ATIS-I-0000045

IP Services Interconnect Technical Report: Assessment of Requirements and Specifications

September 2014

1 Executive Summary

1.1 Problem Statement

With the widespread deployment of high-speed wireless and wired broadband networks, operators are migrating their existing services such as voice and SMS into an all-IP implementation. In addition, new services, such as HD voice and video communications are being introduced into the all-IP networks. As a result, operators need to ensure interconnection and interoperability between service providers (e.g., Fixed, Wireless and Cable) for all IP-based services, both existing and forward-looking services. Standards have been defined by leading SDOs (e.g., IETF, ITU, 3GPP, PacketCable, GSMA, i3Forum, SIPForum, ATIS PTSC) to address interoperability. However, many options exist across these overlapping standards, and agreement as to which of the specifications to base the interconnect upon is often a key initial step.

1.2 Objective/Scope

The objective of the focus group is to define a strategy for developing an IP services interconnect specification suitable for all service provider types (wireless, wireline, and cable), encompassing basic, advanced, and future services.

The scope includes forward-looking services such as HD Voice, Video, Messaging, and Data. Requirements are identified for a set of interconnection profiles built on existing standards to enable these services and to deliver a consistent user experience across all types of networks (e.g., Wireless, Fixed, and Cable).

This analysis is complementary to a joint effort with the SIP Forum (the ATIS/SIP Forum IP-NNI Task Force) to develop a fully specified IP-NNI for voice services.

1.3 Analysis Model & Methodology

Representative use cases are identified for each IP service to assess the standards coverage across the key technical areas of services interconnect including session establishment and signaling, addressing and routing, and media handling.

1.4 SDO/Gap Analysis

Any areas where existing specifications for interconnection of advanced services are inconsistent or incomplete are identified.

2 Reference Architecture

Figure 2.1 illustrates the interconnection reference model created by the ATIS Packet Technologies and System Committee (PTSC) for IP NNI supporting VoIP, video, and data. Illustrated are Border Elements (BEs) to enable IP interconnection between Service Providers. A BE includes an Interconnect Session Border Controller and Data Border Function.

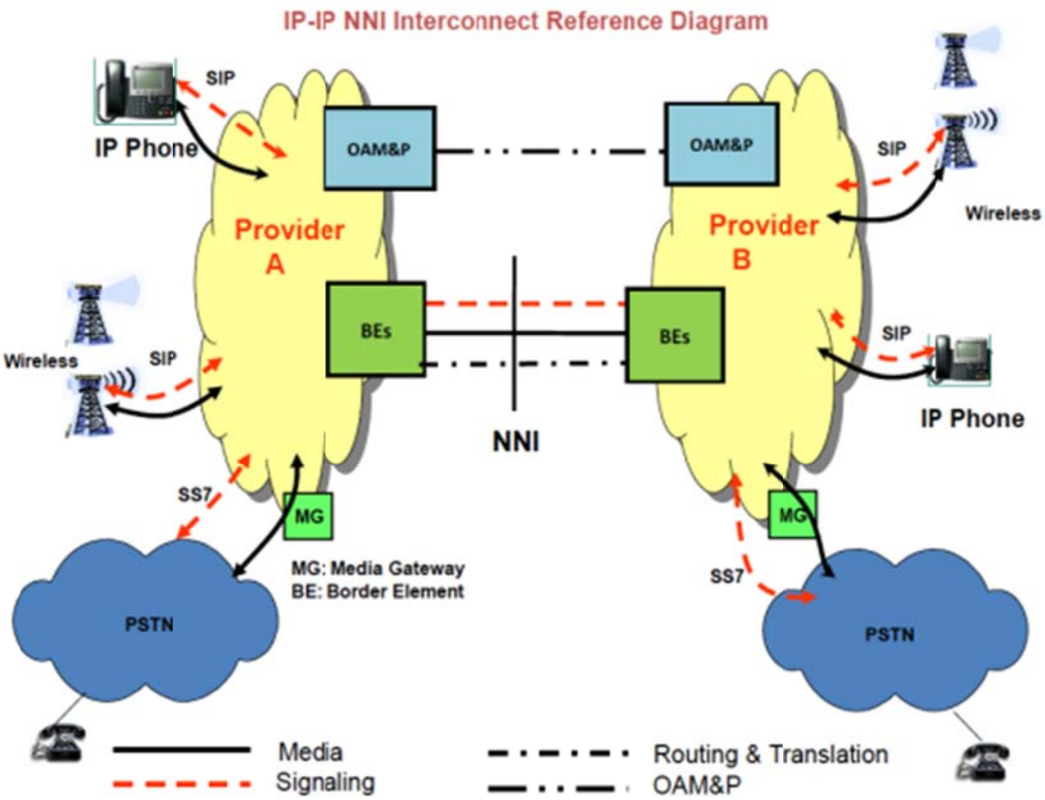

Figure 2. 1 - ATIS IP-IP NNI Interconnect Reference Model

IP interconnection may be accomplished via direct point-point connectivity or via IP eXchange (IPX). Whichever option is chosen is determined by mutual agreement of the operators.

IP services may be deployed on Softswitch-based architectures as well as IMS.

3 IP Services

3.1 HD Voice

Wideband audio, also known as HD voice, refers to the next generation of voice quality for telephony audio resulting in high definition voice quality compared to standard digital telephony "toll quality". It extends the frequency range of audio signals and doubles the sampling rate, resulting in higher quality speech. The range of the human voice extends from 80 Hz to 14 kHz but traditional, voiceband, or narrowband telephone calls limit audio frequencies to the range of 300 Hz to 3.4 kHz. Wideband audio relaxes the bandwidth limitation and transmits in the audio frequency range of 50 Hz to 7 kHz or higher.[1]

3.1.1 Relevant Standards

Key standardized wideband codecs include: ITU G.722, G.722.2 (also known as AMR-WB – Adaptive Multi-rate Wideband), EVRC-NW (Enhanced Variable Rate Narrowband-Wideband), and IETF Opus.

[1] http://en.wikipedia.org/wiki/HD-Voice

With many additional wideband codecs in commercial use, industry convergence towards mandating a single codec is unlikely in the near term. It is hoped that a core subset of wideband codecs will achieve general market adoption and this will help facilitate interworking and reduce the need for transcoding.

HD Voice for 3GPP terminals and networks uses the AMR-WB codec defined in ITU-T Recommendation G.722.2 and 3GPP TS 26.190. The HD voice service on LTE networks complies with 3GPP TS 26.114 and GSMA PRD IR.92 (VoLTE). 3GPP2 (CDMA) terminals that implement HD voice use the EVRC-NW (Enhanced Variable Rate Narrowband-Wideband) as specified in 3GPP2 C.S0014. Further technical details of AMR-WB are described in GSMA PRD IR.36.

ETSI TISPAN defines in TS 102 527 enhancements to the standards used by cordless phones, to enable wideband audio. The DECT Forum references this in their Cordless Advanced Technology – Internet and Quality (CAT-iq) initiative. CAT-iq addresses wireless broadband home connectivity in the protected frequencies used by Digital Enhanced Cordless Telecommunications (DECT) devices. In turn, there has been collaboration between CAT-iq and the Home Gateway Initiative (HGI), which is evolving the building blocks for Broadband Service Providers to deliver digital services to the home.

3.2 Video Call

See GSMA PRD IR.94 and GSMA PRD RCC.07 (RCS 5.1) ¶ 3.9.

Point-to-point video communication between two parties.

IP-based Video Call between two devices with synchronization between the audio and video streams, thus providing lip synchronization. For voice the IP-based Voice Call service is used.

The continuity of an IP-based Video Call relies on the continuity of the IP connectivity. The establishment of the IP Video Call session can be achieved in three possible ways:

1. '**Direct launch**', if no previous IP-based Voice Call was established between the contacts.
2. '**Upgrade to IP-based Video Call'**, if the users were already engaged with each other in an IP-based Voice Call communication.
3. '**Replace a CS voice with an IP-based Video Call**', if the users were already engaged with each other in a CS voice communication.

3.2.1 Relevant Standards

GSMA PRD IR.94 defines an IMS profile for a Conversational Video Service. GSMA PRD RCC.07 (RCS 5.1) defines a set of services known as the Rich Communications Suite. That set includes several video –based services, including Video Call as described above.

Internet browser-based video communication services are now emerging, and a framework of standards known as WebRTC is being jointly developed by the IETF and W3C (reference http://www.w3.org/2011/04/webrtc/ and http://datatracker.ietf.org/wg/rtcweb/documents/).

3.3 Standalone Messaging

See GSMA PRD RCC.07 (RCS 5.1) ¶ 3.2.

Integral part of the RCS framework that includes both text and multi-media messaging services based on OMA CPM as evolution from SMS and MMS. It includes the following main features:

- Standalone messaging (text and multimedia);
- Delivery and Display Notifications;
- Support for multiple devices per user;
- Deferred Messaging;
- Central Message Storage; and
- Interworking with legacy messaging services.

3.3.1 Relevant Standards

GSMA PRD RCC.07 specifies the standalone messaging capability for RCS based on OMA CPM's SIP-based standalone messaging as described in GSMA PRD RCC.11 for the endorsement of OMA CPM 2.0 Conversation Functions.

3.4 One-to-One Chat

See GSMA PRD RCC.07 (RCS 5.1) ¶ 3.3.

The One-to-One Chat service enables the instantaneous exchange of messages in a session between two users.

The one-to-one Chat service enables two users, including two users across two different Service Provider networks, to exchange messages instantly. This section describes the guidelines for the one-to-one session setup and chat message exchange between OMA IM and OMA CPM.

3.4.1 Relevant Standards

GSMA PRD RCC.07 defines two different technical realizations of One-to-One chat; OMA SIMPLE IM or OMA CPM. GSMA PRD RCC.12 (RCS) endorses OMA SIMPLE IM 2.0 and GSMA PRD RCC.09, RCC.10 and RCC.11 endorse relevant aspects of OMA CPM 2.0.

3.5 Group Chat

See GSMA PRD RCC.07 (RCS 5.1) ¶ 3.4. The Group Chat service enables the instantaneous exchange of messages in a session between many users.

3.5.1 Relevant Standards

GSMA PRD RCC.07 defines two different technical realizations of chat; i.e., OMA SIMPLE IM or OMA CPM. Furthermore, GSMA PRD RCC.12 (RCS) endorses OMA SIMPLE IM 2.0 and GSMA PRD RCC.09, RCC.10 and RCC.11 endorse relevant aspects of OMA CPM 2.0.

3.6 File Transfer

See GSMA PRD RCC.07 (RCS 5.1) ¶ 3.5. The File Transfer service enables users to exchange different types of content (files) to another user, including to a user in a different Service Provider during an ongoing session or without having an ongoing session.

3.6.1 Relevant Standards

GSMA PRD RCC.12 endorses OMA SIMPLE IM 2.0, while GSMA PRD RCC.09, RCC.10, and RCC.11 endorse relevant aspects of OMA CPM 2.0. In addition, extensions described in A Session Description Protocol (SDP) Offer/Answer Mechanism to Enable File Transfer or IETF RFC 5547 are taken into account.

3.7 Content Sharing

See GSMA PRD RCC.07 (RCS 5.1) ¶ 3.6. Content sharing is the ability for users to exchange different types of content while in a session, such as a voice call, or as a standalone activity. There can be different sources for the shared content:

- The front camera (“me”).
- The rear camera (“what I see”).
- A file (“video streaming” or “sending of a stored image”).

3.7.1 Relevant Standards

GSMA PRDs IR.74 defines video sharing within a call while GSMA PRD IR.84 defines standalone video sharing (i.e., outside of a call). GSMA PRD IR.79 specifies still image sharing during a call.

3.8 Geolocation Services

See GSMA PRD RCC.07 (RCS 5.1) ¶ 3.10. Geolocation services as part of GSMA RCS comprise the following 2 features:

1. The “Geolocation PUSH” service that allows an RCS user to push location information (that can be the user location or the location of a suggested meeting point) to another RCS user.
2. The “Geolocation PULL” service that allows an RCS user to retrieve the location information about another RCS user.

As it will describe in the next section, geolocation information may be provided as part of exchange of Social Presence information between RCS users. In addition, Geolocation Push is based on the RCS File Transfer service when used during a voice call and based on the RCS Chat service. The “show-us-on-map” feature makes use of Geolocation Push in the context of Group Chat. Geolocation Pull takes advantage of RCS File Transfer to fetch location information.

3.8.1 Relevant Standards

RCS Geolocation Services are described in GSMA PRD RCC.07.

3.9 Presence

Presence is a service that stores, accepts and distributes information on whether someone is available to communicate over a network. Presence can be implemented such that it is transparent to the end user (e.g., to indicate the advanced communication services supported by the device owned by someone in the user’s contact list). It can also be made accessible to the end user, e.g., by allowing the user to indicate his willingness to communicate, either directly or through integration with other applications. The latter is referred to in RCC.07 (RCS 5.1) ¶ 3.7 as Social Presence.

Rich presence is an enhanced form of presence awareness in which participants can determine if other users are online and if so, observe to a limited extent what they are doing and how they are doing it. Basic presence services divulge only the availability of another user. Rich presence goes further; subscribers can let others know for example:

- their location;
- whether their device is mobile;
- its specifications;
- operating system;
- local time;
- personal messages;

- the level of privacy desired; and
- whether the person is typing.

3GPP TR 23.841 (Technical Report) Presence service; Architecture and Functional Description:3GPP definition: The presence service provides the ability for the home network to manage presence information of a user's device, service or service media even whilst roaming. A user's presence information may be obtained through input from the user, information supplied by network entities or information supplied by elements external to the home network. Consumers of presence information, watchers, may be internal or external to the home network.

3.9.1 Relevant Standards

The lead industry standards in this space are based on IETF work and include Session Initiation Protocol for Instant Messaging and Presence Leveraging Extensions (SIMPLE) (RFC 3265, RFC 3856, RFC 3903) and Extensible Messaging and Presence Protocol (XMPP) (RFC 6120, RFC 6121, RFC 6122, RFC 3922, RFC 3923). These solutions have then been further advanced by organizations like OMA and XMPP Standards Foundation to facilitate greater interworking and the creation of modular network functions. There however remains significant technical variety between Presence solutions in deployment, with many solutions remaining closed implementations.

Operators have options to deploy Presence as a component capability for a range of multi-media services. This is the case, for example, for the GSMA Rich Communication Services as specified in GSMA PRD RCC.07, which builds on the Presence capabilities defined in OMA Presence SIMPLE. In GSMA RCS, Presence capabilities are applied not only for the exchange of Social Presence information described before in the section, but also used as mechanism for exchange of capability information and as alternative to the SIP OPTIONS capability discovery mechanism.

3.10 Video Conference/Collaboration

See GSMA PRD IR.39. The High Definition Video Conference (HDVC) service comprise point-to-point video calls and video conferences with one full duplex audio stream with tight synchronization to one main video stream and another video stream aimed for sharing of example presentation slides.

3.10.1 Relevant Standards

GSMA PRC IR.39 defines IMS-based High Definition Video Conference (HDVC) service.

3.11 Social Networking

A social networking service is an online service that focuses on facilitating the building of social networks or social relations among people who, for example, share interests, activities, backgrounds, or real-life connections. A social network service consists of a representation of each user (often a profile), his/her social links, and a variety of additional services. Most social network services are web-based and provide means for users to interact over the Internet, such as e-mail and instant messaging.

3.11.1 Relevant Standards

Social networking incorporates capabilities described in this document such as Presence, Location, Messaging, File Transfer, and Content Sharing. Some of the social networking services are proprietary and use proprietary protocols. Other social networking services are open and use standardized protocols. The set of standardized protocols utilized varies among the various social network services.

