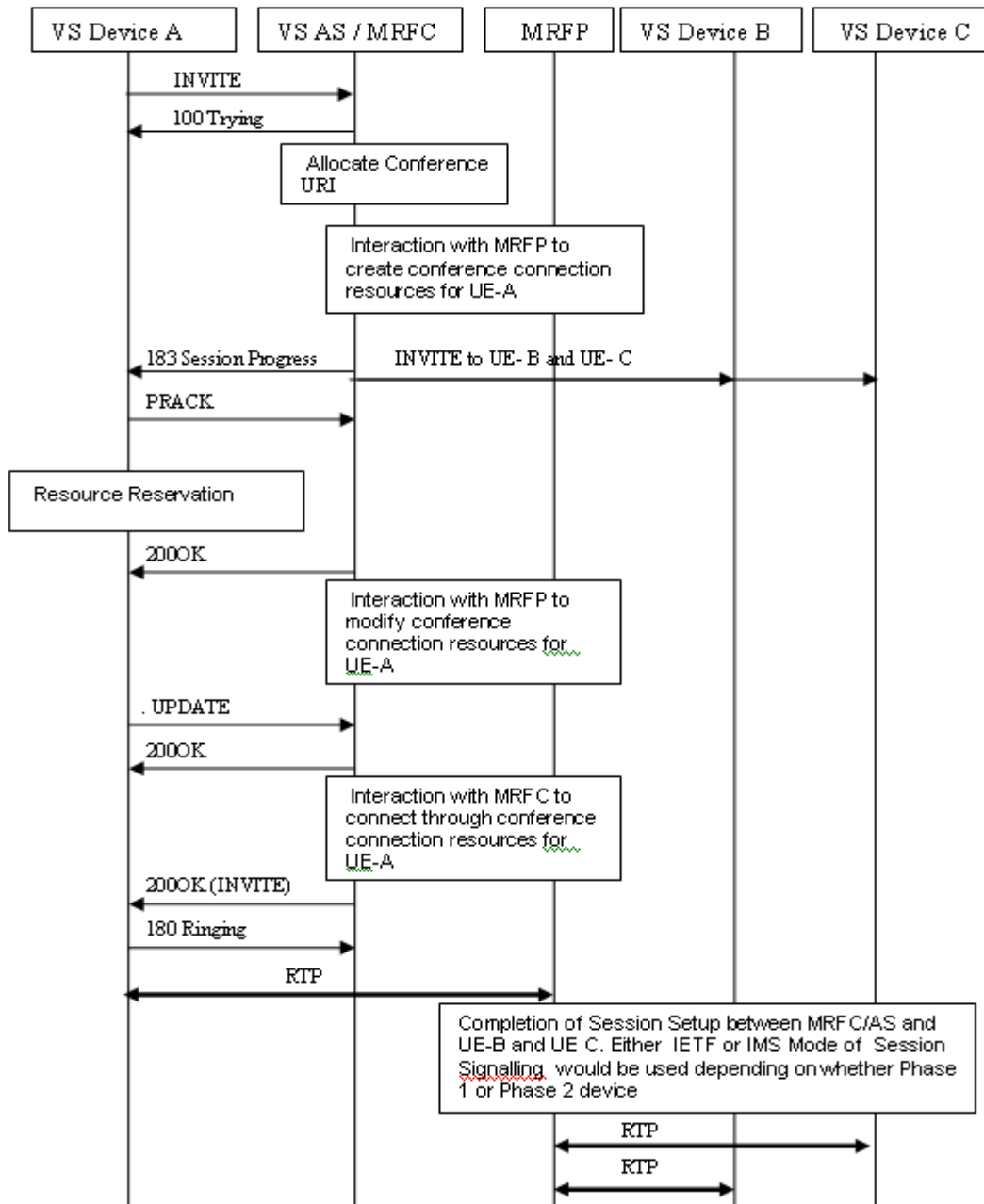

Figure 7: Point to Multipoint VS Session

2.6.2.1   URI List in INVITE

The INVITE generated by the Controlling Participant must contain a URI List in the message body as specified in 3GPP TS 24.147[22] Section 5.4.1.5.4. The INVITE request will need to carry a multipart body: a session description and the URI List.

Following is an example of the message body of the INVITE:

Content-Type: multipart/mixed;boundary="boundary1"

--boundary1

Content-Type: application/sdp

v=

o=

s=

c=

t=

m=video portUE-A RTP/AVP 96

a=sendonly

a=X-type:videolive

a=rtpmap:96 H263-2000/90000

a=framesize:96 176-144

a=framerate:8

a=fmtp:96 profile=0; level=45

b=AS:54

a=curr:qos local none

a=curr:qos remote none

a=des:qos mandatory local send

a=des:qos none remote send

--boundary1

Content-Type: application/resource-lists+xml

Content-Disposition: recipient-list

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

<resource-lists xmlns="urn:ietf:params:xml:ns:resource-lists"

 xmlns:cp="urn:ietf:params:xml:ns:capacity">

 <list>

   <entry uri="sip:VS DEVICE-B@xyz.net" cp:capacity="to" />

   <entry uri="sip: VS DEVICE-C@xyz.net" cp:capacity="to"

  </list>

</resource-lists>

--boundary1--

2.6.2.2   Adding & Removing a Conference Participant

Only the Controlling participant can initiate adding of a new Participant or list of new participants into the Conference.

Similarly only the Controlling participant can initiate removal of a Participant or list of new participants from the Conference. It does so by using the SIP REFER request as described in draft-ietf-sip-multiple-refer [8] & RFC 3515 [12].

A Participant can by itself leave the Session by normal SIP BYE message exchange as specified in 3GPP TS 24.147[22] Section 5.3.1.6.1 & 5.3.2.6.1, which causes the VS-AS to end the session with Leaving participant and notification about the same to the Controlling

Participant. The Section 2.6.3 describes the Session Termination sequence if the Controlling Participant itself initiates a BYE.
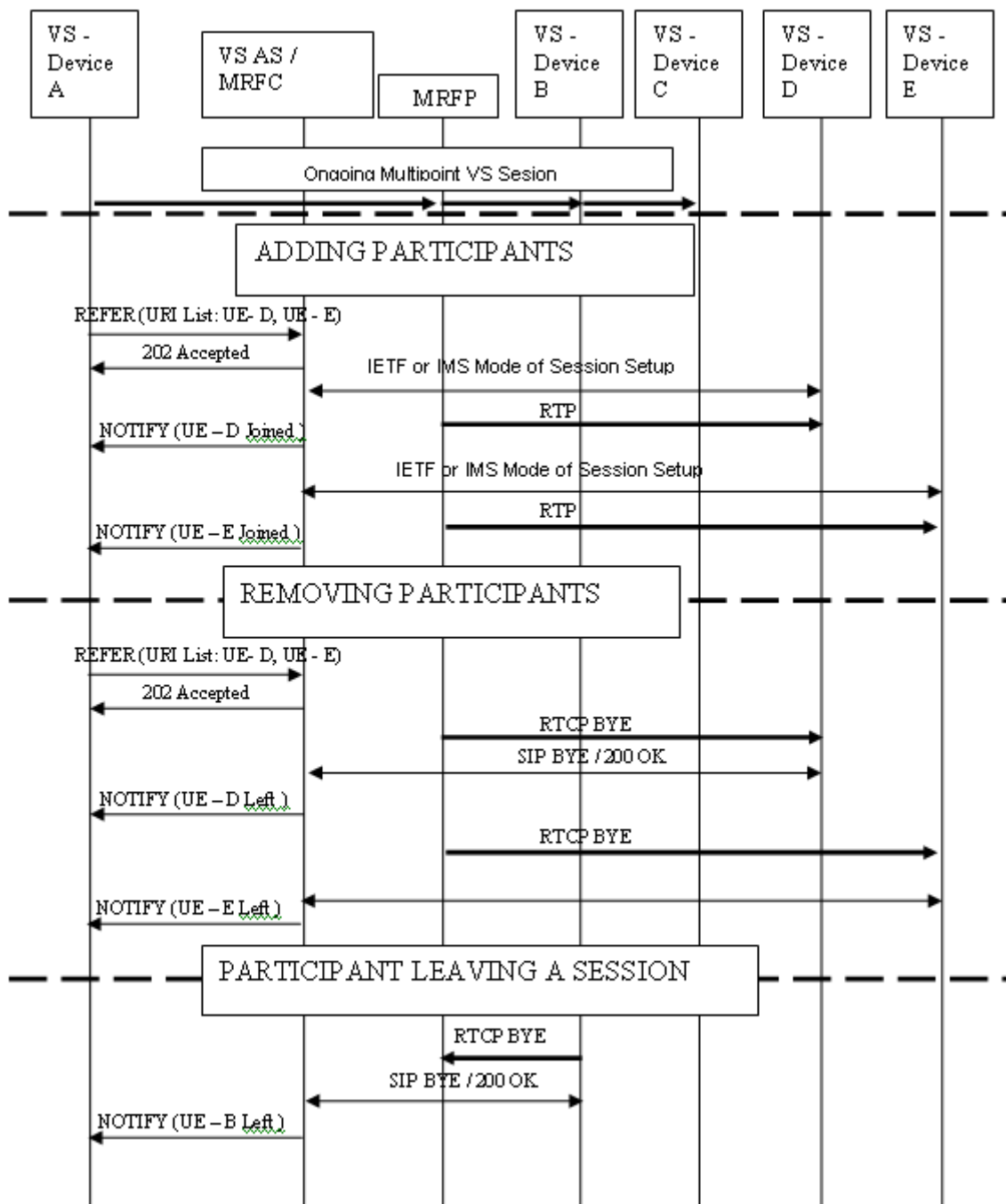

Figure 8: Adding/Removing a participant to the VS Session

Following is an example of REFER SIP Messages used above for adding participants (NOTE that some headers are intentionally not listed for simplicity):

    REFER sip:VSconf-123@xyz.com; SIP/2.0

    To: "Conference 123" <sip:VSconf-123@xyz.com>

    From: VS DEVICE-A <sip:VS DEVICE_A @xyz.com>;tag=32331

    Call-ID: d432fa84b4c76e66710

    CSeq: 2 REFER

Contact: <sip:VS DEVICE_A@xyz.com>
Refer-To: <cid:cn35t8jf02@xyz.com>
Referred-By: <VS DEVICE_A@xyz.com>
Refer-Sub: false
Require: multiple-refer, norefersub
Accept: application/sdp, message/sipfrag
Content-Type: application/resource-lists+xml
Content-Disposition: recipient-list
Content-Length: 362
Content-ID: <cn35t8jf02@xyz.com>
<?xml version="1.0" encoding="UTF-8"?>
<resource-lists xmlns="urn:ietf:params:xml:ns:resource-lists"
    xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance">
<list>
    <entry uri="sip:VS DEVICE_D@xyz.com?method=INVITE" />
    <entry uri="sip:VS DEVICE_E@xyz.org?method=INVITE" />
</list>
</resource-lists>

It is highly recommended that the INVITE request sent out by the VS-AS as a result of receiving the REFER includes the Referred-By header. The invited participant can know the Controlling Participant from this header.

2.6.2.3   Subscription to Conference Event Notifications

The Participants in the Point to Multipoint Session must subscribe to Conference Notification Service provided by the VS-AS/MRFC as defined in 3GPP TS 24.147[22] Section 5.3.3 & RFC 4575[11].

The Participants receive Notifications whenever there is a change in state of the Conference, that is, whenever participants join or leave the Conference.
When Controlling participant generates a REFER request with multiple REFER-Targets, it is recommended to include the "norefersub" option-tag in Require header and will include a Refer-Sub header field set to "false" to suppress the REFER's implicit subscription as described in Section 5 of draft-ietf-sip-mulitple-refer[8].

All subscriptions to Conference Event Package must be terminated in accordance with RFC 4575[11] when the Conference is Terminated.

2.6.3   Extending P2P to Point to Multipoint Session

If a Phase2 VS Device has already initiated and engaged in a P2P VS Session, it can include another Phase1 or Phase2 VS Device, thus transferring the P2P VS Session to a Point to Multipoint Session.