4 Use Case Analysis

4.1 HD Voice

4.1.1 Direct Peering, Transcoder Free

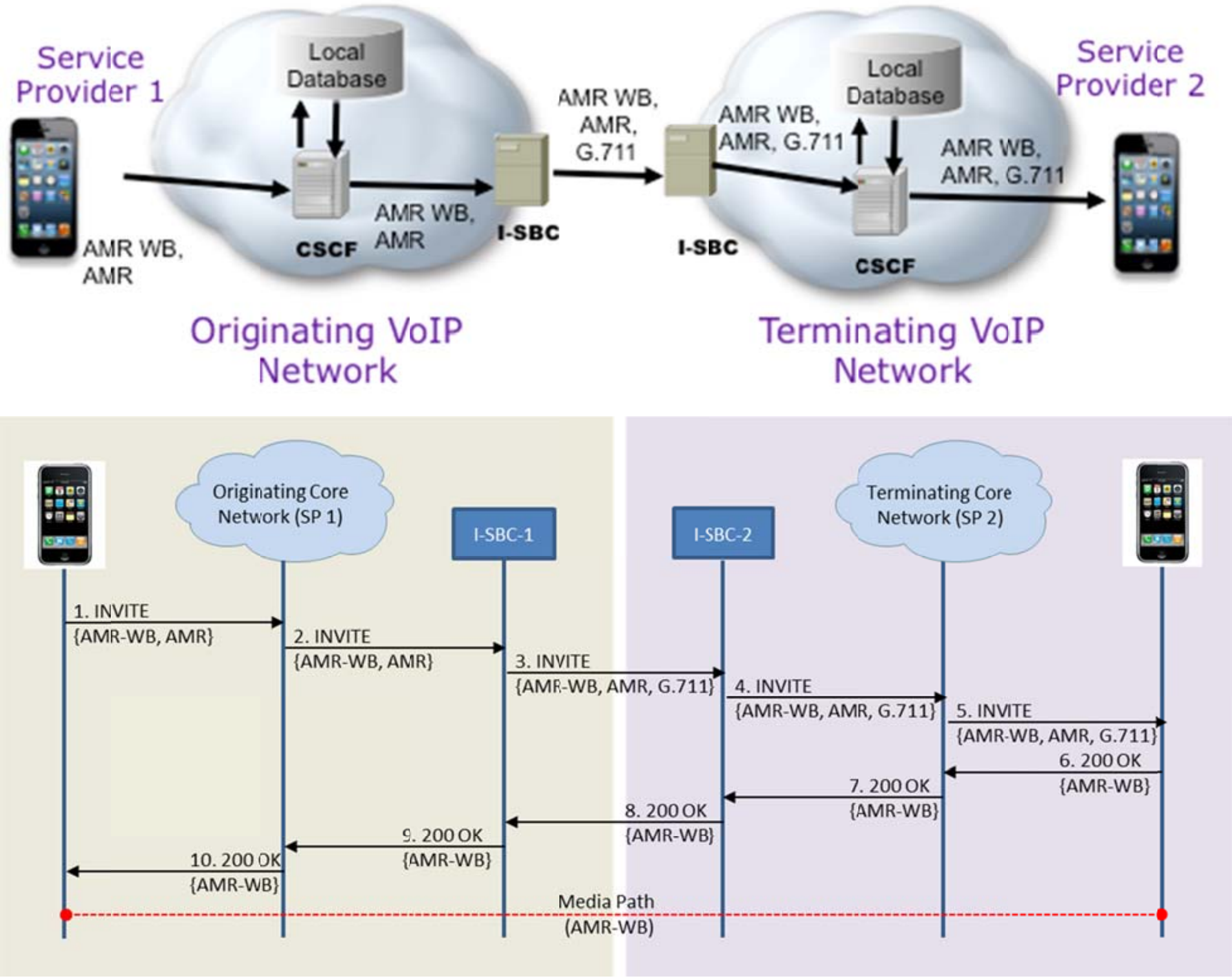

Figure 4. 1 - Direct Peering, Transcoder Free

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1. VoIP device in Service Provider 1 network originates a call request, indicating support for the AMR Wideband (AMR-WB) and AMR narrowband (AMR) codecs, in that order. This indication is referred to as the "offer".
2. Service Provider 1 network provides originating services for the calling party, then determines that the call request should be routed to Service Provider 2. It determines an egress point from its own network (I-SBC-1) through which to forward the call request, to reach a desired ingress point to Service Provider 2's network (I-SBC-2).
3. When the call request reaches the egress point from Service Provider 1's network (I-SBC-1), that network element may augment the "offer" with additional codecs, based on the interconnect agreement between Service Provider 1 and Service Provider 2. Such codecs, if added, are indicated to be less preferable than the ones originally offered by the calling device. In this case, assume that it added G.711. It also allocates whatever

resources are required to perform transcoding, should transcoding prove to be necessary.

4. When the call request reaches the ingress point to Service Provider 2's network (I-SBC-2), that network element may augment the "offer" with additional codecs, based on the policies of Service Provider 2. As above, if codecs are added, they are indicated to be less preferable than the ones already in the offer. In this case, assume that it added nothing.
5. Service Provider 2 network provides terminating services for the called party, then forwards the call request to the device(s) with which the called number is registered. In this case, assume there is one such device (this is usually the case).
6. The terminating device selects the first codec in the offer that it is able to support. Assume that to be AMR-WB. It generates an "answer" which indicates that it wants to use this codec.
7. The answer is propagated back toward the calling device. (Includes Step 8)
9. When the answer reaches I-SBC-1, that network element realizes that transcoding is not required and therefore releases the transcoding resources it allocated in step 3.
10. When the answer reaches the calling device, an end-to-end RTP path is established over which the audio media is exchanged in the AMR-WB format.

4.1.2 Direct Peering Requiring Transcoding

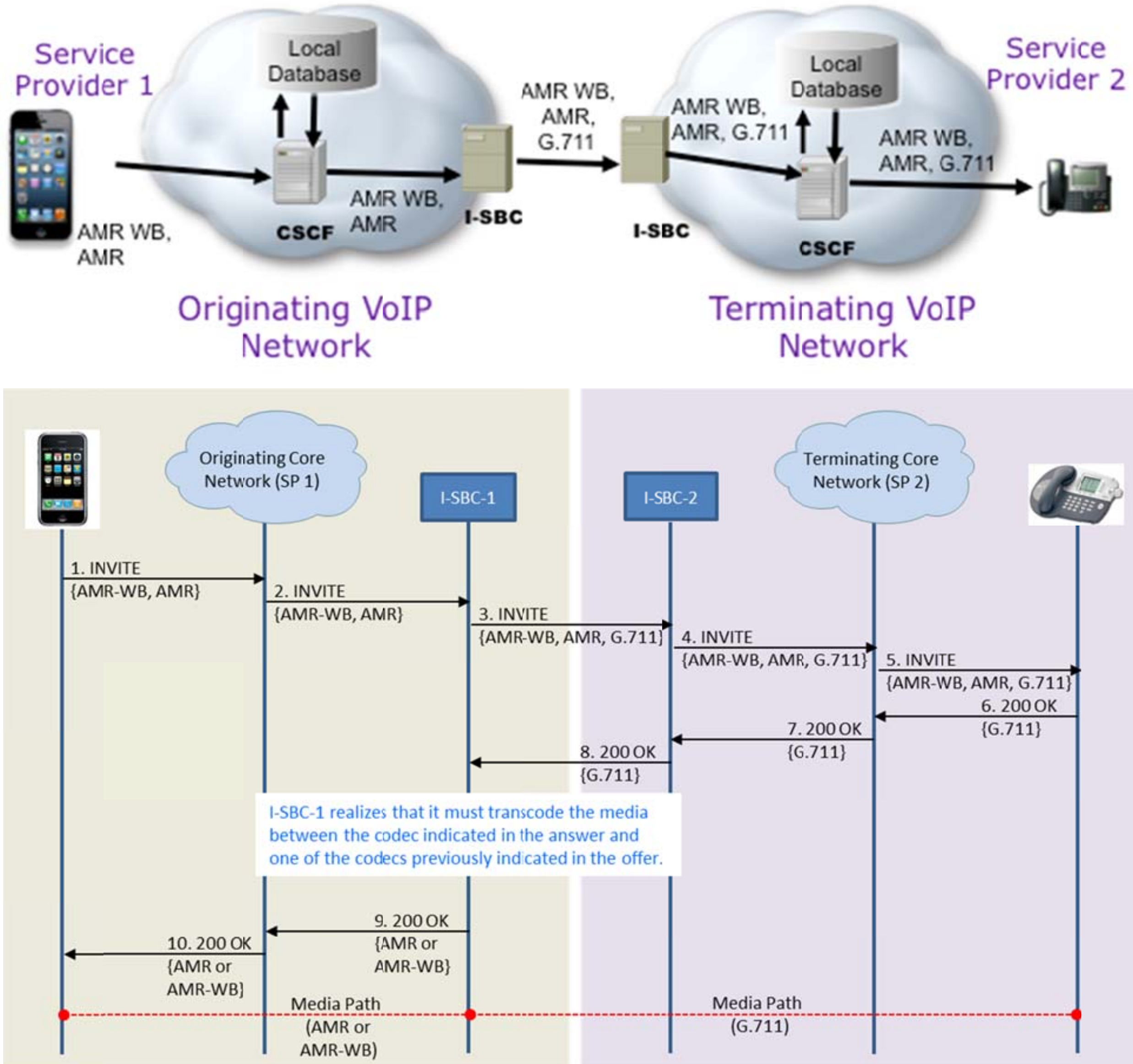


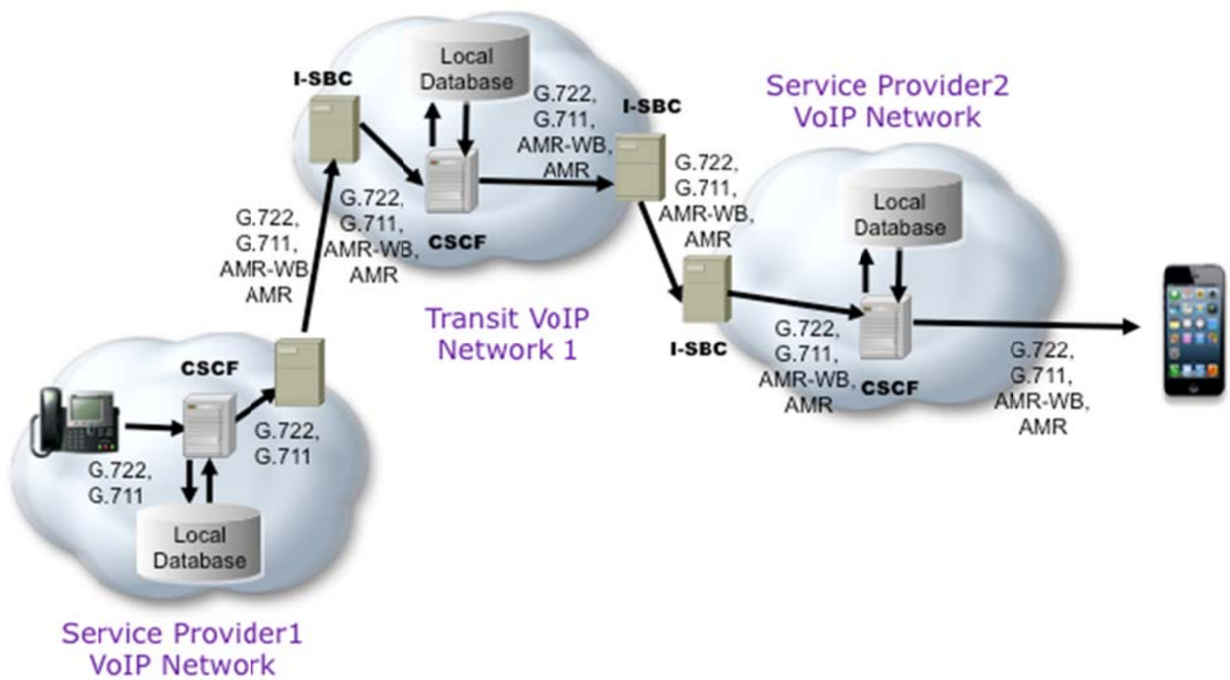
Figure 4. 2 - Direct Peering – Transcode AMR or AMR-WB to G.711

1. VoIP device in Service Provider 1 network originates a call request, indicating support for the AMR and AMR-WB codecs. This indication is referred to as the “offer”.
2. Service Provider 1 network provides originating services for the calling party, then determines that the call request should be routed to Service Provider 2. It determines an egress point from its own network (I-SBC-1) through which to forward the call request, to reach a desired ingress point to Service Provider 2’s network (I-SBC-2).
3. When the call request reaches the egress point from Service Provider 1’s network (I-SBC-1), that network element may augment the “offer” with additional codecs, based on the interconnect agreement between Service Provider 1 and Service Provider 2. Such codecs, if added, are indicated to be less preferable than the ones originally offered by the calling device. In this case,

assume that it added G.711. It also allocates whatever resources are required to perform transcoding, should transcoding prove to be necessary.

4. When the call request reaches the ingress point to Service Provider 2's network (I-SBC-2), that network element may augment the "offer" with additional codecs, based on the policies of Service Provider 2. As above, if codecs are added, they are indicated to be less preferable than the ones already in the offer. In this case, assume that it added nothing.
5. Service Provider 2 network provides terminating services for the called party, then forwards the call request to the device(s) with which the called number is registered. In this case, assume there is one such device (this is usually the case).m
6. The terminating device selects the last codec (G.711) since it is unable to support either AMR or AMR-WB. It generates an "answer" which indicates that it wants to use this codec.
7. The answer is propagated back toward the calling device. (Includes Step 8)
9. When the answer reaches I-SBC-1, that network element realizes that the codec indicated in the answer is one that it added to the offer. It therefore realizes that it needs to transcode the media between the codec selected by the called device (G.711) and one of the codecs offered by the calling device. Depending on the policy of SP 1, it may select either AMR or AMR-WB.
10. When the answer reaches the calling device, an end-to-end RTP path is established over which the audio media is exchanged. Between the calling device and I-SBC-1 the media is exchanged in AMR or AMR-WB format. Between I-SBC-1 and the called device the media is exchanged in G.711 format.

4.1.3 InDirect Peering, Transcoding in the Originating Network



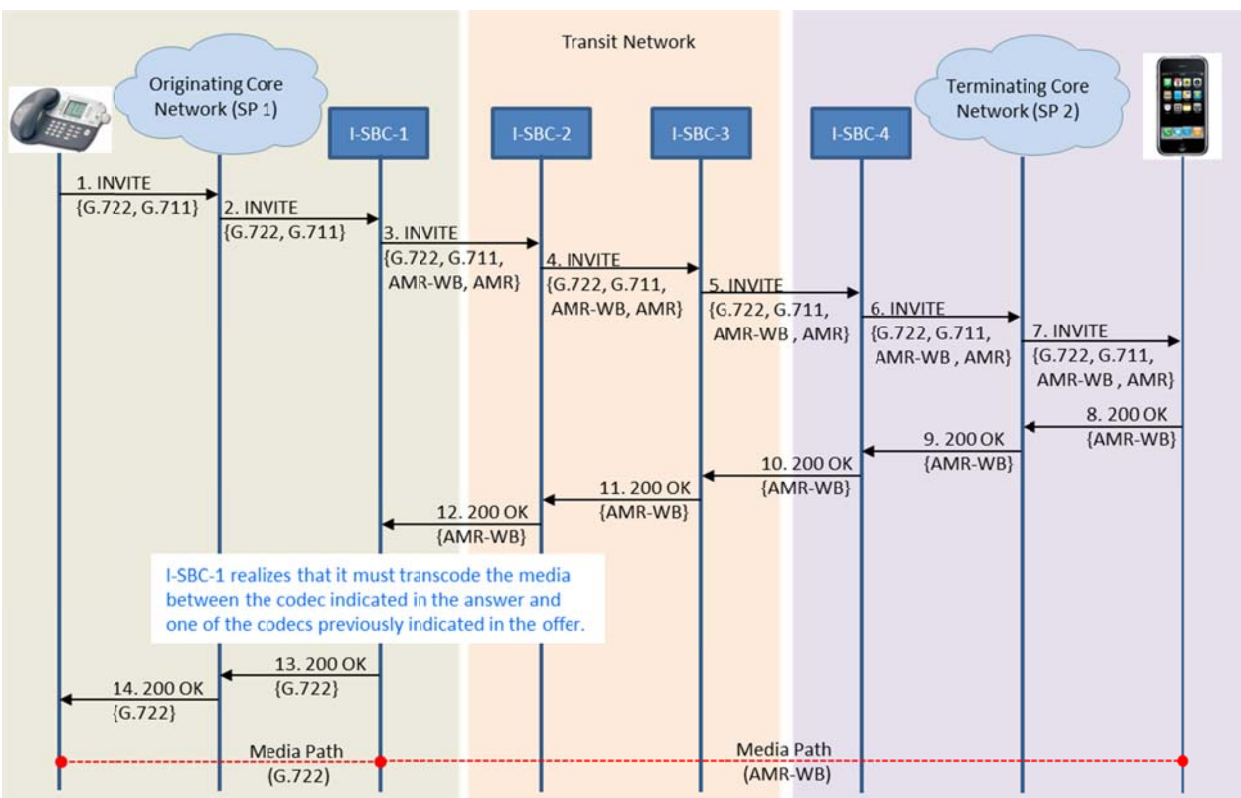


Figure 4. 3 - InDirect Peering, Transcoding in the Originating Network

1. VoIP device in Service Provider 1 network originates a call request, indicating support for the G.722 and G.711 codecs. This indication is referred to as the “offer”.
2. Service Provider 1 network provides originating services for the calling party, then determines that the call request should be routed to Service Provider 2 via a transit network. (This use case assumes there to be an interconnect agreement between SP1 and SP2.) It determines an egress point from its own network (I-SBC-1) through which to forward the call request, to reach a desired ingress point to the transit network (I-SBC-2).
3. When the call request reaches the egress point from Service Provider 1’s network (I-SBC-1), that network element may augment the “offer” with additional codecs, based on the interconnect agreement between Service Provider 1 and Service Provider 2. Such codecs, if added, are indicated to be less preferable than the ones originally offered by the calling device. In this case, assume that it added AMR-WB and AMR. It also allocates whatever resources are required to perform transcoding, should transcoding prove to be necessary.
4. When the call request reaches the ingress point to the transit network, no SDP manipulation is required since the transit network does not terminate the media.
5. When the call request reaches the egress point from the transit network facing Service Provider 2 (I-SBC-2), that network element may augment the “offer” with additional codecs, based on the interconnect agreement between the transit provider and Service Provider 2. As above, if codecs are added, they are indicated to be less preferable than the ones already in the offer. In this case assume that it added nothing.
6. When the call request reaches the ingress point to Service Provider 2’s network (I-SBC-3), that network element may augment the “offer” with additional codecs, based on the policies of Service Provider 2. As above, if codecs are added, they are indicated to be less preferable than the ones already in the offer. In this case, assume that it added nothing.
7. Service Provider 2 network provides terminating services for the called party, then forwards the call request to the device(s) with which the called number is registered. In this case, assume there is one such device (this is usually the case).
8. The terminating device selects the AMR-WB codec since that is the first codec in the offer that it supports. It generates an “answer” which indicates that it wants to use this codec.
9. The answer is propagated back toward the calling device. (Includes Steps 10-12)
13. When the answer reaches I-SBC-1, that network element realizes that the codec indicated in the answer is one that it added to the offer. It therefore realizes that it needs to transcode the media between the codec selected by the called device (G.711) and one of the codecs offered by the calling device. By selecting a wideband codec (G.722), I-SBC-1 can ensure an end-to-end wideband media path. As before, the codec it selects is determined by the policies of SP 1. In this case, assume it selected G.722.
14. When the answer reaches the calling device, an end-to-end RTP path is established over which the audio media is exchanged. Between the calling device and I-SBC-1 the media is exchanged in G.722 format. Between I-SBC-1 and the called device the media is exchanged in AMR-WB format.