Figure 9 illustrates the procedure. VS DEVICE-A is already engaged in a P2P VS Session with VS DEVICE-B. Now in order to include VS DEVICE-C into the Session, VS DEVICE-A uses the procedure described in Section 2.6.2 to make a point to Multipoint Session involving VS DEVICE-B and VS DEVICE-C. However, here, VS DEVICE-A includes the SIP

Dialog (refer RFC 3261[19] for definition of SIP Dialog) info, that is, {call-id;to-tag;from-tag} of the already existing dialog with VS-AS corresponding to the P2P Session with VS DEVICE-B, as part of URI of VS DEVICE-B in the URI List according to procedures in sub-clause 19.1.1 of RFC 3261[19]. Thus VS DEVICE-A indicates to VS-AS/MRFC that existing dialog with VS DEVICE-B is to be replaced. The VS-AS/MRFC must include the Replaces header in the INVITE containing the dialog-id (call-id;to-tag;from-tag) of the already existing dialog with VS DEVICE-B as described in RFC 3891.
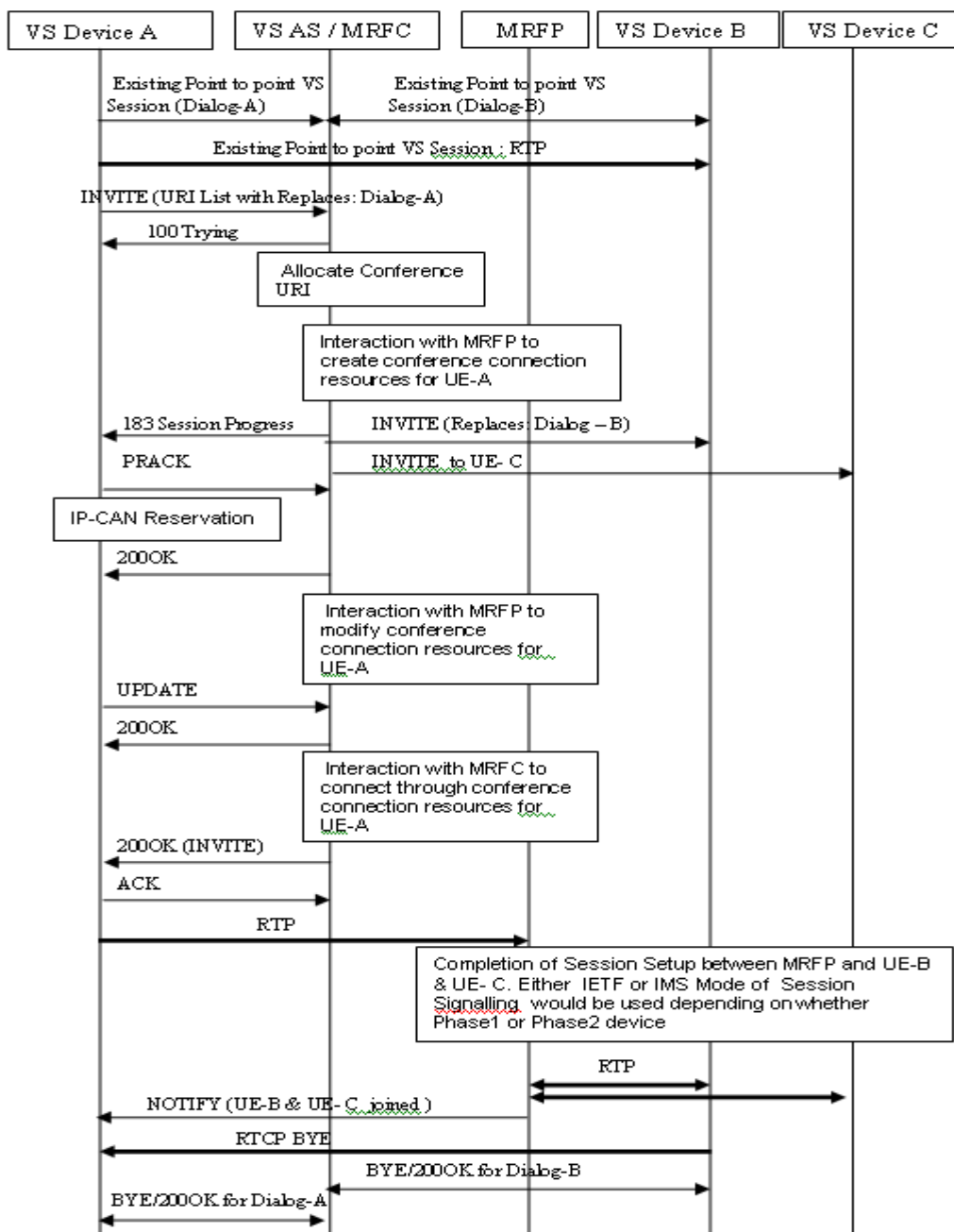

Figure 9: Transfer of P2P to Point to Multipoint VS Session

After new session is setup between VS-AS and VS DEVICE- B, VS DEVICE-B must release the old Session with VS-AS and in turn the VS-AS releases the corresponding session with VS DEVICE-A.

It is not possible for a Phase2 VS Device engaged in a P2P VS Session with Phase1 VS Device to Transfer a P2P session to a Point to Multipoint Session because the Phase1 VS Device may not support Replaces header. In such a case, the VS-AS can decline the INVITE request.

Following is an example of the URI List contained in the INVITE from VS DEVICE-A to VS-AS in the Figure above:

> *Content-Type: application/resource-lists+xml*
>
> *Content-Disposition: recipient-list*
>
> *<?xml version="1.0" encoding="UTF-8"?>*
>
> *<resource-lists xmlns="urn:ietf:params:xml:ns:resource-lists"*
>
> > *xmlns:cp="urn:ietf:params:xml:ns:capacity">*
>
> *<list>*
>
> > *<entry uri="sip:VS DEVICE-B@xyz.net" ? Call-ID= dialog-a&to-tag=12345&from-tag=54321/>*
>
> *<entry uri="sip: VS DEVICE-C@xyz.net"*
>
> *</list>*
>
> *</resource-lists>*

Following is an example of the INVITE (without the Body) from VS-AS to VS DEVICE-B in the figure above (NOTE that some headers are intentionally not listed for simplicity):

> *INVITE sip:VS DEVICE_B@xyz.net SIP/2.0*
> *From: <sip:conf-id@xyz.net>;tag=171828*
> *To: <sip:VS DEVICE_B@xyz.net>*
> *Call-ID: cb03a0s09a2sdfglkj490333*
> *Cseq: 127 INVITE*
> *Referred-By: <sip:VS DEVICE_A@xyz.net>*
> *Contact: <sip:conference1@mrfc1.home1.net>;isfocus*
> *Replaces: dialog-b;to-tag=7743;from-tag=6472*
>
> *Content-Type: application/sdp*

2.6.4   Session termination

In the case of Point to Multipoint Sessions, if the Controlling Participant leaves the session by procedure described in Section 2.6.2.2. above, then entire session is terminated, that is the VS-AS must initiate SIP BYE towards all the other participants in the Session.
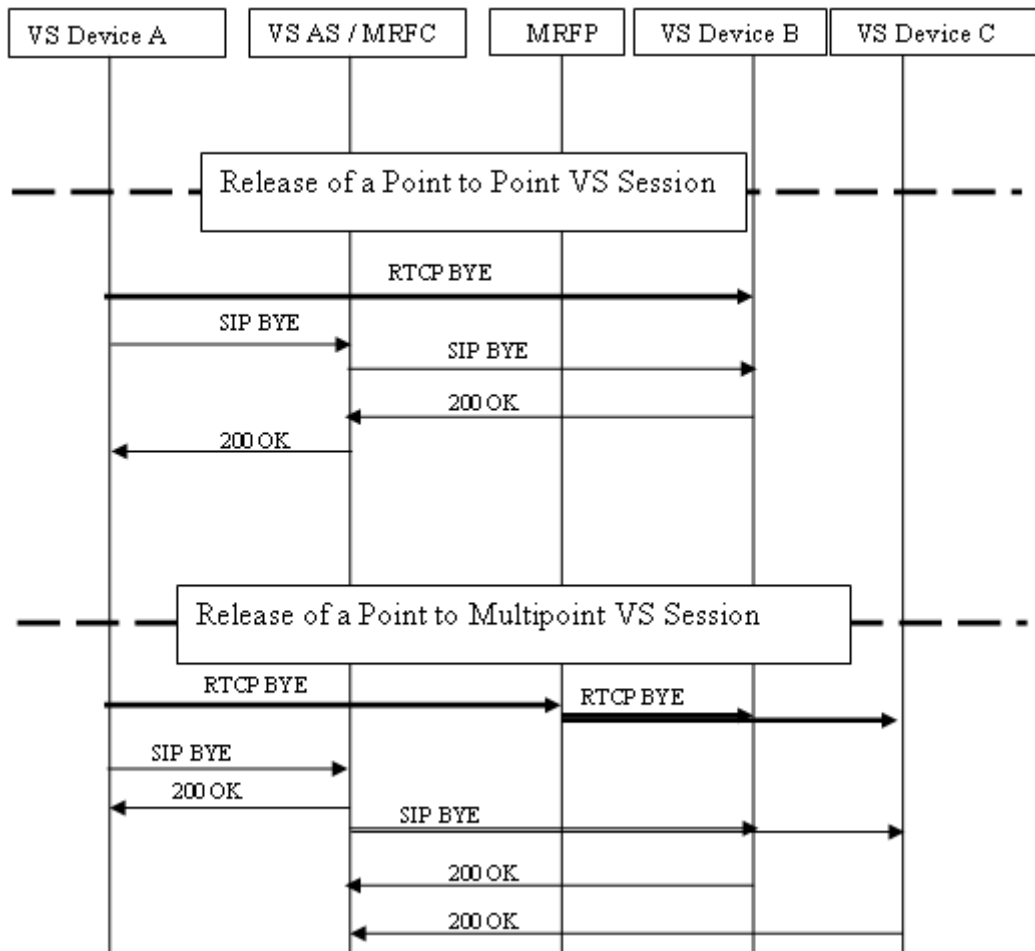
Figure 10 : VS Session Termination

2.6.5   VS Session with Video sourced from Video Storage

This is a variation of the normal P2P VS Session. In a normal P2P VS Session, the video is sourced from the originating VS-Device. Within the originating VS-Device, the source can either be a video clip that is stored on the device or a live feed from the camera. In this variation of the normal P2P VS Session, the source of the video is a video clip that is stored in the network based Video Storage. The P2P VS Session can be initiated during a CS Voice Call or outside a CS Voice Call.

Please note that Point to Multipoint sessions with Video sourced from Video storage is not supported.

The originating VS-Device communicates the name (the string may include the path as on the Video Storage) of the video clip to use as part of the INVITE message using the SDP session level attribute X-vrsrc-id. The SIP signaling flows remains the same as the normal VS session, as described in Section 2.6.1. However the SDP content in the offer and answer exchanged between VS Device-A and the VS-AS varies in terms of directionality of media (that is, in media level direction attribute and in direction-tag of QoS Pre-condition attribute) and existence of X-vrsrc-id attribute, as explained in the informative example's below.