4.1.4 InDirect Peering, Transcoding in both Originating and Transit Networks

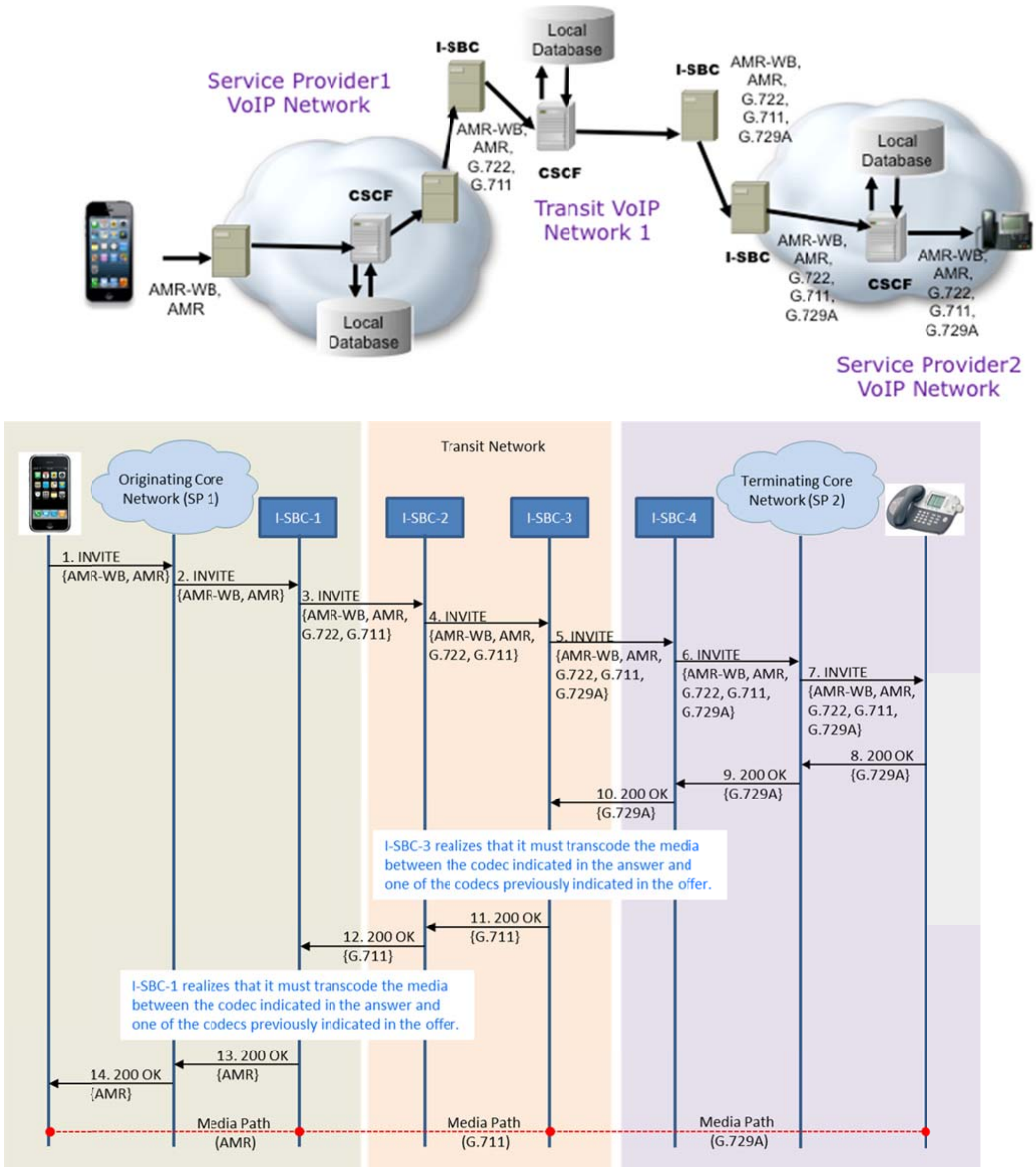


Figure 4. 4 - InDirect Peering – Transcoding in both Originating and Transit Networks

1. VoIP device in Service Provider 1 network originates a call request, indicating support for the AMR-WB and AMR codecs. This indication is referred to as the “offer”.
2. Service Provider 1 network provides originating services for the calling party, then determines that the call request should be routed to a transit network. (This use case assumes that the originating network does not know or has no interconnect agreement with, the terminating network) It determines an egress point from its own network (I-SBC-1) through which to forward the call request, to reach a desired ingress point to the transit network (I-SBC-2).
3. When the call request reaches the egress point from Service Provider 1’s network (I-SBC-1), that network element may augment the “offer” with additional codecs, based on the interconnect agreement between Service Provider 1 and the transit network. Such codecs, if added, are indicated to be less preferable than the ones originally offered by the calling device. In this case, assume that it added G.722 and G.711. It also allocates whatever resources are required to perform transcoding, should transcoding prove to be necessary.
4. When the call request reaches the ingress point to the transit network, no SDP manipulation is required since the transit network does not terminate the media. The transit network determines that the call request is destined for Service Provider 2, and identifies an egress point from its own network (I-SBC-3) through which to forward the request, to reach a desired ingress point to the terminating network (I-SBC-4).
5. When the call request reaches the egress point from the transit network facing Service Provider 2 (I-SBC-3), that network element may augment the “offer” with additional codecs, based on the interconnect agreement between the transit provider and Service Provider 2. As above, if codecs are added, they are indicated to be less preferable than the ones already in the offer. In this case assume that it added G.729A.
6. When the call request reaches the ingress point to Service Provider 2’s network (I-SBC-4), that network element may augment the “offer” with additional codecs, based on the policies of Service Provider 2. As above, if codecs are added, they are indicated to be less preferable than the ones already in the offer. In this case, assume that it added nothing.
7. Service Provider 2 network provides terminating services for the called party, then forwards the call request to the device(s) with which the called number is registered. In this case, assume there is one such device (this is usually the case).
8. The terminating device selects the G.729A codec since that is the first codec in the offer that it supports. It generates an “answer” which indicates that it wants to use this codec.
9. The answer is propagated back toward the calling device.
10. When the answer reaches I-SBC-3, that network element realizes that the codec indicated in the answer is one that it added to the offer. It therefore realizes that it needs to transcode the media between the codec selected by the called device (G.711) and one of the codecs that was in the offer it received in step 4. The codec it selects is determined by the policies of the transit network provider. In this case, assume it selected G.711. It is possible, but not mandatory, for SBC3 to accept the first codec offered in order to avoid additional transcoding (this includes Step 11).
12. When the answer reaches I-SBC-1, that network element realizes that the codec indicated in the answer is one that it added to the offer. It therefore realizes that it needs to transcode the media between the codec it sees in the answer (G.711) and one of the codecs that was in the offer it received in step 2. The codec it selects is determined by the policies of SP 1. In this case, assume it selected AMR.
13. When the answer reaches the calling device, an end-to-end RTP path is established over which the audio media is exchanged. Between the calling device and I-SBC-1 the media is exchanged in AMR format. Between I-SBC-1 and I-SBC-3 the media is exchanged in G.711 format. Between I-SBC-3 and the called device the media is exchanged in G.729A format. (this includes Step 14).

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4.1.5 Direct Peering – Transcoding OPUS to AMR-WB

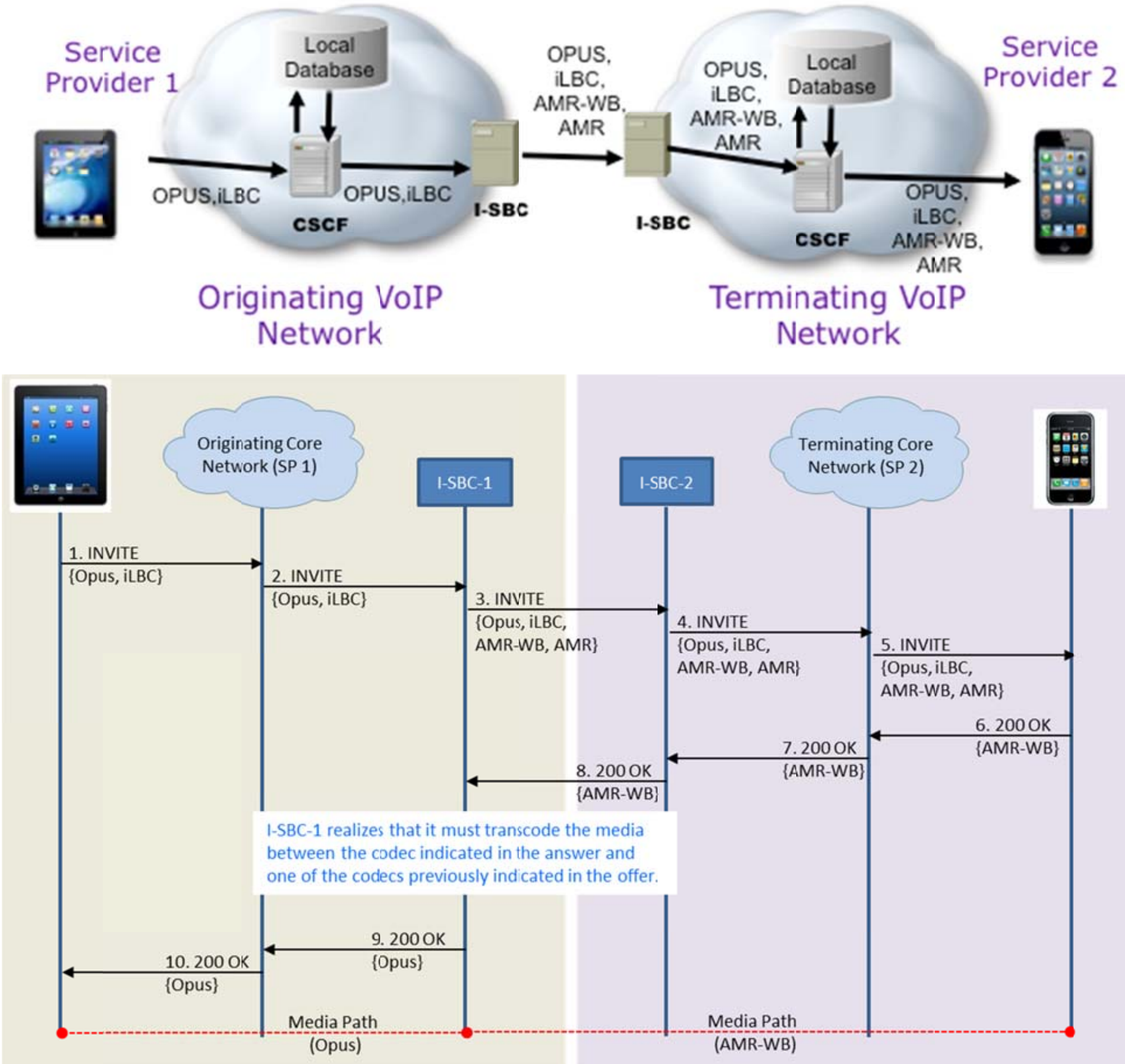

Figure 4. 5 - Direct Peering – Transcoding Opus to AMR-WB

1. VoIP device (e.g., Tablet supporting webRTC) in Service Provider 1 network originates a call request, indicating support for the Opus and iLBC codecs. This indication is referred to as the "offer".
2. Service Provider 1 network provides originating services for the calling party, then determines that the call request should be routed to Service Provider 2. It determines an egress point from its own network (I-SBC-1) through which to forward the call request, to reach a desired ingress point to Service Provider 2's network (I-SBC-2).
3. When the call request reaches the egress point from Service Provider 1's network (I-SBC-1), that network element may augment the "offer" with additional codecs, based on the interconnect agreement between Service Provider 1 and Service Provider 2. Such codecs, if added, are indicated to be less preferable than the ones originally offered by the calling device. In this case, assume that it added AMR-WB and AMR. It also allocates whatever resources are required to perform transcoding, should transcoding prove to be necessary.

4. When the call request reaches the ingress point to Service Provider 2's network (I-SBC-2), that network element may augment the "offer" with additional codecs, based on the policies of Service Provider 2. As above, if codecs are added, they are indicated to be less preferable than the ones already in the offer. In this case, assume that it added nothing.
5. Service Provider 2 network provides terminating services for the called party, then forwards the call request to the device(s) with which the called number is registered. In this case, assume there is one such device (this is usually the case).
6. The terminating device selects the AMR-WB codec since that is the first codec in the offer that it supports. It generates an "answer" which indicates that it wants to use this codec.
7. The answer is propagated back toward the calling device.
8. When the answer reaches I-SBC-1, that network element realizes that the codec indicated in the answer is one that it added to the offer. It therefore realizes that it needs to transcode the media between the codec selected by the called device (AMR-WB) and one of the codecs offered by the calling device. Depending on the policy of SP 1, it may select either Opus or iLBC. In this case assume it selected Opus (since doing so yields an end-to-end wideband audio stream). (Includes Step 9.)
10. When the answer reaches the calling device, an end-to-end RTP path is established over which the audio media is exchanged. Between the calling device and I-SBC-1 the media is exchanged in Opus format. Between I-SBC-1 and the called device the media is exchanged in AMR-WB format.
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4.1.6 SDO Assessment/Gap Analysis

4.1.6.1 3GPP

4.1.6.1.1 Session Establishment & Signaling

3GPP TS 24.229 defines the functions at the SIP protocol layer for an Interconnection Border Control Function (IBCF). IBCFs communicate with each other to perform interconnection between IP Multimedia networks. Trust relationships and trust domains may be defined by inter-operator agreements for individual services and/or individual SIP header fields. The offer/answer model with the SDP as defined in IETF RFC 3264 is used for codec negotiation and selection.

4.1.6.1.2 Addressing & Routing

3GPP TS 24.229 defines URI formats that *must* be supported on the IP-NNI. Additional routing capabilities are necessary for support of transit, roaming, and interconnection traffic. Routing procedures should analyze the destination address and determine whether to route to another network, directly, or via the IBCF, or to the BGCF, or the I-CSCF in its own network. This analysis may use public (e.g., DNS, ENUM) and/or private database lookups, and/or locally configured data and may, based on operator policy, modify the Request-URI (e.g., to remove number prefixes, to translate local numbers to global numbers, etc).

4.1.6.1.3 Media Handling

Codecs applicable at the IP-NNI may be a subject of interworking agreements. To avoid the scenario where transcoding is performed several times, applicable codecs at the NNI should be restricted as little as possible in the inter-operator agreements. Transcoding can be performed in an IMS network serving an SDP offerer or in an IMS network serving an SDP answerer. Inter-operator agreements can clarify if it is preferred that the IMS network serving an SDP offerer or IMS network serving an SDP answerer modify an SDP offer to offer media transcoding. Optimal Media Routing (OMR) may be used to identify and remove unnecessary media functions from the media path for each media stream associated with a session, to minimize end-to-end delay and to minimize the use of transport network resources. OMR procedures should be agreed between operators.

4.1.6.2 GSMA

4.1.6.2.1 Session Establishment & Signaling

GSMA PRD IR.92 as IMS profile for Voice of LTE defines that the "IMS well-known APN" is required for the establishment of the PDN connection needed to set up a VoLTE session. The IMS well-known APN is defined in GSMA PRD IR.88.

Within the SP network, if the PDN connection established during the initial attach is to an APN other than the IMS well-known APN, then prior to registering with IMS the UE must establish another PDN connection to the IMS well-known APN.

A default bearer must be created when the UE creates the PDN connection to the IMS well known APN, as defined in 3GPP TS 23.003. A standardized QCI value of five (5) must be used for the default bearer. It is used for IMS SIP signaling.

UE and IMS core network follow 3GPP TS 24.229 for establishment and termination of a VoLTE call. For the purpose of indicating an IMS communication service to the network, the UE must use an ICSI value in accordance with 3GPP TS 24.229. The ICSI value used must indicate the IMS Multimedia Telephony (MMTel) service, which is urn:urn-7:3gpp-service.ims.icsi.mmtel, as specified in 3GPP TS 24.173.

A VoLTE UE must be ready to receive responses generated due to a forked request and behave according to the procedures specified in IETF RFC 3261, 3GPP TS 23.228, and 3GPP TS 24.229.

In addition, if a VoLTE UE detects that in-band information is received from the network as early media, the in-band information received from the network shall override locally generated communication progress information as described in 3GPP TS 24.628.

Similar functionality is endorsed in GSMA PRD IR.58, the IMS profile for Voice over HSPA.

4.1.6.2.2 Addressing & Routing

GSMA PRD IR.65 defines network-level roaming and interworking guidelines for IMS services such as for Voice over IMS (i.e., GSMA PRDs IR.92 and IR.58). In addition, GSMA PRD IR.67 provides recommendations on DNS (including ENUM) to facilitate successful interworking of inter-Service Provider services.

GSMA PRD IR.65 endorses interworking by means of the IMS NNI as documented in 3GPP TS 29.165, and the support of IPv4 only, IPv6 only or both. Support of the different IP versions on the Inter-Service Provider IP Backbone network (e.g., IPX) is specified in GSMA PRD IR.34 and GSMA PRD IR.40.

IMS user addressing is defined in 3GPP TS 23.228 and its format is defined in 3GPP TS 23.003. GSMA PRD IR.92 further clarifies that UEs and IMS core network must support Public User Identities in the form of SIP URIs (both alphanumeric and those representing Mobile Subscriber ISDN Numbers (MSISDNs)) and Tel URIs. Support of MSISDN as a Public User Identity, the network must associate a Tel URI with an alphanumeric SIP URI using the mechanisms specified in TS 23.228 and TS 24.229.