Figure 11 shows a high level interaction sequence between the VS Devices, VS-AS and Video Storage.
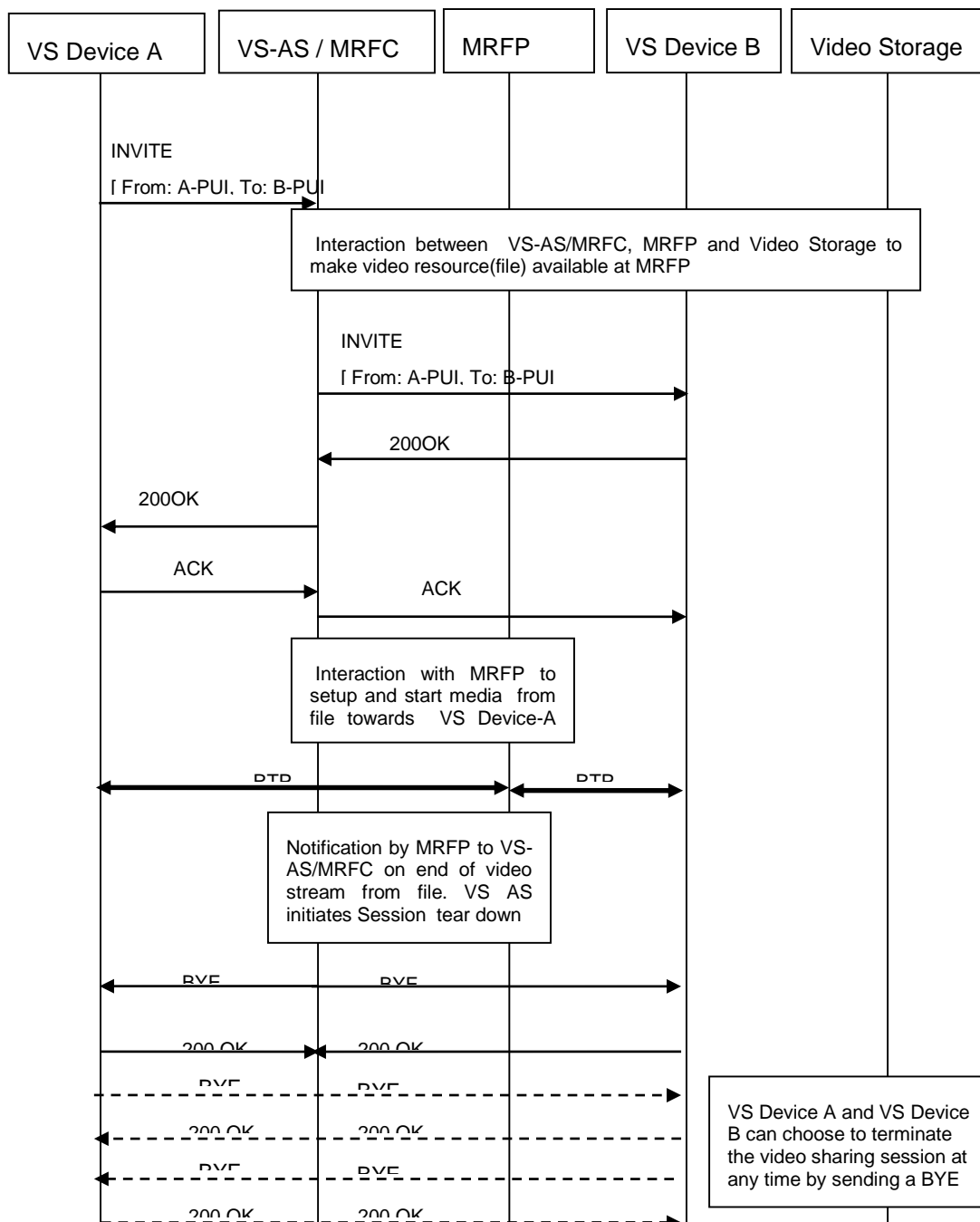

Figure 11: P2P VS Session with Video Sourced from Video Storage

Following is informative example showing usage of x-vrsrc-id and QoS Precondition SDP attributes:

Offer from VS Device-A

m=video port UE-A RTP/AVP 96
a=recvonly

a=X-vrsrc-id:bob123/abc19871.h263
a=rtpmap:96 H263-2000/90000
a=framesize:96 176-144
a=framerate:8
a=fmtp:96 profile=0; level=45
b=AS:54
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local recv
a=des:qos none remote recv

Answer from VS-AS

m=video port UE-B 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
b=AS:54
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory remote send
a=des:qos mandatory local send
a=conf:qos remote send

When the VS-AS/MRFC receives the name of the video clip (via X-vrsrc-id) to be shared, a HTTP URL of the video clip on the Video Storage is composed by appending the received value in the X-vrsrc-id to the HTTP URL of the Video Storage (for example:if X-vrsrc-id= abc19871.3GP, then the composed HTTP URL would be "http://content-server.xyz.net/content/bob123/abc19871.3GP"). Using this, HTTP URL VS-AS initiates downloading of the video clip to a temporary file in the MRFP through the Wb2 interface.

The VS-AS/MRFC may start streaming the video clip to originating and terminating VS-devices once the video clip has been completely downloaded or before the entire video clip has been completely downloaded.

The video sharing session is terminated by VS-AS when it has completed streaming the selected file. However, the originating or terminating user can also terminate the video sharing session at any time (while the video stream is in progress).

2.7   Audio/Video Encoding

Video codec H.263-2000 profile 0 level 45 is mandatory. Note that the indication of H.263 profile 0 level 45 in SDP implies support of H.263 profile 0 level 10.

When using the H.263 video codec, only QCIF resolution video must be supported for Video Share. The recommended frame rate is 15fps. The 'framerate' and 'framesize' media-level SDP attributes are used as specified in RFC4566 [14], to indicate the same.

Optionally, MPEG4 Visual Simple Profile 0b and H.264/AVC Baseline Profile Level 1b maybe supported.

The AMR Narrowband Audio Codec must be used in 12.2kbps mode as specified in RFC 4867[16].

The encoding name to be used with the media –level SDP 'rtpmap' attribute is 'AMR/8000/1'. The 12.2kbps mode is indicated using the fmtp parameter 'mode-set=7'.

Some examples SDP are discussed in Section 2.6.1.1 above.

Video Share over LTE network is not excluded. Implementations may use LTE Quality of Service class identifier (QCI) 7, 8 or 9 for video stream. QCI 7 is recommended.

For LTE, Video codec H.264 Baseline profile level 1.3 is mandatory. RTP payload format for H.264 is used as specified in RFC 3984. It should be noted that in case of handover taking place e.g. between LTE and WCDMA during the Video Share session, the codec will remain the same but lower available bandwidth could mean switching to a lower level of H.264.

2.8   RTP Packetization

The RTP Profile for Audio and Video Conferences with Minimal Control (RFC 3551 [13]), also called RTP/AVP, must be used for transport.

When using H.263 video codec, the RTP payload format & MIME subtype for H.263-2000 specified in RFC 4629[40] must be used.

When using MPEG4 Visual Simple Profile 0b, the RTP payload format is as specified in RFC 3016[41] and for H.264/AVC Baseline Profile Level 1b, the RTP payload format is as specified in RFC 3984[42]

The AMR speech codec RTP payload format as specified in IETF RFC 3267[38] is to be used.