Support of IP addressing used by the user is subject to the requirements for the underlying IP network, e.g., for LTE, see GSMA PRD IR.88, respectively.

As such, GSMA PRD IR.65 specifies that Roaming should be at least as optimal as that of the Circuit Switched domain.

The Inter-Service Provider IP Backbone should support OMR functionality as specified in 3GPP TS 29.079, if it is allowed between two operators to prevent the user plane to be routed back to HPLMN of roaming users.

4.1.6.2.3 Media Handling

GSMA PRD IR.92 specifies that the UE must support the Adaptive Multi-Rate (AMR) speech codec, as described in 3GPP TS 26.071, TS 26.090, TS 26.073, and TS 26.104, including all eight modes and source rate controlled operations, as described in 3GPP TS 26.093. The UE must be capable of operating with any subset of these eight codec modes.

If wideband speech communication is offered, the UE must support AMR wideband codec as described in 3GPP TS 26.114, TS 26.171, TS 26.190, TS 26.173, and TS 26.204, including all nine modes and source controlled rate operation 3GPP TS 26.193. The UE shall be capable of operating with any subset of these nine codec modes.

Entities in the IMS core network that terminate the user plane in support of Circuit Switched interworking and apply TFO and/or TrFO shall support:

- AMR speech codec modes described in 3GPP TS 26.071, TS 26.090, TS 26.073, and TS 26.104.

Entities in the IMS core network that terminate the user plane supporting wideband speech communication and supporting TFO and/or TrFO shall support:

- AMR-WB speech codec modes described in 3GPP TS 26.171, TS 26.190, TS 26.173, and TS 26.204.

IR.92 endorses media capabilities specified in 3GPP TS 26.114. GSMA PRD IR.92 also describes the needed SDP support in UEs and in the IMS core network and it describes the necessary media capabilities both for UEs and for entities in the IMS core network that terminate the user plane.

4.1.6.3 PacketCable

4.1.6.3.1 Session Establishment & Signaling

There are two versions of PacketCable™: PacketCable 1.5 and PacketCable 2.0. PacketCable 1.5 is a Softswitch-based architecture. SIP session establishment signaling is defined in PKT-SP-CMSS1.5-I07-120412, which in turn follows the IETF SDP offer/answer procedures defined in RFC 3264. PacketCable 1.5 supports G.722 as defined in PKT-SP-CODEC1.5-I04-120412.

PacketCable 2.0 is an IMS-based architecture that supports the session signaling procedures defined in 24.229 (that ultimately support the IETF SDP offer/answer procedures in RFC 3264). PacketCable 2.0 HD-voice extensions are defined in PKT-SP-HDV-SIP-I03-120823 and PKT-SP-CODEC-MEDIA-C01-140314.

PacketCable NNI SIP signaling procedures are defined in PKT-SP-IGS-C01-130930. It assumes a direct-IP-peering architecture (no transit network). Scope is limited to basic narrow-band voice calling, plus common voice features. Use of PRACK is avoided, to minimize signaling traffic, and since the frequency of lost messages in today's networks is very low. QoS preconditions are not used, to minimize post-dial ringing delay, and since local policy normally dictates that media session is established via best-effort when QoS allocation fails.

Summary: PacketCable is similar to 3GPP in that it supports the session establishment procedures described in 24.229, and ultimately RFC 3264. In general, PacketCable does not adopt many of the private P_ headers defined for IMS (e.g., does not support P-Early-Media, P-Asserted-Service-ID, etc).

4.1.6.3.2 Addressing & Routing

Similar to 3GPP at the session-signaling level.

PKT-SP-ENUM-PROV defines the architecture and procedures for sharing routing information between peering partners (i.e., how a service provider pushes its ingress routing information to a peer, the data models for the peer routing registry, and how a provider can add their egress routing information to the ingress routing info received from a peer).

PKT-SP-ENUM-SRV defines the procedures used in the originating network for obtaining peer routing information at call setup time (i.e., procedures for querying the registry routing database. It defines two options; one based on ENUM, and one based on SIP (i.e., using SIP redirect). The ENUM procedures are similar to what is defined in 24.229, but with more detail.

> NOTE: The above specifications are not currently supported in deployed cable networks. In the context of this document, these specifications should be viewed simply as informative resources describing one of many possible NNI routing solutions.

4.1.6.3.3 Media Handling

PKT-SP-IGS-C01-130930 defines procedures for early media, including the case of sequential forking in terminating network; say where, before 200OK-answer, call is routed to media server to collect PIN, then routed to called user to receive custom ringback tone, etc).

Currently only wideband codec mandated is G.722. The expectation is that additional wideband codecs will be added to the list in the future (e.g., OPUS).

4.1.6.4 I3forum

The i3forum document “Enabling HD voice continuity in international calls” Release 1.0 addresses a broad set of high definition voice codecs including but not limited to the codecs dealt with by GSMA with its HD Voice specification: i.e., G.722 for fixed communication and G.722.2 (WB-AMR) for mobile communication.

NOTE: i3forum only addresses indirect interconnection for international voice.

4.1.6.4.1 Session Establishment & Signaling

Ordinarily Media negotiation use the offer-answer model (RFC 3264): the offer is contained in the SDP body of the initial INVITE, and, given that the call is successfully established, the answer is in the body of the 200 OK message.

- For TrFO, follows TS 23.153 where wideband codecs, if available, should be given preference in the codec negotiation.
- For AMR-WB signaling, all of the control parameters are mapped in the SDP body of the SIP messages. They are included in the a=fmtp attribute of the AMR-WB codec.

Transit carrier could be “Proactive” (involved in the Codec negotiation) or “Passive” (not involved) but in Proactive Mode the Carrier should not intervene if:

- SDP body with a distinctive attribute a=3gOoBTC, which conveys a SIP session establishment using the TrFO procedure is being negotiated by the SPs.
- There are cconnectivity preconditions (IETF RFC 5898). The terminating user will not be alerted unless end-to-end connectivity has been previously ensured.
- There are QoS preconditions (IETF RFC 3312). The terminating user will not be alerted unless QoS parameters can be guaranteed.
- There are Security preconditions (IETF RFC 5027). The terminating user will not be alerted unless a secure communication between endpoints can be guaranteed.

4.1.6.4.2 Addressing & Routing

I3 Forum White Paper Techniques for Carrier's Advanced Routing and Addressing Schemes (Release 1.0 May 2010 and Release 2.0 May 2011) offers a strategy for the evolution of routing and addressing taking into account the realities of the current environment. This strategy focuses on leveraging existing and planned service provider efforts to create registries for number portability corrected data and supplementing these to meet the particular needs of international carriers.

Routing: Rarely takes into account the media capabilities of the Service Providers for the configuration of its routing policy.

4.1.6.4.3 Media Handling

Non TrFO Indirect scenarios analyzed by the i3F include:

- Preconditions.
- The offer/answer are transported in the SIP INVITE/OK messages respectively.
- There is only "audio" as media type in the m= line(s).
- No ReINVITEs that modify the media session are sent in the middle of the call.
- No SIP Preconditions are used, or they have already been met.
- The IPX Provider has transcoding capabilities.
- Scenarios:
    - Session initiated by a SP offering only mandatory wideband codecs (G.722/AMR-WB).
    - The terminating SP accepts one or both the wideband codecs present in the offer.
    - The terminating SP only accepts narrowband codecs.
    - The terminating SP supports one or more wideband codecs, but different from the ones contained in the offer.
- Session initiated by a SP offering mandatory wideband codecs.
- Session initiated by a SP offering only Opus.
    - The terminating SP accepts Opus.
    - The terminating SP accepts only AMR-WB or G.722, not Opus.
    - The terminating SP accepts other codecs.
- Other scenarios.
    - The initiating SP only offers non-mandatory high quality codecs.

4.1.6.5 ATIS

4.1.6.5.1 Session Establishment & Signaling

Currently under discussion in PTSC in their Issue S0040, IP Network-to-Network Interface, Phase 2), Interconnection parameters are based on 3GPP TS 24.229. It describes the procedures the BCSF shall use for both the trusted and non-trusted case. Support for SCTP is optional.

4.1.6.5.2 Addressing & Routing

The following URI formats are supported:

- SIP URI defined in IETF RFC 3261 [13];
- tel URI, or SIP URI with tel URI parameters, as defined in IETF RFC 3966 [14];
- IM URI defined in IETF RFC 3860 [15]; and
- PRES URI defined in IETF RFC 3859 [16].

Interworking may be IPv4, IPv6, or both.

Currently under discussion in PTSC in their Issue S0040, IP Network-to-Network Interface, Phase 2 (Interconnection Models) does not indicate a preference for a particular interconnection model, but warns of the dangers of routing the signaling and media separately. RAVEL and OMR are identified as possible alternative routing strategies when roaming.

ATIS-0416001-0001[2] (Next Generation Networks Carrier Interconnection Use Cases for Ordering) supports both direct and indirect interconnect use cases.

A joint ATIS SIP Forum task force is developing the IP Interconnection Routing Document. The document discusses the existing in-use and proposed routing solutions to facilitate the exchange of traffic associated with IP-based services between North American service providers.

4.1.6.5.3 Media Handling

Codecs applicable at the II-NNI may be a subject of interworking agreements, but the list of possible codecs are found in 3GPP TS 26.114 and ETSI TS 181 005. AMR-WB (G.722.2), G.722, EVRC-WB are supported wideband codecs.

The IBCF, the MRFC, or other IMS network entities may do transcoding.

Media security may be none (physical security), IPsec, or SRTP.

4.1.6.6 ITU

4.1.6.6.1 Session Establishment & Signaling

ITU-T Recommendation Q.3401 NGN NNI signalling profile (protocol set 1) defines a SIP profile for the service control function at the NNI interface based on IETF RFC 3261.

4.1.6.6.2 Addressing & Routing

There can be several ways for determining the destination of the SIP message based on the tel URI; ENUM-based routing, number-based routing. In the ITU Recommendation, it is assumed that each network knows the address information of the peer network based on the number.

4.1.6.6.3 Media Handling

- Media availability in a SIP session: The terminating-side network of the NNI shall pass any media packets in the direction toward the originating party as soon as they are available.
- The originating-side network of the NNI:
    - shall pass media packets from the originating party in the direction toward the terminating party upon and after receiving a final SDP answer within a SIP 2xx response to the INVITE for normal dialog; may pass media packets from the originating party in the direction toward the terminating party as early as the first SDP answer has occurred. A network, as a policy, may choose not to send media packets from the originating party until the final SDP offer/answer has been made in order to avoid theft-of-service in cases where usage-sensitive billing is employed.

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[2] This document is available from the Alliance for Telecommunications Industry Solutions (ATIS) at <https://www.atis.org/docstore/product.aspx?id=28154>.

Codec support:

To enable the provision of voice service with a superior quality, it is highly recommended that the list contain a wideband codec such as AMR-WB [ITU-T G.722.2], VMR-WB [TIA-1016], G.722 [ITU-T G.722], G.729.1 [ITU-T G.729.1].

4.1.6.7 W3C WebRTC & IETF RTCWeb

4.1.6.7.1 Session Establishment & Signaling

Session establishment signaling is not specified although the WebRTC API creates and consumes SDP offers and answers to negotiate media capabilities for establishing sessions between browsers.

IETF Internet-Draft draft-ietf-rtcweb-jsep-06, defines extensions to SDP offer/answer to:

a) multiplex multiple media streams on same 5-tuple; i.e., transport (UDP/TCP), plus source/destination IP:port,
b) secure streams with DTLS/SRTP,
c) establish data channel, and
d) other miscellaneous enhancements.

Draft-ietf-rtcweb-jsep normatively references additional IETF draft documents, such as draft-ietf-mmusic-sdp-bundle-negotiation, that must be supported to enable the above extensions.

4.1.6.7.2 Addressing & Routing

Not Specified.

4.1.6.7.3 Media Handling

WebRTC proposes that all implementations must support OPUS for wideband voice, but still allows other codecs to be used if both sides support them, and if they are negotiated.

4.1.7 Key Issues to be Addressed by IP Services NNI

4.1.7.1 Wideband Codec Support

The more broadly supported HD codecs currently specified in standards include:

- G.722.
- G.722.2 (Adaptive Multi-rate - Wideband).
- OPUS.

As a general guideline, transcoding is to be avoided at the NNI where possible. The joint ATIS SIP Forum IP-NNI Task force has as a key objective to specify an audio codec selection strategy that minimizes the need for transcoding and a transcoding strategy that balances the workload between originating and terminating carrier. In order to provide a more seamless solution for HD voice end-to-end it is recommended that the ATIS/SIP Forum NNI Task Force address mandatory HD codecs as part of the definition of the NNI IP profile.

4.1.7.2 Support of QoS Preconditions

Preconditions are a set of constraints about a session that are introduced in the offer. The recipient of the offer generates an answer but does not alert the user or proceed with the session establishment until after the preconditions are met. Quality of Service preconditions are included in the SDP description. PacketCable and some other industry segments do not currently support the use of preconditions. It is recommended that the ATIS/SIP Forum NNI Task Force evaluate the need to address preconditions as part of the definition of the NNI IP profile, and that the same procedures be used for the other services considered in this document.

4.1.7.3 Early Media Support

Early media is the concept of delivering a media stream (e.g., audio and video) prior to call answer or session establishment. Support for early media is important both for interoperability with the Public Switched Telephone Network (PSTN) and for billing purposes. Varying implementations exist for the support of early media. In some cases early media is indicated by the presence of a P-Early-Media header in a 18X response. It is recommended that the ATIS/SIP Forum NNI Task Force define the procedures for early media across the NNI as part of the IP Interconnect profile, and that the same procedures be used for other services.

4.1.7.4 IP Interconnection Addressing Data

GSMA is exploring the concept of service identification at the NNI. The requested service needs to be clearly identified at the NNI in a fashion that allows each network to understand what it needs to provide and to determine the applicable charging, if any. It is noted that some service providers consider exposing their subscriber service adoption to be undesirable from a competitive standpoint.

In principle this can all be derived from the various headers in the SIP session request (see RFC-5897). However that derivation can be computationally complex. IETF has specified in RFC-6050 a pair of SIP headers (P-Preferred-Service and P-Asserted-Service) that may be used to encode in the SIP message the result of this derivation, so that downstream network elements need not repeat the process of “figuring out” what service is being requested. Use of these headers should be considered in future NNI work.

4.2 Video Call

4.2.1 Capability Discovery

It’s sometimes desirable to be able to discover the enhanced services supported by the device at the far end of an existing communications session; including a session that spans an NNI. Similarly, it can be useful to know prior to requesting a communications session what the remote device(s) support. For example, the following are sample applications of such mechanisms:

- Originating network can screen/update the SDP to exclude video attributes to minimize the chances of audio call failure due to far end device’s inability to handle SDP with video and audio. For example, some devices cannot handle large SDP message bodies, nor SDP message bodies containing multiple media (m=) lines.
- The UI display can be updated to reflect far end capability. For example, the contact list entry on a mobile device corresponding to a number assigned to a friend’s PC Client, can have a “video call” icon activated that in other cases (where the remote device is not known to be video capable) is greyed out.

The following are some options for discovering the services supported by other subscribers.

- Static Device Configuration (if allowed).
- NNI configuration (e.g., one network might be configured with the knowledge that the peer network does not support certain services).
- An industry database could be configured with the supported services of the far end party.
- Dynamic discovery.
    - Prior to initiation of a session request
    - After a session has been established
- Standard codec negotiation (video attributes in SDP) during session establishment.

During the IP Transition period, there may be a high proportion of advanced services that are not possible end-to-end until service adoption is more ubiquitous. It may be highly desirable to optimize the efficiency of the network with a database lookup rather than incur the extra call setup signaling if the majority of these sessions will be rejected or downgraded to basic voice. This database concept has been raised under the “Addressing and Routing” sections amongst the various use cases throughout this document. This need is not limited to video sessions.

How such an industry database is provisioned and by whom, the protocols (e.g., ENUM) used to access it, and whether this database stores the far end data or is a pointer to such data or a hybrid of these models is beyond the scope of this Focus Group. Refer to the SIP NNI joint ATIS/SIP Forum taskforce for more information on industry databases to support efficient call setup and routing.

The first two options are network-specific and not subject to standardization.

GSMA PRD RCC.07 (RCS 5.1) specifies two mechanisms to achieve dynamic service discovery; one based on SIP Options (¶ 2.6.1.1) and one based on use of Presence (¶ 2.6.1.2). Either approach can be used either prior to initiation of a session request or after a session has been established.

Dynamic service discovery during the process of session establishment could be “just SIP”; with the caveat that only “positive inferences” can be drawn. For example one user might attempt to discover another user’s ability to support video, by sending a request for a video session. If that request is accepted, the first user can infer that the second user supports video. But if that request is rejected, the second user cannot know why.

The following diagram shows an example where capability discovery occur independent of session establishment.