Specifically,

- Octet-aligned payload format to be used
- The SDP attribute 'octet-align=1' is to be used as specified in the RFC 4867[16]
- The max length of time represented by the Audio in the a RTP packet must be 240 milliseconds (12 frames per RTP Packet) and can be a minimum of 20 milliseconds.
- The media-level SDP attribute 'maxptime' & 'ptime' as specified in RFC4566[14] and RFC 4867 [16] are to be used respectively.

Some example SDP are discussed in Section 2.6.1.1 above.

2.9 Aspects of Video Storage and VS with non-VS capable terminals UC's

2.9.1 Web - Portal

It is a Logical entity on network whose capabilities and function are as follows:
It provides a Web Interface (that is, Wb1 in Figure 1) to the Subscriber using which the Subscriber can Login (using a Subscriber ID and password) and access and manage (for example, management of access right authorisations provided to other users) the Video Clips uploaded to the Video Storage, for example, in Use Case UC3.2.

It provides Web Interface(that is, Wb1 in Figure 1) to facilitate Web Client access to a Video Stream being shared by a VS Terminal to the VS-AS as expected in Use Case UC 4.2 in Service Definition[1]. Specifically, it serves the Web Client with the RTSP URL (via the Wb1 interface) that point to Video Stream in MRFP.

VS-AS/MRFC provides the Web-Portal with all necessary information about the list of video contents that is made available at given time to a particular subscriber or other users, as well as associated URLs which enable the user to get access to video contents either from video storage (that is. retrieving video content from video storage in Use Case UC3.3) or from MRFP (for example, in case of live video sharing with non-VS capable terminal via RTSP streaming in Use Case UC4.2).

Detailed specification of interfaces between Web-Portal, Video Storage and VS-AS are outside of the scope of this document.

2.9.2 Video Storage (Multimedia Storage)

Video Storage is a Logical entity on network whose capabilities and function are as follows:

- It Holds the Storage Space for each VS Phase 2 Subscriber. The Subscriber uploads video to this space via the Video Storage use case specified by UC3.2 in the VS Phase 2 Service Definition[1].
- It provides an interface (that is, Wb1 in Figure 1) to facilitate web client access to a Video Stream, for example, in Use Case UC4.3 as well as an interface (that is, Wb2 in Figure 1) enabling storage of Video Stream sent by VS terminal towards MRFP (e.g. in Use Case UC3.2)

The Video Storage function may rely on more generic multimedia storage function (for example, for image, multimedia contents) offered in operator network.

Detailed specification of interfaces between Web-Portal, Video Storage and VS-AS are outside of the scope of this document.

2.9.3 Video Sharing to Video Storage and Web - Portal

As part of Session Setup signaling for use case UC 3.2 (refer Service Definition [1] document), the Offer SDP may contain the *X-vrsrc-id* attribute suggesting the name of the file in which the video is to be stored on the Video Storage. The VS-AS will check if the file can be created with this filename, if it is not possible (if another file already exists with the same name) it will use an internally generated string. The implementer will choose a file name generating method that guarantees uniqueness across the network.

The Answer SDP must contain the *X-vrsrc-id* which will carry the name of the file that will be stored in the Video-Storage. This may or may not match the value specified in the Offer SDP.

The MRFP creates a file with the Video received and transfers the file to the Video StorageI over the Wb2 interface.

The Subscriber has prior knowledge of the URL of the Web-Portal. The subscriber can login to his space on the Web-Portal using the Web Interface, that is, Wb1(eg: HTTP) in Figure1 and access the Video File. The file can be made accessible via HTTP from the Internet (for example, Wb1 in Figure 1) to other users. VS Terminal's from another Operator's IMS Network can access the file over the Wb1(HTTP) interface. Discussion on the Web Interface is out of scope of this specification.

It is to be noted that the Web portal may be accessed via VS-AS that is VS-AS may act as a HTTP proxy hence the HTTP request & responses may be routed via VS-AS.

2.9.4   Video Sharing with non-VS capable via SMS or MMS

When UE-A initiates a Video Share towards a non-VS capable UE-B, based on service policies (ref to Service Policy parameter set to VSSMS or VSMMS as detailed in use case UC 4.1 and UC 4.3 in Service Definition [1] document) the MRFP stores the incoming video from UE-A in a temporary video file and uploads the file into Video Storage over the Wb2 (for example, HTTP) interface. Subsequently, the VS-AS can either dispatch a SMS to UE-B containing the HTTP URL of the temporary video file or sends this file as an MMS message (through the MMSC) to UE- B.

2.9.5   Live Video Sharing with non-VS capable terminal via RTSP Streaming

This is corresponding to UC 4.2 in the Service Definition [1]. First a VS Session is setup between VS DEVICE-A (the sharer) and the VS-AS.

The remote can access the live Video Stream either via the Web Portal or directly through MRFP. In both cases, the VS-AS must generate a Video Stream Id and optionally a password. They are included in the X-vrsrc-id attribute of the Answer SDP.

- Video Stream access from remote Web Client via Web-Portal:

The VS-AS must also compose the RTSP URL (which points to the MRFP) and post it to the Web-Portal.

The Web Portal URL, the Video Stream Id and optionally password are made available to the remote by VS-AS directly (for example, via SMS) or via the sharer (thanks to the X-vrsrc-id attribute of the Answer SDP sent by VS AS to the VS DEVICE-A)

The remote terminal may have prior knowledge of the HTTP URL of the Web-Portal.

The remote uses a Web browser and accesses the Web-Portal via the Web-Portal URL and fetches the RTSP URL by providing the Video Stream id & optionally the password. The RTSP URL points to the MRFP, which establishes a RTSP Session towards remote terminal over Wb3 interface (for example, RTSP) in order to stream down the Video.

It is to be noted that the Web-Portal may be accessed via VS-AS, that is, VS-AS may act as a HTTP proxy hence the HTTP request & responses may be routed via VS-AS

- Video Stream access from remote terminal directly through MRFP:

The VS-AS must compose the RTSP URL (which points to the MRFP)

The RTSP URL, the Video Stream Id and optionally password are made available to the remote by VS-AS directly (for example, via SMS).

The remote uses a Web browser to access the RTSP URL.

The RTSP URL points to the MRFP, which establishes a RTSP Session towards remote terminal over Wb3 interface (for example, RTSP) in order to stream down the Video.

2.10 Media Hold & Resume

During an ongoing VS Session, the media, that is, Video can be put on Hold on request from remote. It may be triggered either by User itself or as a result of IMS Service behavior on CS Call Hold Supplementary Service.

If VS DEVICE-A needs to request VS DEVICE-B to Hold the media, it does so by sending a new offer in a SIP UPDATE with the media level SDP attribute "a=inactive" for the m-line describing the Video Stream, as specified in RFC 3264[39] Section 8.4.

Subsequently VS DEVICE-A can resume the media by sending another offer in a SIP UPDATE with the SDP attribute "a=recvonly" or "a=sendonly" depending on what it was previously set to. This behavior is in accordance with Section 4.5.2 of 3GPP TS 24.410[27].

Figure 12 illustrates the operation sequence:

In case of a Point to Point VS Session, both Participants shall be allowed to Resume regardless which one has initiated the Hold.

In case of Point to Multipoint VS Sessions to following is the behavior:

- If Controlling participant initiates a Hold then Video streams to all the Participants are halted.
- If Controlling participant initiates a resume then Video streams to all the Participants are resumed.
- If a Participant initiates a Hold then Video Stream is halted to only that Participant.
- If a Participant initiates a Resume then Video Stream is resumed to only that Participant.
- If Participant has put the Video on Hold then only that Participant is allowed to Resume it, a Resume by Controlling participant will not have any effect.
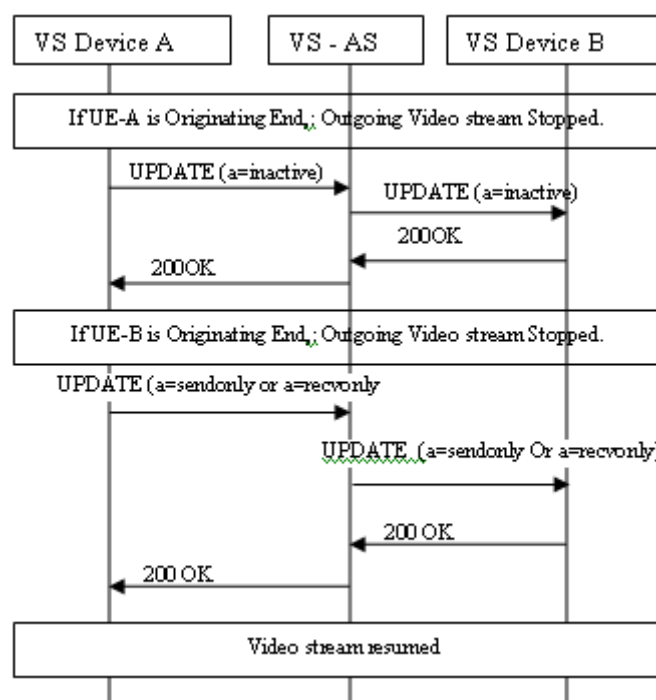
Figure 12: VS Hold & Resume

2.11 Session behaviour on CS Supplementary Services

2.11.1 Caller ID Restriction (CLIR)

If CLIR is enabled for the voice call then the VS session needs to follow the privacy recommendation of 3GPP 24.228 as follows:

-   If the initiating user desires the session privacy for the asserted identity, the tag "id" is used in Privacy header and anonymous username is used in the From header.

    Example:

    >   Privacy: id

    >   From: "Anonymous" <sip:anonymous@anonymous.invalid>

If the destination user desires the P-Asserted-Identity to be private, the UE indicates this by including tag "id" as value in the Privacy header of the first non-100 response to the initial INVITE

2.11.2 Call Hold

An Ongoing CS Call is put on Hold (Call Hold Supplementary Service) either when another CS call is originated or when an incoming Call is Accepted without disconnecting the ongoing Call.

If there is an ongoing CSI VS Session with the same remote as the ongoing CS Call then the Video Stream is put on Hold by the End Point where Call Hold is initiated and resumed when the corresponding CS Call gets resumed. The operation of Hold and Resume of Video is explained in Section 2.10 above.

2.12   Quality of Service (QoS)

In Phase2, Quality of Service is to be provided in two folds, one at the Radio Access Bearer path and other in the Core PS Network.

2.12.1   QoS over Radio Access Bearer

The Radio Access for Video Share Service is WCDMA. (Video Share over EDGE/DTM is not excluded as noted in IR.74 [1] Section 3.5).

A Radio Access Bearer is setup by activating a PDP Context, the PDP context terminates at GGSN in the Core PS Network.

The PDP Context is setup with appropriate QoS parameters in order to ensure QoS over the Radio Access Bearer. The mechanism to specify QoS parameters in PDP Context is explained in 3GPP TS 24.008[28] Section 10.5.6.5. The recommended Traffic Class to be used is Streaming (refer 3GPP TS 23.107[32]). The Maximum Bit Rate and Guaranteed Bit Rate QoS parameters used must be at least 128kbps and 64kbps respectively.

A Secondary PDP context for Media can be supported, in which case the Traffic Class recommended to be used by the Primary PDP Context is Interactive and for the Secondary is Streaming.

2.12.2   QoS over PS Core network

The parameters and mechanisms to achieve QoS in PS Core PS network is as described in 3GPP TS 29.060[29] and Section 8 of GSMA IR.34 discusses aspect of QoS over Inter-PLMN backbone.

In order to provide for QoS in the PS Core Network the nodes in the VS Media must support DiffServ mechanism and support DSCP marking of IP Packets associated with Video Share media streaming. Following Table describes the Traffic Class and DiffServ related details for Video Share Usage. More details on DiffServ usage can be found in GSMA IR.34.