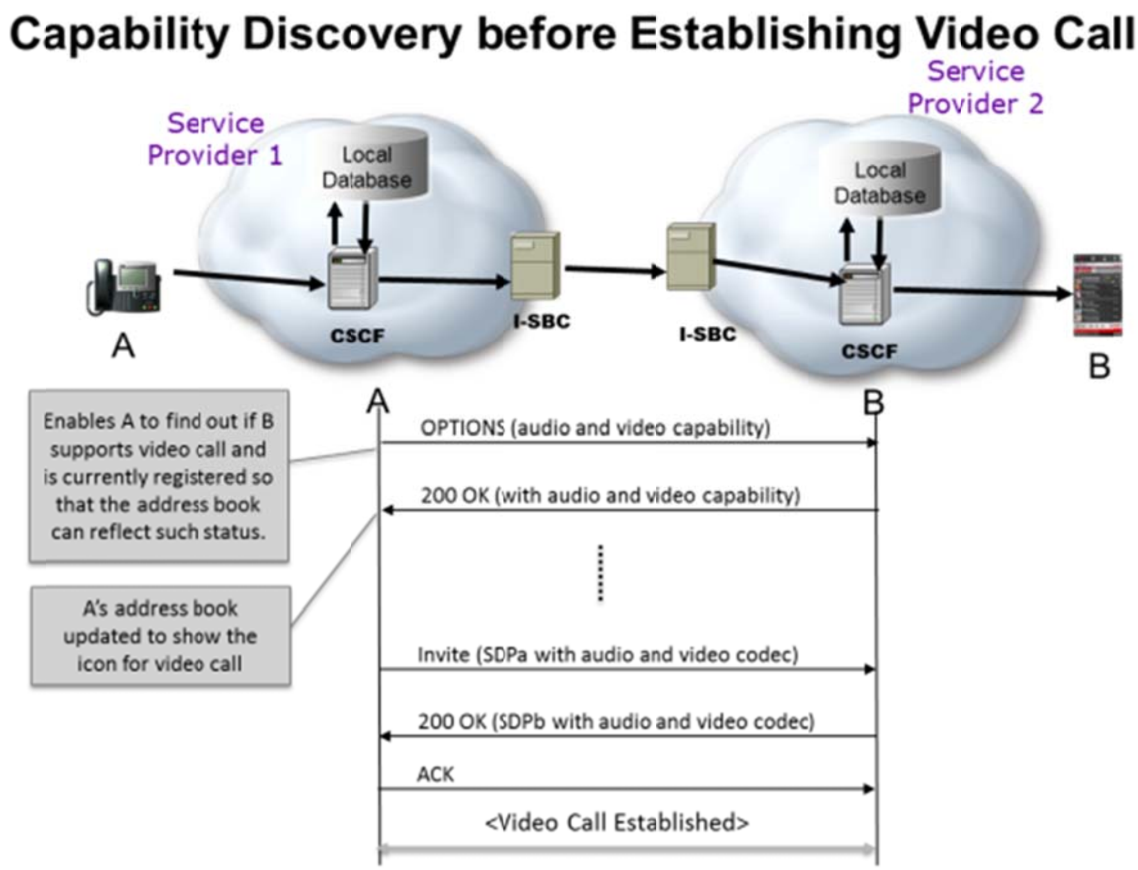

Figure 4. 6 - Capability Discovery before Establishing Video Call

NOTE: Support of using OPTIONS for capability discovery across NNI is *optional* as indicated in IETF RFC 3840. OPTIONS messaging can be used to discover far end video support status without trying to establish a video session. Trying to establish a video session implies the far end user may be alerted which may not be appropriate for capability discovery.

Although, permitted in a SIP dialog, the use of the SIP OPTIONS method to obtain capability information from the far-end is considered optional and based on bilateral agreement. If a UE uses SIP OPTIONS to learn the far-end capability and the UE at far-end doesn't implement this capability discovery mechanism, then the video call may not be set up.

4.2.2 Video Call

The following diagram shows an example video call in an IMS environment. This diagram is provided to illustrate the basic scenarios although it is recognized that in practice additional mechanisms may be required to ensure synchronization of audio and video.

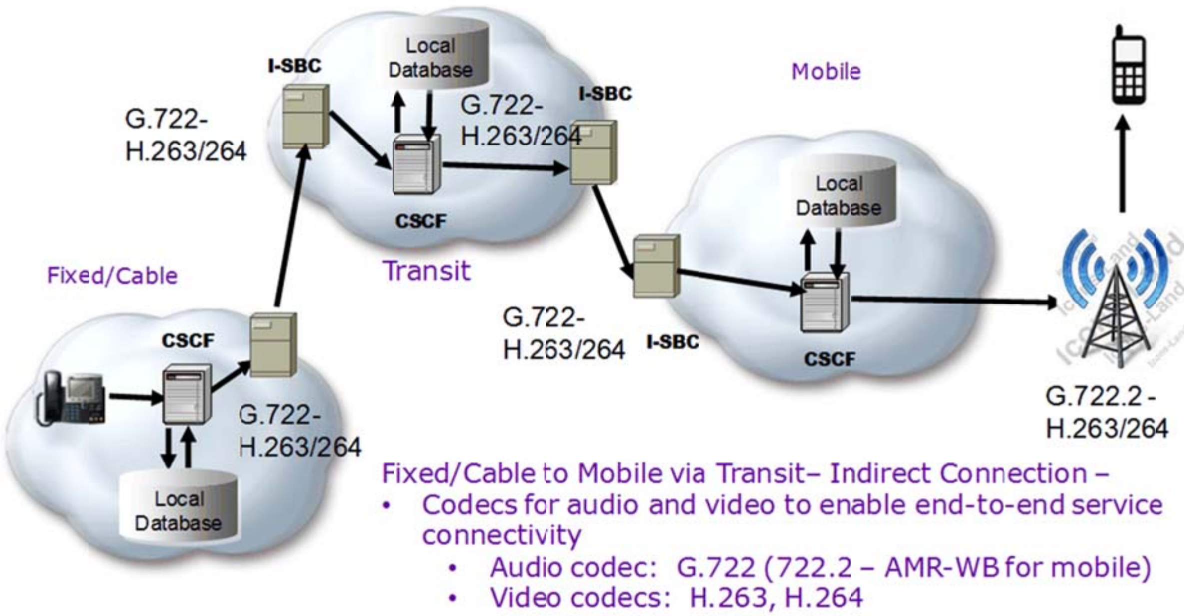

Figure 4. 7 - Video Call - Indirect Connection

4.2.3 Use Case - Upgrading Audio Call to Video

The following diagram illustrates a successful upgrade from audio to video call.

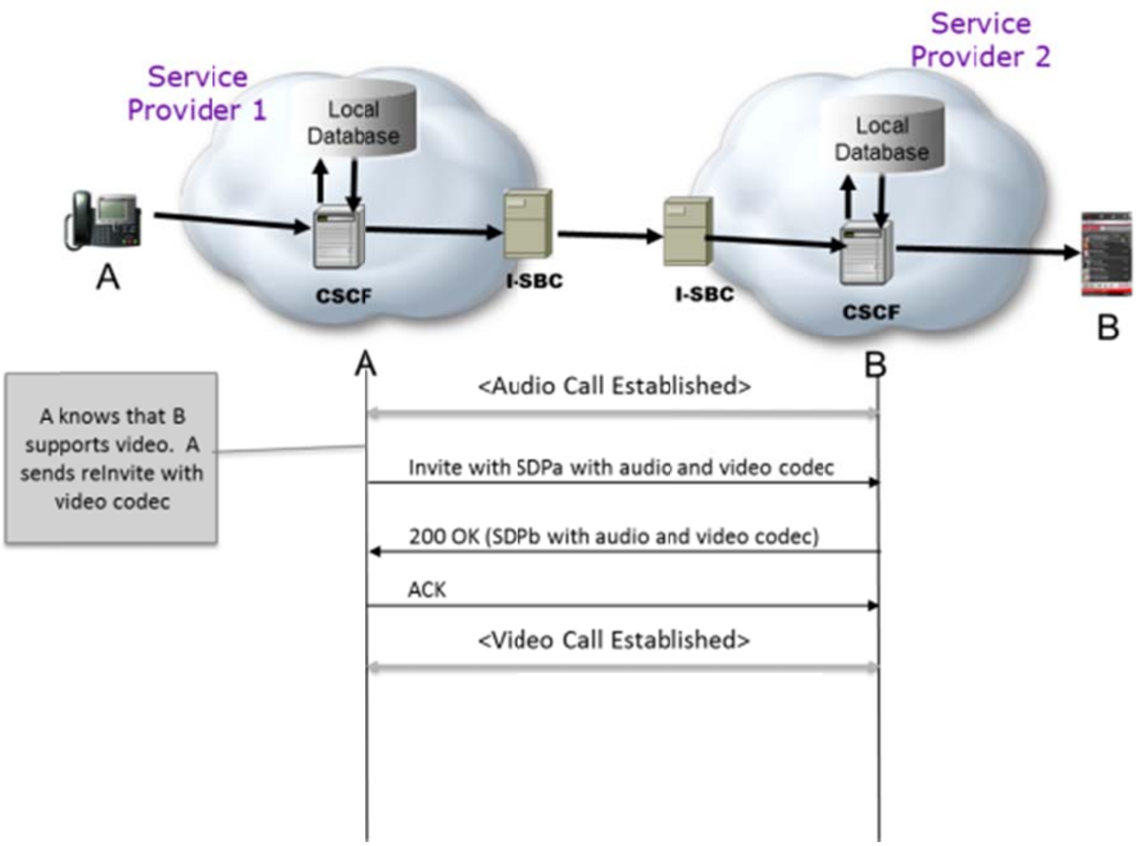

Figure 4. 8 - Upgrading Audio Call to Video

The following diagram shows how a rejection of video upgrade should work.

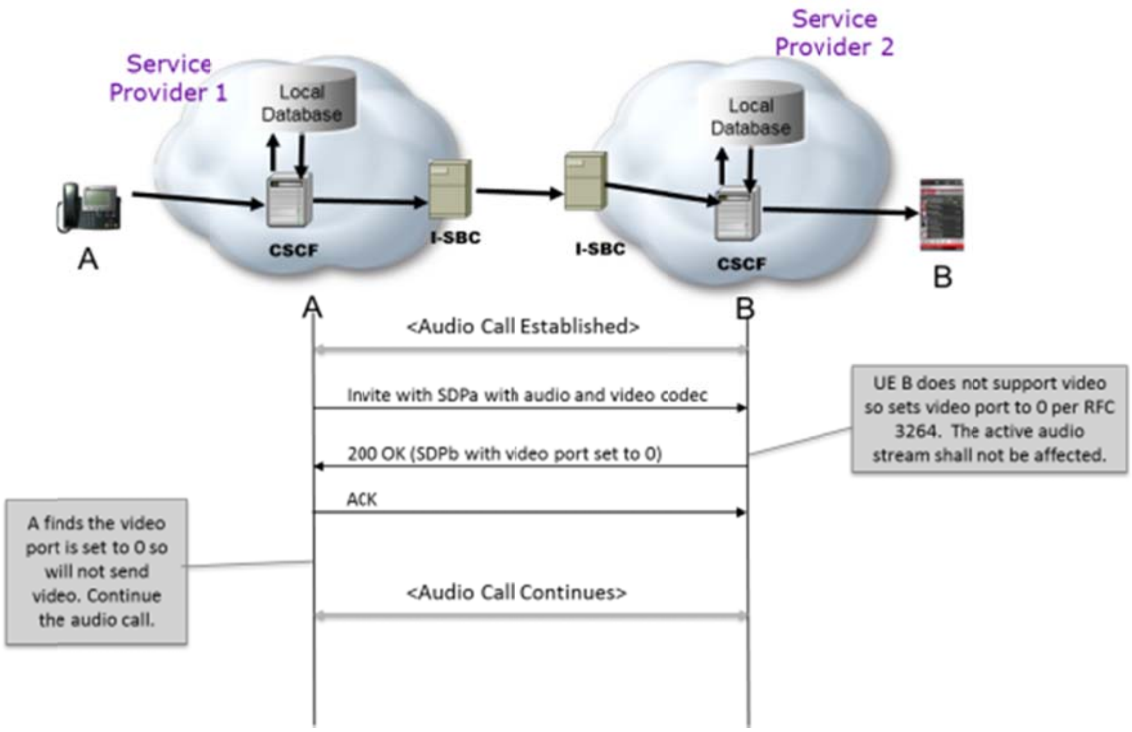

Figure 4. 9 - Rejected request to upgrade to video

The following diagram shows a variant of how reject can happen.

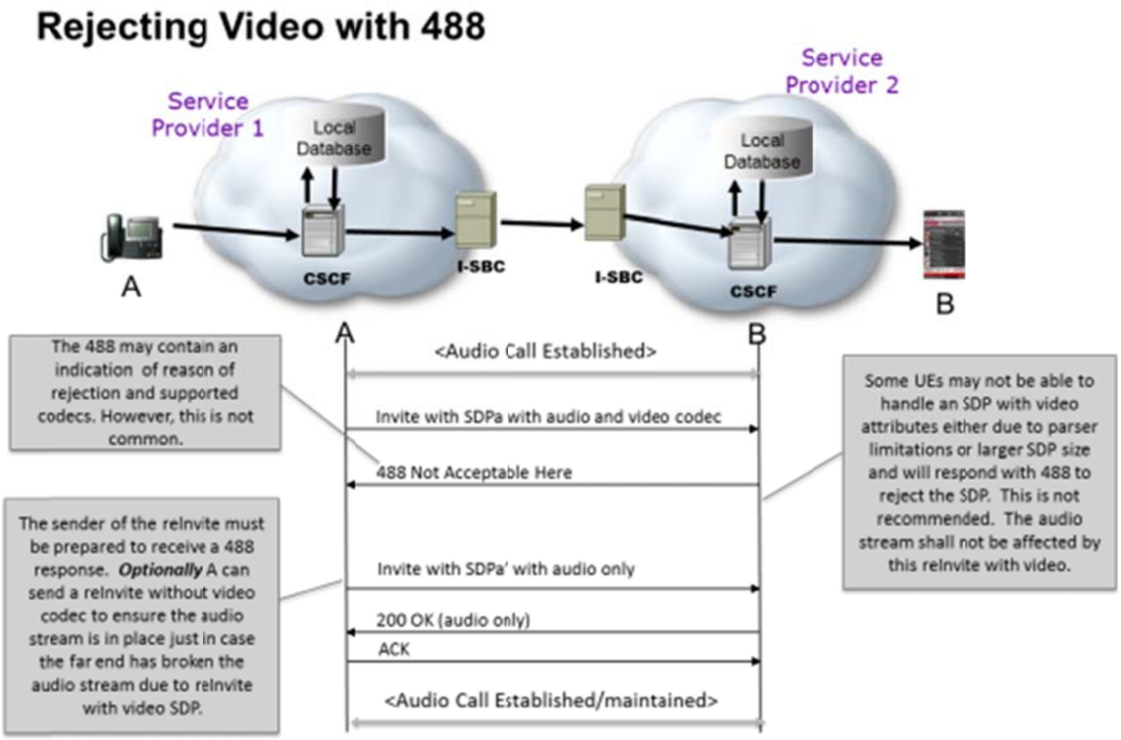

Figure 4. 10 - Rejecting Video with 488

NOTE: The B Party should accept whatever sub-set of the offer it can support. In this case, the "m=Audio" line in the SDP. The originating network is not obligated to make a different offer which is non-standard. Thus the above flow may be simplified where the equipment can accept the audio portion without re-initiating the invite.

4.2.4 Use Case – Video then back to Audio Call

The intent is to describe a possible consequence of the following two initial scenarios in an LTE network:

a) Two users starting a voice over IP call (i.e., audio only) and then promoting the session to video over IP call (i.e., upgrade voice + video call) adding an additional media stream to the ongoing session;

b) or engaging in a "direct video call" from the get go (i.e., both user's devices can support a video over IP call) and do not need to establish a voice call first to be upgraded to video later as in (a).

The receiving user's device in an ongoing video over IP call may request that its video connection be removed by sending a SIP re-INVITE request with an SDP offer where the port number of the video descriptor is set to zero. The video stream can be re-established by sending a new SIP re-INVITE request with an SDP offer where the port number of the video descriptor is set to a non-zero value. If the conference participant that established the video conference sends a SIP re-INVITE where the video is removed it is up to network policies if the video over IP call shall be degraded to a voice only conference or if the video conference can continue with the conference originator participating only using voice.
If the receiving user agrees to participate in the call as audio only, declining the video part of the communication, only a voice over LTE (or VoLTE) call is established or continued without affecting an ongoing voice over IP call.

As indicated in GSMA RCC.07 (RCS 5.1) a video call may be also declined by the receiving user, and no communication is established between the receiving user's device. The call may be redirected to a voice or video messaging system depending on the policies of the receiving user's network.
An active video over IP call can be terminated and "downgraded" to voice over IP call if either of the user's devices sends a SIP re-INVITE request to the AS managing the session indicating signaling that the video media component has been removed in the SDP offer the communication regardless of whether the call was initiated directly as a video over IP call or initially started as a voice over IP call only.

4.2.5 SDO Assessment/Gap Analysis

4.2.5.1 3GPP

4.2.5.1.1 Session Establishment & Signaling

In a similar manner as for HD Voice, conversational video or Video Calling relies on procedures defined in 3GPP TS 24.229 for the establishment of a session and required SIP based signaling. The media type "video" in SDP m-lines is supported within SIP INVITE dialogues at the inter-IMS NNI. For interoperability and forward-compatibility reasons, a device capable to support video calls will add a "video" media feature tag to the Contact header field of the REGISTER request, as described in IETF RFC 3840. According to 3GPP TS 26.114, if video is used in a session, bandwidth, RTP profile, video codec, profile, and level shall be determined during session setup. Signaling of the current orientation of the image captured on the sender side to the receiver for appropriate rendering and displaying (Coordination of Video Orientation) needs to be negotiated as well.

4.2.5.1.2 Addressing & Routing

Addressing for multimedia services such as video telephony is defined in 3GPP TS 23.228 and its format is defined in 3GPP TS 23.003. Support of MSISDN as a Public User Identity and mechanism to associate a Tel URI and an alphanumeric SIP URI is specified in TS 23.228 and TS 24.229. 3GPP TS 29.165 defines interworking by means of routing at the NNI.

4.2.5.1.3 Media Handling

3GPP and 3GPP2 specify mandatory and recommended codecs to allow cellular video telephony and streaming with QCIF resolution (176 x 144), consistent with a cellular handsets' small screen size and cellular networks' limited bandwidth_[TS 26.234] and_[TS 26.235]. Applicable codecs are defined in the list of codecs to be included in inter-operator agreements and to guarantee support at the NNI by the peer operators.

3GPP TS 26.114 specifies a client for the Multimedia Telephony Service for IMS (MTSI) supporting conversational speech, video, and text transported over RTP. As such, an MTSI client used for the delivery of video between end points will at least offer H.264 (AVC) Constrained Baseline Profile support at a level higher than Level 1.2.

4.2.5.2 GSMA

4.2.5.2.1 Session Establishment & Signaling

Conversational video services over LTE are specified in GSMA PRD IR.94. Video service clients compliant with GSMA PRD IR.94 and conform to procedures defined in GSMA PRD IR.92 (VoLTE), and for interoperability and compatibility reasons indicate the capability to handle video calls by adding a "video" media feature tag to the Contact header field of the REGISTER request, as described in IETF RFC 3840.

4.2.5.2.2 Addressing & Routing

Conversational video services conform to addressing guidelines specified for IMS based multiparty services defined in GSMA PRD IR.65, which refers to procedures defined in 3GPP TS 23.228 and addresses formats defined in 3GPP TS 23.003.

Conversational video clients also conform to routing guidelines specified in GSMA PRD IR.65 requirements for routing between IMS networks as specified in 3GPP TS 29.165, and DNS query handling specified in GSMA PRD IR.67 to ensure end-to-end service interoperability.

4.2.5.2.3 Media Handling

The conversational video services over LTE comprise calls with full duplex voice and simplex/full-duplex video media with tight synchronization between the constituent streams. The call can be a point to point call or a multiparty conference call. The conversational video service over LTE can also be used to interact with dial-in video conference services, such as High-Definition Video Conference (HDVC), and based on procedures specified in GSMA PRD IR.39 (refer to section 5.10 – Video Conference/Collaboration in this document).

- An IR.94 compliant client must indicate the capability to handle video calls (i.e., “video” media feature tag as described in IETF RFC 3840 and RFC 3841) and also include both voice and video media descriptors in the respective SDP offer as specified for Multimedia Telephony (MMTel) according to 3GPP TS 26.114.
- IR.94 clients and the network must follow the voice media requirements and requirements on DTMF events as defined in GSMA PRD IR.92.
- In addition, IR.94 compliant clients and the network must fulfill the requirements on supplementary services specified in GSMA PRD IR.92 and video media related additions, such as Coordination of Video Orientation (CVO) as specified in 3GPP TS 26.114. For CVO two (2) bits granularity are supported to terminate the user plane between UE and the entities in the IMS core network.
- IR.94 clients and the IMS core network terminating the user plane must support of ITU-T Recommendation H.264 Constrained Baseline Profile (CBP) Level 1.2 as specified in 3GPP TS 26.114. Support of ITU-T Recommendation H.263 Profile 0 Level 45 as specified in 3GPP TS 26.114 is not required in the client.
- Change of video resolution mid-stream by transmission of new parameter sets in the RTP media stream must be supported, as long as those parameter sets conform to the negotiated profile and level.

If due to certain circumstances video is removed, the conference shall be degraded to a voice only conference, or if the video conference can continue with the conference originator participating only using voice.

4.2.5.3 PacketCable

4.2.5.3.1 Session Establishment & Signaling

The PacketCable HD-voice session establishment and signaling gaps described in section **Error! Reference source not found.** generally apply to video as well. Video session establishment signaling for PacketCable 1.5 and PacketCable 2.0 is described in PKT-SP-CODEC1.5-I04-120412 and PKT-SP-CODEC-MEDIA-C01-140314 respectively.

4.2.5.3.2 Addressing & Routing

The PacketCable HD-voice session addressing and routing gaps described in section **Error! Reference source not found.**. generally apply to video as well.

4.2.5.3.3 Media Handling

PacketCable requires video codecs to be supported for various types of consumer devices, especially personal computers. 3GPP and 3GPP2 support well-known video codecs, but these codecs are limited to profiles for resolutions suitable for cellular devices. PacketCable defines additional codecs and profiles to support multiple resolutions as is necessary to provide user satisfaction with both small-screen and larger-screen devices.

Section 4.2.6 below provides more details on video codec specification for PacketCable.

4.2.6 PacketCable Video Codec Specification

The PacketCable architecture enables cable operators to provide a wide range of IP multimedia services. In addition to VoIP and data-centric services, video-over-IP represents another important application area for PacketCable, and opens up new revenue streams for cable operators. From the perspective of cable customers, PacketCable video services can significantly enrich their communication and entertainment experiences, thus strengthening their preference of using the cable network for broadband access and digital entertainment

PacketCable video services can potentially be delivered over different hardware platforms, which include the following:

- Standalone Video Telephony System: This is the traditional video telephony platform. Operating in the PacketCable environment, this platform can provide increased video quality due to the higher bandwidth and enhanced QoS afforded by the PacketCable network, with the video quality ultimately being limited by the capability of the integrated display device.
- Wireless CPE: With their mobility, wireless CPE devices (such as 3G handset, Wi-Fi IP phone, and Wi-Fi-enabled PDA) are emerging as an important video platform fulfilling the niche role for mobile interactive and streaming video applications. In the past, the video quality of a wireless CPE was severely limited by both the bandwidth of the wireless network and the capability of the integrated display device. However, with increasing bandwidth of the wireless network and improving display capability, new classes of wireless CPE devices supporting ever increasing video resolutions are quickly emerging.
- Set-Top Box (STB): As an asset owned by cable operators, the STB plays a critical role in providing customers with traditional digital TV programming and associated video services (e.g., VOD and PVR). With PacketCable, the STB's role can be expanded to offer complementary IP-based video telephony and entertainment, reinforcing the prominence of cable operators in providing innovative video services to the customers. These complementary services can greatly benefit from the natural video interface of the TV display - especially with its high-definition capability and wide screen.

In addition, by standardizing video codecs, PacketCable can support interoperability and feature interaction among different video platforms. For instance, a cable customer can use the STB to conduct video conferencing on TV with a remote cellular user. Also, the customer can instruct the STB to re-render the locally stored PVR video content and stream it to a remote cellular user, and vice versa.

As can be seen, an important characteristic of PacketCable video-over-IP applications is their diversity. In general, such diversity can be viewed from two perspectives or dimensions:

1. Video Resolution: Compatible with their video display devices and available network bandwidth, video applications target different video resolutions, such as QCIF, CIF, SD, and HD.
2. Video Stream Direction: Different video applications deal with three different video stream directions: send only (1-way encode), receive only (1-way decode), and simultaneously send and receive (2-way encode/decode).

Since 1990, various video codecs have been developed to cater for different applications, mainly through two international standards bodies-VCEG (Video Coding Experts Group) of ITU-T and MPEG (Moving Picture Experts Group) of ISO/IEC. H.261 was developed by ITU-T as the first video codec for video conferencing. MPEG-1 was introduced for video compact-disk storage, and was evolved into MPEG-2 (or H.262 as adopted by ITU-T) as the standard codec for DVD, digital TV and HDTV. Subsequent H-series and MPEG-series video codecs include H263 and MPEG-4 Part 2. More recently, a high-performance and general-purpose codec, H.264/AVC, has been developed jointly by ITU-T and ISO/IEC.

Associated with each of these video codecs are its profiles and levels. A profile defines a set of coding tools or algorithms that can be used in generating a compliant video bitstream, and a level places constraints on certain key parameters of the bitstream. Each profile-level combination of a codec represents a conformance point that facilitates the interoperability among different video devices. Collectively, profiles and levels define the theoretical capability and flexibility of the associated codec.

As a new-generation architecture for IP multimedia services over the cable network, PacketCable requires video codecs that are versatile and future-proof. At the same time, it needs to accommodate codecs that are used in legacy video systems. Furthermore, to support cable/cellular integration initiative, PacketCable video codec requirements need to be compatible with video codecs standardized by cellular standards bodies such as 3GPP and 3GPP2.

4.2.6.1 Supported Codecs

This section describes every video codec supported in PacketCable. Whether a codec is mandatory, recommended or optional depends on the application for which it is used. Therefore, the normative status of each codec is indicated in the associated application capability documents. However, if a particular codec is supported for an application, all the requirements for that codec as specified in this section MUST be met.

4.2.6.2 H.263

First approved in 1996, H.263[H.263] is a video codec standardized by ITU-T VCEG for low-bit-rate video telephony. It was designed initially for circuit-switched networks such as PSTN, and has since been applied for ISDN and packet networks. H.263 *MAY* be supported on User Equipment and Media Gateways.

H.263 incorporates improvements over H.261[H.261], the previous ITU-T standard for video telephony, in the areas of performance and error recovery. It has been designed to stream video at bandwidths as low as 20-24 Kbps. As a general rule, H.263 improves the coding efficiency over H.261 by 100% (i.e., requires half the bandwidth to achieve the same video quality). In addition, as shown in [H.263 Annex X] it supports a wider range of video resolutions than H.261 (which only supports QCIF and CIF). As a result, H.263 has essentially replaced H.261.

4.2.6.2.1 Profile/Level Requirements

If User Equipment and Media Gateways support H.263, the following requirements apply:

- H.263 Profile 0 @ Level 45 MUST be supported for QCIF applications.
- H.263 Profile 3 @ Level 45 *SHOULD* be supported for QCIF applications.

- H.263 Profile 0 @ Level 40 *MUST* be supported for CIF applications.
- H.263 Profile 3 @ Level 40 *SHOULD* be supported for CIF applications.

The H.263 support for SD and HD application types is not specified for PacketCable.

The above requirements are summarized in Table 4.1.

Table 4. 1 - PacketCable Requirements for H.263

Direction Resolution	One-Way Decode	One-Way Encode	Two-Way Codec (Interactive)
QCIF	H.263 Profile 0 @ Level 45 (MANDATORY) H.263 Profile 3 @ Level 45 (RECOMMENDED)	H.263 Profile 0 @ Level 45 (MANDATORY) H.263 Profile 3 @ Level 45 (RECOMMENDED)	H.263 Profile 0 @ Level 45 (MANDATORY) H.263 Profile 3 @ Level 45 (RECOMMENDED)
CIF	H.263 Profile 0 @ Level 40 (MANDATORY) H.263 Profile 3 @ Level 40 (RECOMMENDED)	H.263 Profile 0 @ Level 40 (MANDATORY) H.263 Profile 3 @ Level 4 (RECOMMENDED)	H.263 Profile 0 @ Level 40 (MANDATORY) H.263 Profile 3 @ Level 40 (RECOMMENDED)
SD	Not Specified	Not Specified	Not Specified
HD	Not Specified	Not Specified	Not Specified

With the above requirements, User Equipment and Media Gateways supporting QCIF are able to interoperate with mobile devices supporting 3GPP packet-switched conversational multimedia services [TS 26.235] and 3GPP packet-switched streaming services [TS 26.234], which all mandate H.263 Profile 0 @ Level 45 and recommend H.263 Profile 3 @ Level 45. The same requirements apply to 3GPP2.

4.2.6.2.2 Payload Header Formats

The RTP payload format for H.263-encoded video media MUST be compliant with [RFC 4629].

4.2.6.2.3 Session Description

The session description for H.263-encoded video media *MUST* be compliant with[RFC 4629].

In particular, the MIME media type video/H263-2000 string is mapped to fields in the Session Description Protocol [RFC 4566] as follows:

- The media name in the "m=" line of SDP *MUST* be video.
- The encoding name in the "a=rtpmap" line of SDP *MUST* be H263-2000 (the MIME subtype).
- The clock rate in the "a=rtpmap" line *MUST* be 90000.
- The optional parameters "profile" and "level". The "profile" parameter corresponds to the H.263 profile number, in the range 0 through 10, specifying the supported H.263 annexes/subparts. The "level" parameter corresponds to the level of bitstream operation, in the range 0 through 100, specifying the level of computational complexity of the decoding process. The specific values for the profile and level parameters and their meaning are defined in [H.263 Annex X]. Note that the

RTP payload format for H263- 2000 is the same as for H.263-1998, but additional annexes/subparts are specified along with the profiles and levels.

An example of media representation in SDP is as follows (Profile 0, Level 45):

```
m=video 49170 RTP/AVP 98
a=rtpmap:98 H263-2000/90000
a=fmtp:98 profile=0; level=45
b=TIAS:2048000
```

4.2.6.3 H.264/AVC

H.264[H.264] (or MPEG-4 Part 10), is a new generation of video codec jointly developed by the ITU-T VCEG and ISO/IEC MPEG. It was first approved in 2003. The codec is also referred to as AVC, for Advanced Video Coding. H.264 *MAY* be supported on User Equipment and Media Gateways.

H.264/AVC is designed to offer good video quality at bit rates that are substantially lower (e.g., half or less) in comparison with previous video codecs (e.g., MPEG-2, H.263, or MPEG-4 Part 2). In addition, H.264/AVC is also designed to be a general-purpose codec that can be applied to a wide variety of applications (e.g., low-high bit rates, and low-high video resolutions) and can work robustly on a wide variety of networks and systems (e.g., narrowband and wideband, wireline and wireless, broadcast, streaming, DVD storage, and video telephony). As reported in one benchmarking [H.264], H.264/AVC (with Main Profile) offers coding-efficiency improvements over MPEG-2 (with Main Profile), H.263 (with High Latency Profile) and MPEG-4 (with Advanced Simple Profile) by about 64%, 48%, and 38%, respectively. The much improved coding efficiency results from numerous enhancements, including intra-picture prediction, a new 4x4 integer transform, multiple reference frames, variable block sizes, 1/4-pixel precision for motion compensation, a deblocking filter, and enhanced entropy coding.

In general, H.264/AVC has a much higher complexity than the previous video codecs, and requires substantially higher signal-processing capability for encoder and decoder. This is especially the case if full coding efficiency of the codec needs to be realized.

4.2.6.3.1 Profile/Level Requirements

If User Equipment and Media Gateways support H.263, the following requirements apply:

- H.264 Baseline Profile @ Level 1b *MUST* be supported for QCIF applications.
- H.264 Baseline Profile @ Level 1.2 *MUST* be supported for CIF 1-way applications.
- H.264 Baseline Profile @ Level 1.2 *MUST* be supported for CIF 2-way applications.
- H.264 Baseline Profile @ Level 1.3 *SHOULD* be supported for CIF 1-way applications.
- H.264 Baseline Profile @ Level 1.3 *SHOULD* be supported for CIF 2-way applications.
- H.264 Main Profile @ Level 3 *MUST* be supported for SD 1-way applications.
- H.264 Baseline Profile @ Level 3 *MUST* be supported for SD 2-way applications.
- H.264 High Profile @ Level 4 *MUST* be supported for HD 1-way decode applications.

The H.264 support for HD 1-way encode and HD 2-way encode/decode is not specified for PacketCable.

When operating in conformance with the Baseline Profile, an encoder MUST be able to generate a bitstream conformant with constraint_set1_flag=1, such that the bitstream can be decoded by a Main Profile decoder. However, if the communicating User Equipment or Media Gateways negotiate the use of any of the Baseline Profile tools that are not in Main Profile, e.g., FMO, ASO, or redundant slices, the encoders *MAY* operate with constraint_set1_flag=0.

These requirements are summarized in Table 5.3 below.

Table 5. 1 - PacketCable Requirements for H.264/AVC

Direction Resolution	One-Way Decode	One-Way Encode	Two-Way Codec (Interactive)
QCIF	H.264 Baseline Profile @ Level 1b, with constraint_set1_flag = 1 (MANDATORY)	H.264 Baseline Profile @ Level 1b, with constraint_set1_flag = 1 (MANDATORY)	H.264 Baseline Profile @ Level 1b, with constraint_set1_flag = 1 (MANDATORY)
CIF	H.264 Baseline Profile @ Level 1.2, with constraint_set1_flag = 1 (MANDATORY) H.264 Baseline Profile @ Level 1.3, with constraint_set1_flag = 1 (RECOMMENDED)	H.264 Baseline Profile @ Level 1.2, with constraint_set1_flag = 1 (MANDATORY) H.264 Baseline Profile @ Level 1.3, with constraint_set1_flag = 1 (RECOMMENDED)	H.264 Baseline Profile @ Level 1.2, with constraint_set1_flag = 1 (MANDATORY) H.264 Baseline Profile @ Level 1.3, with constraint_set1_flag = 1 (RECOMMENDED)
SD	H.264 Main Profile @ Level 3 (MANDATORY)	H.264 Main Profile @ Level 3 (MANDATORY)	H.264 Baseline Profile @ Level 3, with constraint_set1_flag = 1 (MANDATORY)
HD	H.264 High Profile @ Level 4 (MANDATORY)	Not Specified	Not Specified

With these requirements, PacketCable video devices are able to interoperate with 3GPP mobile devices supporting H.264/AVC Baseline Profile Level 1b, which is recommended for 3GPP packet-switched conversational multimedia services [TS 26.235] and H.264/AVC Baseline Profile Level 1.2, which is recommended for 3GPP packet-switched streaming services [TS 26.234]. The same requirements apply to 3GPP2.

4.2.6.3.2 Payload Header Formats

The RTP payload format for H.264/AVC-encoded video media *MUST* be compliant with [RFC 3984] and [RFC 3555].

4.2.6.3.3 Session Description

The session description for H.264/AVC-encoded video media *MUST* be compliant with [RFC 3984] and [RFC 3555].

In particular, the MIME media type video/H264 string is mapped to fields in the Session Description Protocol (SDP)[RFC 4566] as follows:

- The media name in the "m=" line of SDP MUST be video.
- The encoding name in the "a=rtpmap" line of SDP MUST be H264 (the MIME subtype).
- The clock rate in the "a=rtpmap" line MUST be 90000.
- The optional parameters "profile-level-id", "max-mbps", "max-fs", "max-cpb", "max-dpb", "max-br", "redundant-pic-cap", "sprop-parameter-sets", "parameter-add", "packetization-mode", "sprop-interleaving-depth", "deint-buf-cap", "sprop-deint-buf-req", "sprop-init-buf-time", "sprop-max-don-diff", and "max-rcmd-nalu-size", when present, MUST be included in the "a=fmtp" line of SDP. These parameters are expressed as a MIME media type string, in the form of a semicolon separated list of parameter=value pairs.