Traffic Class	Diffserv PHB	DSCP	Video Share Usage
Streaming	AF41	100010	Video Streaming
Interactive	AF31	011010	Signalling Path
[As per Operator Specification]	AF21	010010	Clip download from Storage

Table 3: Traffic Class and DiffServ details for VS usage

2.12.3 QoS negotiation

It is necessary that IMS Mode signaling is used for Session Setup, as part of which the QoS pre-condition and bandwidth modifier must be included in SDP (as described in Section 2.6.1 above) in order to negotiate the QoS parameter (bandwidth) with the remote VS device and establishing Session only if the QoS is obtained.

The QoS signaling flows will follow the guidelines described in 3GPP TS 29.208[31] and 3GPP TS 29.207[30].

Considering the Session setup flow illustrated in Figure 5, the following:

- If UE A gets a lower Guaranteed Bitrate than the one requested, the obtained bitrate is reflected in the SIP UPDATE. The Rate Adaptation function in the UE must operate considering the obtained Bitrate as the Upper Limit
- If UE B gets a lower Guaranteed Bitrate than the one requested, the obtained Bitrate is reflected in the 200 OK (INVITE)
- If UE-A or UE-B fails wants to use a Secondary PDP Context, but fails to get a Secondary PDP Context during session setup it can fallback to Primary PDP and the updated Offer (SDP, since IP & Port may change) can be sent in UPDATE and 200OK(INVITE)

2.13 Other features

- SigComp can be used, but it is not mandatory.
- Both PDP Always-on and PDP Per Call modes can be used.
- Because the use of IMS is an inherent part of the Video Share service, it is likely that a PDP context for the transport of SIP signalling will be active and in the Always-On mode of operation. Hence in PDP Per Call mode, the context to be established is likely to only be that for the media plane. The Always-on method is preferred over the Per call method due to a decreased the risk of SIP registration race conditions and less radio access traffic load.
- The use of PDP contexts for Signalling and Media plane can follow one of three models:

  - The signaling and media plane are transported within the same context, but separated on the Gi (Gm and Mb) interface by the IP address of the IP datagram's transported within the context. This allows usage of a single APN, but requires the context to support the maximum required QoS parameters for the signalling and media paths. Hence when the media path is added, an Update PDP Context procedure may be required.
  - The PDP context for the signaling messages is established and the signaling triggers establishment of a secondary PDP context (where the signaling PDP context is the associated Primary PDP context). This implies use of the same APN but offers some flexibility around differentiating the QoS of the two contexts.
  - The PDP context for the signalling messages is established and the signalling triggers establishment of an additional Primary PDP context. This allows separation of traffic at the APN level and complete flexibility with regard to allocating different QoS parameters to the PDP contexts transporting signalling and media, but is less resource efficient.

- Both Tel URI and SIP URI addressing schemes can be used

  - Networks, terminals and Video Share application need to support tel URI addressing
  - Terminals must be able to initiate Capability Query or Session Setup (explained in Sections 2.5, 2.6 and 2.9 above) with Target as Tel URI. The Tel URI will be in E.164 format.

- IPv4 will be used for Video Share (terminals might support also IPv6)
- The CDR formats for Video Share Phase2 comply with following specifications 3GPP TS 32.260[23], 32.275[24] , 32.297[25], 32.298[26]

3   Support for 'Legacy' (Pre-Phase 1) Terminals

Video Share terminals based on legacy specifications (*Video Sharing 1.0, TurboCall, weShare 2.0*) will not be interoperable with Vide Share Phase2 terminals.

Annex A   Document Management

A.1   Document History

Version	Date	Brief Description of Change	Approval Authority	Editor / Company
1.0	16/06/2009	Initial Draft (as produced at end of VS Phase2 Project) with following changes - Document owner correction to IREG - Changes to Video Codec details to be consistent with IR.74 v1.3 Following changes as a result of enhancements proposed by Orange - Section 2.1, Level 10 modified to Level 45 - Section 2.4.1 , typo correction "<version> set to 2.0" - Change request for Section 1.1, in order to indicate that IPX is illustrated in architecture diagram as a solution (among others) regarding interconnection between operator networks. - Change request in Section 2.2 and 2.10.2, to clarify the relationships between voice call and VS Phase 2 session. - Change Request for Enhanced Architecture diagram. - Change request to improve Section 2.8 and Section 1.1 based on Enhanced Architecture Diagram -		Mahesh Anjanappa (mahesha@samsung.com) Samsung India Software Operations
2.0	12/11/2009	- Changes to support" VS Session with Video Source from Video Storage" use case as required by content sharing features in RCS R3. Introduced Section 2.5.5, updated Section 2.8.4 and moved it into Section 2.5.1, updated section 2.3.2. - Eliminating multiple Contact header in Section 2.5.1.1 - Deprecating usage of g.3gpp.app_ref and introducing g.3gpp.iari-ref and g.3gpp.icsi-ref - Replacing 'urn-xxx' to 'urn-7' - Prefixing vsrc-type and vrsrc-id with "X-" - Changed shall & should to must & will respectively	IREG	Mohan Palat – Aylus Networks Mahesh Anjanappa - Samsung India Software Operations
2.1	16/08/2010	- mCR002 to IR.84 on Ability to benefit from VS AS for VS phase 1 terminals	PACKET #44	Mahesh Anjanappa - Samsung India Software Operations
2.2	30/12/2010	- CR003 ref RILTE Doc 13_012. Details below. - added an explicit statement(Section 2.7) that also LTE is in scope of Video Share and when LTE is used then H.264 Baseline profile level 1.3 will be utilized as	RILTE#13	Mahesh Anjanappa - Samsung India Software Operations

Version	Date	Brief Description of Change	Approval Authority	Editor / Company
		the codec.Corresponding changes in Section 25.1.1, Example 1 & 2 - To meet the Content Sharing Enhancement requirements that form a part of RCS Release 4 , change's done to the way media hold & resume works, allowing both participants in a point-to-point VS session to resume the video stream		
3.0	16/01/2015	Added an ICSI for use with with P-Preferred/Asserted-Service for Video share	NG	Javier Sendin (GSMA)

Other Information

Type	Description
Document Owner	Network Group
Editor / Company	Mahesh Anjanappa - mahesha@samsung.com (Samsung India Software Operations)

It is our intention to provide a quality product for your use. If you find any errors or omissions, please contact us with your comments. You may notify us at prd@gsma.com

Your comments or suggestions & questions are always welcome.