An example of media representation in SDP is as follows (Baseline Profile, Level 3.0, some of the constraints of the Main profile may not be obeyed):

```
m=video 49170 RTP/AVP 98

a=rtpmap:98 H264/90000

a=fmtp:98 profile-level-id=42A01E; sprop-parameter-sets=Z0IACpZTBYmI,aMljiA==

b=TIAS:10000000
```

4.2.6.4 MPEG-2

MPEG-2[ISO 13818-2] is a video codec standardized by ISO/IEC MPEG and was first approved in 1995. It has been widely used for traditional cable and satellite digital broadcast TV programming and DVD storage. A large repository of video content is available in MPEG-2-encoded format. MPEG-2 *MAY* be supported on User Equipment and Media Gateways.

4.2.6.4.1 Profile/Level Requirements

If User Equipment or Media Gateways support MPEG-2, then the following requirements apply:

- MPEG-2 Main Profile @ Main Level *SHOULD* be supported for SD 1-way encode and decode applications.

The MPEG-2 support for other application types is not specified for PacketCable.

If User Equipment or Media Gateways support MPEG-2, then the following requirements apply:

- MPEG-2 Main Profile @ Main Level *SHOULD* be supported for SD 1-way encode and decode applications.

The MPEG-2 support for other application types is not specified for PacketCable.

The above requirements are summarized in Table 4.2.

Table 4. 2 - PacketCable Requirements for MPEG-2

Direction Resolution	One-Way Decode	One-Way Encode	Two-Way Codec (Interactive)
QCIF	Not Specified	Not Specified	Not Specified
CIF	Not Specified	Not Specified	Not Specified
SD	MPEG-2 Main Profile @ Main Level (RECOMMENDED)	MPEG-2 Main Profile @ Main Level (RECOMMENDED)	Not Specified
HD	Not Specified	Not Specified	Not Specified

4.2.6.4.2 Payload Header Formats

The RTP payload format for MPEG-2-encoded video media MUST be compliant with [RFC 2250] and [RFC 3555].

4.2.6.4.3 Session Description

The session description for MPEG-2-encoded video media MUST be compliant with [RFC 2250] and [RFC 3555].

In particular, for MPEG-2 Transport Stream, the MIME media type video/MP2T string is mapped to fields in the Session Description Protocol (SDP)[RFC 4566] as follows:

- The media name in the "m=" line of SDP MUST be video.
- The encoding name in the "a=rtpmap" line of SDP MUST be MP2T (the MIME subtype).
- The clock rate in the "a=rtpmap" line MUST be 90000.

For MPEG-2 Elementary Stream, the MIME media type video/MPV string is mapped to fields in the Session Description Protocol (SDP)[RFC 4566] as follows:

- The media name in the "m=" line of SDP MUST be video.
- The encoding name in the "a=rtpmap" line of SDP MUST be MPV (the MIME subtype).
- The clock rate in the "a=rtpmap" line MUST be 90000.
- The optional parameter "type" MUST be included in the "a=fmtp" line to indicate either "mpeg2-halfd1" (half-D1 video resolution) or "mpeg2-fulld1" (full-D1 video resolution).

An example of MPEG-2 Transport Stream media representation in SDP is as follows:

```
m=video 49170 RTP/AVP 98
a=rtpmap:98 MP2T/90000
```

An example of MPEG-2 Elementary Stream media representation in SDP is as follows:

```
m=video 49170 RTP/AVP 98
a=rtpmap:98 MPV/90000
a=fmtp:98 type=mpeg2-fulld1
```

4.2.6.5 MPEG-4 Part 2

MPEG-4 Part 2[ISO 14496-2] is a video codec that belongs to the MPEG-4 standard family and was first approved in 1999. This codec has been employed by some portable devices such as 3G handsets and digital still cameras that capture and playback video clips. As benchmarked in [ISO 14496-2] MPEG-4 Part 2 (with Advanced Simple Profile) offers a coding-efficiency improvement over MPEG-2 (with Main Profile) and H.263 (with High Latency Profile) of approximately 42% and 16%, respectively. MPEG-4 Part 2 MAY be supported on User Equipment and Media Gateways.

4.2.6.5.1 Profile/Level Requirements

If User Equipment or Media Gateways support MPEG-4 Part 2, then the following requirements apply:

- MPEG-4 Part 2 Simple Profile @ Level 0b SHOULD be supported for QCIF applications.
- MPEG-4 Part 2 Simple Profile @ Level 3 SHOULD be supported for CIF 1-way applications.
- MPEG-4 Part 2 Simple Profile @ Level 2 SHOULD be supported for CIF 2-way applications.

The MPEG-2 support for other application types is not specified for PacketCable.

The above requirements are summarized in Table 4.3.

Table 4. 3 - PacketCable Requirements for MPEG-4 Part 2

Direction Resolution	One-Way Decode	One-Way Encode	Two-Way Codec (Interactive)
QCIF	MPEG-4 Part 2 Simple Profile @ Level 0b (RECOMMENDED)	MPEG-4 Part 2 Simple Profile @ Level 0b (RECOMMENDED)	MPEG-4 Part 2 Simple Profile @ Level 0b (RECOMMENDED)
CIF	MPEG-4 Part 2 Simple Profile @ Level 3 (RECOMMENDED)	MPEG-4 Part 2 Simple Profile @ Level 3 (RECOMMENDED)	MPEG-4 Part 2 Simple Profile @ Level 2 (RECOMMENDED)
SD	Not Specified	Not Specified	Not Specified
HD	Not Specified	Not Specified	Not Specified

With the above recommendation, PacketCable video devices are able to interoperate with 3GPP mobile devices supporting MPEG-4 Part 2 Simple Profile Level 0b, which is recommended for 3GPP packet-switched conversational multimedia services [TS 26.235] and MPEG-4 Part 2 Simple Profile Level 3, which is recommended for 3GPP packet-switched streaming services [TS 26.234]. The same requirements apply to 3GPP2.

4.2.6.5.2 Payload Header Formats

The RTP payload format for MPEG-4 Part 2-encoded video media MUST be compliant with [RFC 3016] and [RFC 3555].

4.2.6.5.3 Session Description

The session description for MPEG-4 Part 2-encoded video media MUST be compliant with [RFC 3016] and [RFC 3555].

In particular, the MIME media type video/MP4V-ES string is mapped to fields in the Session Description Protocol (SDP)[RFC 4566] as follows:

- The media name in the "m=" line of SDP MUST be video.
- The encoding name in the "a=rtpmap" line of SDP MUST be MP4V-ES (the MIME subtype).
- The optional parameter "rate" goes in "a=rtpmap" as the clock rate.
- The optional parameter "profile-level-id" and "config" go in the "a=fmtp" line to indicate the coder capability and configuration, respectively. These parameters are expressed as a MIME media type string, in the form of as a semicolon separated list of parameter=value pairs.

The following is an example of media representation in SDP, with Simple Profile/Level 2, rate=90000 (90kHz), "profile-level-id" and "config" present in "a=fmtp" line:

```
m=video 49170/2 RTP/AVP 98
a=rtpmap:98 MP4V-ES/90000
a=fmtp:98 profile-level-
id=2;config=000001B001000001B50900000100000001200008440FA282C2090A21F
```

4.2.6.5.4 Summary of Supported Codecs

Table 4.4 summarizes all video codecs that are supported by PacketCable, with profile and level requirements specified with respect to various application types. The choice of which resolution/direction combinations to support is vendor-specific, but if User Equipment and Media Gateways support a particular resolution/direction combination, the requirements specified for that combination *MUST* be met.

Table 4. 4 - Summary of PacketCable Video Codec Requirements

Direction Resolution	One-Way Decode	One-Way Encode	Two-Way Codec (Interactive)
QCIF	H.263 Profile 0 @ Level 45 (MANDATORY) H.264 Baseline Profile @ Level 1b, with constraint_set1_flag = 1 (MANDATORY) H.263 Profile 3 @ Level 45 (RECOMMENDED) MPEG-4 Part 2 Simple Profile @ Level 0b (RECOMMENDED)	H.263 Profile 0 @ Level 45 (MANDATORY) H.264 Baseline Profile @ Level 1b, with constraint_set1_flag = 1 (MANDATORY) H.263 Profile 3 @ Level 45 (RECOMMENDED) MPEG-4 Part 2 Simple Profile @ Level 0b (RECOMMENDED)	H.263 Profile 0 @ Level 45 (MANDATORY) H.264 Baseline Profile @ Level 1b, with constraint_set1_flag = 1 (MANDATORY) H.263 Profile 3 @ Level 45 (RECOMMENDED) MPEG-4 Part 2 Simple Profile @ Level 0b (RECOMMENDED)
CIF	H.263 Profile 0 @ Level 40 (MANDATORY) H.264 Baseline Profile @ Level 1.2, with constraint_set1_flag = 1 (MANDATORY) H.263 Profile 3 @ Level 40 (RECOMMENDED) H.264 Baseline Profile @ Level 1.3, with constraint_set1_flag = 1 (RECOMMENDED) MPEG-4 Part 2 Simple Profile @ Level 3 (RECOMMENDED)	H.263 Profile 0 @ Level 40 (MANDATORY) H.264 Baseline Profile @ Level 1.2, with constraint_set1_flag = 1 (MANDATORY) H.263 Profile 3 @ Level 40 (RECOMMENDED) H.264 Baseline Profile @ Level 1.3, with constraint_set1_flag = 1 (RECOMMENDED) MPEG-4 Part 2 Simple Profile @ Level 3 (RECOMMENDED)	H.263 Profile 0 @ Level 40 (MANDATORY) H.264 Baseline Profile @ Level 1.2, with constraint_set1_flag = 1 (MANDATORY) H.263 Profile 3 @ Level 40 (RECOMMENDED) H.264 Baseline Profile @ Level 1.3, with constraint_set1_flag = 1 (RECOMMENDED) MPEG-4 Part 2 Simple Profile @ Level 2 (RECOMMENDED)
SD	H.264 Main Profile @ Level 3 (MANDATORY) MPEG-2 Main Profile @ Main Level (RECOMMENDED)	H.264 Main Profile @ Level 3 (MANDATORY) MPEG-2 Main Profile @ Main Level (RECOMMENDED)	H.264 Baseline Profile @ Level 3, with constraint_set1_flag = 1 (MANDATORY)
HD	H.264 High Profile @ Level 4 (MANDATORY)	Not Specified	Not Specified

4.2.7 Error Recovery

Communication errors can degrade video quality. There are many sources of such errors, including burst bit errors resulting from communication channel impairments and the loss of packets resulting from undesirable network conditions such as congestion.

There exist multiple mechanisms to mitigate the effect of communication errors on video quality:

- Forward error correction (FEC);
- Packet Retransmission;
- Error concealment; and
- Error-resilient coding.

The first two types of error-control mechanisms are application-specific, and are outside of the scope of this specification.

Different video codecs usually have their own algorithms for handling error concealment and error-resilience. These algorithms may also be specific to a particular profile/constraint of a video codec. This specification does not specify any error-concealment and error-resilience algorithms which are not included in the mandated or recommended video codecs.

4.2.8 Codec Naming & FlowSpec Parameters for Video Codecs

The video codecs defined in this specification MUST be encoded with the string names in the rtpmap parameters as shown in Table 4.5.

Table 4. 5 - Video Codecs rtpmap Parameters

Codec	Literal Codec Name	rtpmap Parameter
H.263	H263-2000	H263-2000/90000
H.264/AVC	H264	H264/90000
MPEG-2 Transport Stream	MP2T	MP2T/90000
MPEG2 Elementary Stream	MPV	MPV/90000
MPEG-4 Part 2	MP4V-ES	MP4V-ES/90000

Unknown rtpmap parameters *SHOULD* be ignored if they are received.

For every recommended codec (whether it is represented in SDP as a static or dynamic payload type), Table 4.6 describes the mapping that *MAY* be used from either the payload type or ASCII string representation to the bandwidth requirements for that codec, with the bandwidth requirements being expressed as Flowspec.

Table 4. 6 - Mapping of Video Codec Session Description Parameters to Flowspec

Parameters from Session Description			Flowspec parameters		Comments
RTP/AVP code	**Rtpmap**	**ptime (msec) (Note 1)**	**Values b,m,M (Note 2)**	**Values r,p (Note 3)**	
H.263	H263 2000/90000	N/A	b, M = 1500 bytes m = 128 bytes	Level 45: 20K bytes/sec Level 40: 308K bytes/sec	b, M and m are default values. r, p = Max Compressed Bit Rate x 120%
H.264	H264/90000	N/A	b, M = 1500 bytes m = 128 bytes	Level 1b: 20K bytes/sec Level 1.2: 58K bytes/sec Level 1.3: 116K bytes/sec Level 3: 1.5M bytes/sec Level 4: 3M bytes/sec	b, M and m are default values. r, p = Max Compressed Bit Rate x 120%
MPEG-2	Transport Stream: MP2T/90000 Elementary Stream: MPV/90000	N/A	b, M = 1500 bytes m = 128 bytes	Main Level: 2.25M bytes/sec	b, M and m are default values. r, p = Max Compressed Bit Rate x 120%
MPEG-4 Part 2	MP4V-ES/90000	N/A	b, M = 1500 bytes m = 128 bytes	Level 0b: 19.2K bytes/sec Level 2: 19.2K bytes/sec Level 3: 57.6K bytes/sec	b, M and m are default values. r, p = Max Compressed Bit Rate x 120%

NOTES:

(1) Ptime is not applicable to video in general. An application can set this parameter to an accurate value if it is known.

(2) The parameters b, M and m are set to their default values, since the packet sizes and maximum burst rates are not regular for video. An application can set these parameters to their accurate values if such values are known.

(3) Since the packet rates for video are not regular, the IP/UDP/RTP overhead data rate cannot be derived accurately in general. As a work-around, 20% of the maximum compressed data rate is assumed for the overhead. An application can set these parameters to their accurate values if such values are known.

For non-well-known codecs, the bandwidth requirements cannot be determined by the media name and transport address (m) and the media attribute (a) lines alone. In this situation, the SDP must use the bandwidth parameter (b) line to specify its bandwidth requirements for the unknown codec. The bandwidth parameter line (b) is of the form:

```
b= <modifier> : <bandwidth-value>
```

For example:

b= AS:99

The bandwidth parameter (b) will include the necessary bandwidth overhead for the IP/UDP/RTP headers. In the specific case where multiple codecs are specified, the bandwidth parameter should contain the least-upper-bound (LUB) of the desired codec bandwidths.

b is bucket depth (bytes). *m* is minimum policed unit (bytes). *M* is maximum datagram size (bytes).

r is bucket rate (bytes/sec). *p* is peak rate (bytes/sec).

4.2.9 Key Recommendations to be Addressed by IP Services NNI

- RFC 3264 support is required across NNI.
- When an offer containing both audio and video is received and the endpoint supports audio only, then the port for video must be set to 0 and the SDP should not be rejected. The audio stream should not be affected by the attempt to upgrade to video.
- SDP Size – must support SDP size of up to N (e.g., 1200) bytes where N is agreed to by both sides of NNI.
- Capability discovery via OPTIONS may be supported.
- R-URI of the OPTIONS can be a user agent, in which case, we are only querying the end user capabilities, not the nodes along the path which may not be able to support video.
- OPTIONS can be used as a heart-beat mechanism, but OPTIONS should not be sent unnecessarily if there are messages flowing back and force. This should be covered in other forums.
- Invite as a capability discovery mechanism may not work in all cases.
- It is also possible to discover video capabilities via Presence.

4.2.9.1 Video Codec Support

The following are the candidate codecs for support.

- H.263 profile 0 level 45 [10][11] should be supported for compatibility with earlier content and UEs.
- H.264 Level 1.2 {Default VoLTE VT}.
- H.264 (AVC) [52] Constrained Baseline Profile (CBP) Level 1.3 shall be supported.
- H.264 (AVC) [52] High Profile Level 3.1 with frame_mbs_only_flag=1 should be supported by MMS clients supporting HDTV video content at a resolution of 1280x720 (720p) with progressive scan at 30 frames per second. Maximum VCL Bit Rate shall be constrained to 14Mbps by cpbBrVclFactor & cpbBrNalFactor being fixed to 1000 and 1200 respectively, irrespective of the profile. Note that peak Bit Rate is determined by the CPB size.
- H.265 (HEVC) [62] Main Profile, Main Tier, Level 3.1 decoder should be supported. H.265 (HEVC) Main Profile shall be used with general_progressive_source_flag equal to 1, general_interlaced_source_flag equal to 0, general_non_packed_constraint_flag equal to 1, and general_frame_only_constraint_flag equal to 1.
- VP8 is royalty free highly efficient compression technology by Google. VP8 is currently supported in Nexus 5.

4.3 StandaloneMessaging

4.3.1 Use Case

The following diagram illustrates a successful Standalone messaging session using pager mode.

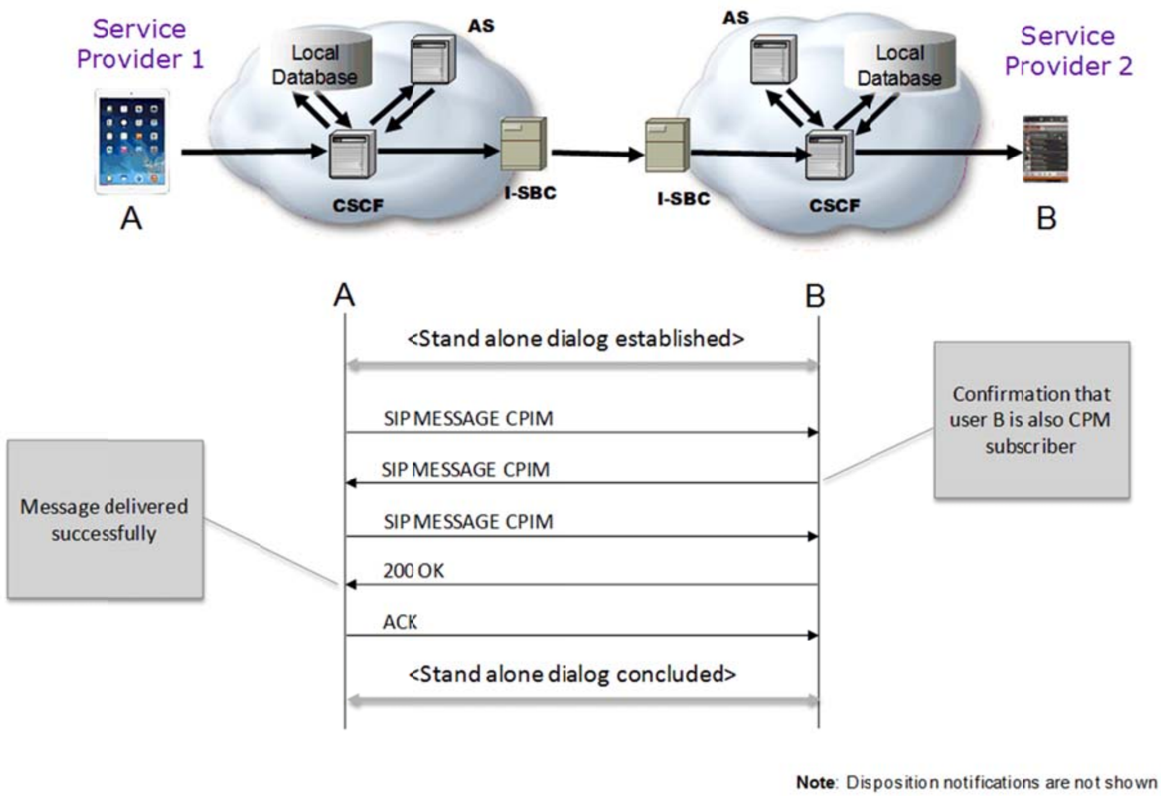

Figure 4. 11 - Standalone Message Session applying Pager Mode

Besides Pager Mode for small size messages, the Standalone messaging services offers the Large Message Mode messaging mechanism for messages exceeding 1300 bytes. Hence, in Standalone messaging the mode is selected based on the message size. Additional scenarios including interworking with legacy messaging services (SMS, MMS) are described in GSMA PRD RCC.07.

4.3.2 SDO Assessment/Gap Analysis

4.3.2.1 GSMA

4.3.2.1.1 Session Establishment & Signaling

According to GSMA PRD RCC.07, Standalone messaging takes advantage of the OMA CPM pager mode and large message mode mechanisms in order to consolidate text and multimedia messaging mechanisms into a single and unified messaging solution. Depending on the size, messages are communicated in a transparent fashion for the user, without requiring the user to select a particular messaging mode. Hence, there is no distinction between text and multimedia messages for the user. In addition, Standalone messaging as defined for GSMA RCS also supports interworking to SMS and MMS legacy messaging services involving mechanisms implemented in OMA CPM.

3GPP TS 24.229 specifies that the identification of services for charging and accounting purposes is expected to be based upon the contents of the P-Asserted-Service header and the actual media related contents of the SIP request and not the Accept-Contact header field contents or the contact reached.

As part of the session establishment, a messaging client will populate the P-Preferred-Service header field in all CPM requests with the CPM Feature tag defined for the service. The S-CSCF or AS that performs the service assertion in the originating network shall add the P-Asserted-Service header field set to the value of the asserted CPM service ICSI and remove the P-Preferred-Service header field before further routing the request.

In case of Pager Mode messages using the SIP MESSAGE method the following header fields will be populate as follows:

- P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.oma.cpm.msg.
- Accept-Contact header field containing feature tag +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.msg" and not carrying content of the type "message/imdn+xml".

Large Message mode messages are transported using MSRP and the corresponding is established using SIP INVITE with the following header fields populated as follows:

- P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.oma.cpm.largemsg.
- Accept-Contact and Contact header fields containing feature tag +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.largemsg".

The disposition notifications are carried in SIP MESSAGE requests with the feature tag included, and mapped by the terminating Messaging Server according to the messaging technology used before relaying to the end-point devices. If the receiving device does not support delivery of notifications, the terminating network Messaging Server shall generate the delivery notification on behalf of the device when the delivery notification is requested.

4.3.2.1.2 Addressing & Routing

Messaging services such as Standalone messaging conform to addressing guidelines specified for IMS based multiparty services defined in GSMA PRD IR.65, which refers to procedures is defined in 3GPP TS 23.228 and address formats defined in 3GPP TS 23.003.

Standalone messaging clients also conform to routing guidelines specified in GSMA PRD IR.65 requirements for routing between IMS networks as specified in 3GPP TS 29.165, and DNS query handling specified in GSMA PRD IR.67 to ensure end-to-end service interoperability.

Inter-Service Provider aspects for Messaging services are specified in GSMA PRD IR.90 and are applicable to both an individual implementation or as part of the RCS framework.

4.3.2.1.3 Media Handling

Messages delivered in pager mode use the SIP MESSAGE method and are limited to a maximum size of 1300 bytes. Messages exceeding this limit are delivered in large message mode. In case of interworking with devices supporting legacy messaging, standalone messages in pager mode are delivered to device via SMS as defined in 3GPP TS 23.040 or the SMS over IP as defined in GSMA PRD IR.92.

If large message mode is used, then messages are carried out in SIP/MSRP sessions and are only subject to size limitations specified by Service Providers. Multimedia messages in large message mode are delivered to a device supporting legacy messaging via MMS as defined in 3GPP TS 22.140 and 3GPP TS 23.140.

Since MSRP is used as a media transport messages in Large Message Mode, the MSRP message can be sent in chunks. However, the MSRP chunk size is not negotiable at the protocol level, according to GSMA PRD IR.90; the global maximum MSRP chunk size is set to 500 Kbytes in order to reduce the risk of MSRP message rejection or MSRP session disconnection by the MSRP receiver side. Consequently, the sender shall send chunks no greater than 500 Kbytes and receiver shall be able to handle chunks up to at least 500 Kbytes. For proper interworking Service Providers need to support the MSRP chunk size up to 500 Kbytes and specified in interworking agreements.

4.4 One-to-One Chat

4.4.1 Use Case

The following diagram illustrates a successful One-to-one chat session based applying the OMA CPM enabler.

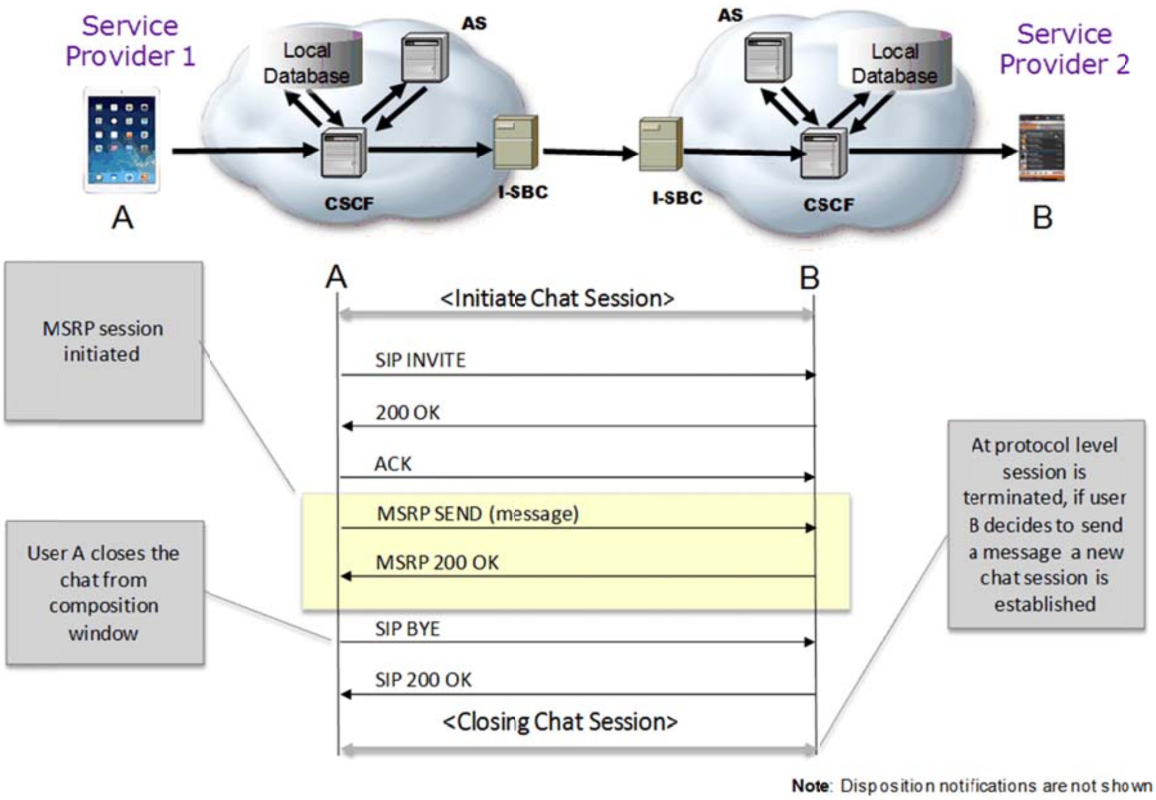

Figure 4. 12 - One-to-one Chat Session (based on OMA CPM)

One-to-one chat services can be realized using OMA SIMPLE IM or OMA CPM messaging enablers. Additional scenarios are described in GSMA PRD RCC.07.

4.4.2 SDO Assessment/Gap Analysis

4.4.2.1 GSMA

4.4.2.1.1 Session Establishment & Signaling

Since One-to-One messaging is based on OMA SIMPLE IM or OMA CPM, the IMS Core shall be configured to trigger the Messaging Server based on feature tags specified for either messaging technology. Regardless of the technology used chat sessions use MSRP for the transport of messages and chat sessions are established using the SIP INVITE method.

According to GSMA PRD IR.90 Interworking agreements need to define whether the Chat NNI is as per OMA IM with first message in SIP INVITE request, or as per OMA CPM without first message in SIP

INVITE request. RCC.07 specifies that the NNI between two OMA CPM networks should not carry a message in the SIP INVITE request, and sent to the conference factory.

If interworking is not required for the first message in SIP INVITE request, then the feature tag of the request shall be mapped by the terminating Messaging Server according to the messaging technology used by the operator before relaying to the endpoints. An interworking function can be either collocated with the originating Messaging Server, with the terminating Messaging Server, or a separate network entity on the terminating or originating side. As such, the interworking function needs to be aware of the far-end network and the technology used in such network.

The SIP INVITE request in case of OMA CPM based chat sessions will be populated as follows;

- P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.oma.cpm.session,
- Accept-Contact and Contact header fields containing feature tag +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.session",
- Contact header field without "isfocus" feature tag,
- Request URI is set to a user's address (not to a Group Chat session identity) and username part of the URI in P-Asserted-Identity header field different from "rcse-standfw".

When using OMA SIMPLE IM the SIP INVITE header fields will be populated in the following manner;

- Accept-Contact and Contact header fields containing feature tag +g.oma.sip-im and without SDP containing a=file-selector,
- Contact header field without "isfocus" feature tag,
- Request URI is set to a user's address (not to a Group Chat session identity) and username part of the URI in the P-Asserted-Identity header field different from "rcse-standfw".

Disposition notifications are carried in SIP MESSAGE requests with the feature tag included, and mapped by the terminating Messaging Server according to the messaging technology used before relaying to the end-point devices.

The MESSAGE method used for disposition notifications used for both OMA CPM based Chat and Standalone messaging are populated as follows;

- P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.oma.cpm.msg.
- Accept-Contact header field containing feature tag +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.msg" and carrying content of the type message/imdn+xml.

For SIMPLE IM based Chat disposition notifications the MESSAGE will use the feature tag: +g.oma.sip-im.

Once the MSRP session is established, an end-to-end Chat session is in place: The interworking function acts as a Back-to-Back User Agent (B2BUA) to relay all messages and notifications exchanged between the sender and the recipient.

4.4.2.1.2 Addressing & Routing

Messaging services such as Chat conform to addressing guidelines specified for IMS based multiparty services defined in GSMA PRD IR.65, which refers to procedures is defined in 3GPP TS 23.228 and address formats defined in 3GPP TS 23.003.

Chat clients also conform to routing guidelines specified in GSMA PRD IR.65 requirements for routing between IMS networks as specified in 3GPP TS 29.165, and DNS query handling specified in GSMA PRD IR.67 to ensure end-to-end service interoperability.

Inter-operator aspects for Messaging services are specified in GSMA PRD IR.90 and are applicable to both an individual implementation or as part of the RCS framework.

4.4.2.1.3 Media Handling

RCC.07 indicates that during normal session setup according IETF RFC 4975, each end-point and interworking function identify the maximum chat message size allowed during the session, as well as the allowed content-types. Any failure of delivery of a message because of size or content-type issues will be indicated via the MSRP procedures defined in IETF RFC 4975.

As indicated in section 5.3.2.1.3 MSRP message chunking requirements specified in GSMA PRD IR.90 are also applicable for Chat services.

4.5 Group Chat

4.5.1 Use Case

The following diagram illustrates the addition of a participant in a Group chat session, where the Messaging Application Server (Controlling Function) associated with the user initiating the chat coordinates the session.

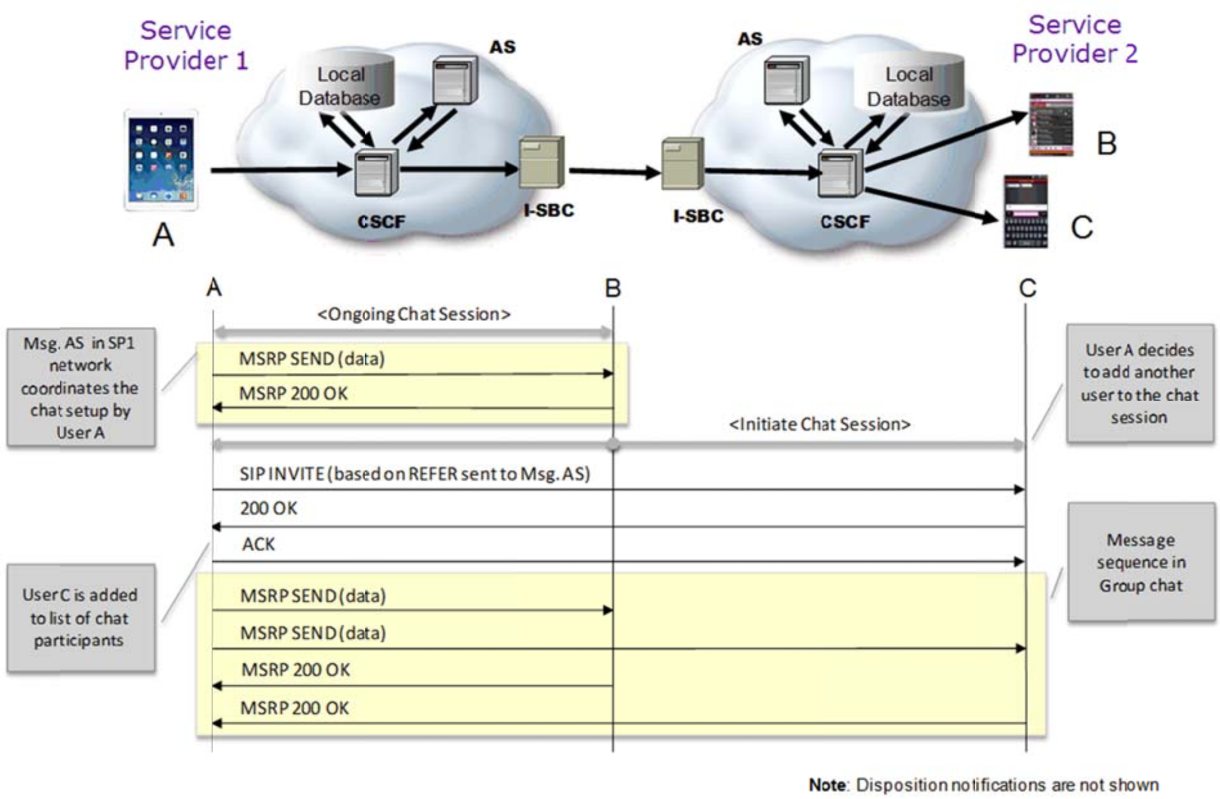

Figure 4. 13 - Adding new users to a Group Chat

Additional Group chat scenarios including regular and closed group chat sessions are described in GSMA PRD RCC.07.

4.5.2 SDO Assessment/Gap Analysis

4.5.2.1 GSMA

4.5.2.1.1 Session Establishment & Signaling

GSMA PRD IR.90 specifies that the feature tag of the request to establish a Group chat session shall be mapped at the terminating network according to the messaging technology used before relaying to the endpoints. Chat session feature tags are defined in GSMA PRD RCC.07.

The session establishment of a Group Chat is almost identical to a One-to-One Chat session. Similar applies to the delivery of notifications. If Chat session identifiers are not present in the incoming SIP INVITE request, then the terminating Messaging Server implementing OMA CPM will add these.

The SIP INVITE, SUBSCRIBE, and REFER methods used in case of OMA CPM based chat sessions will be populated as follows:

- P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.oma.cpm.session,
- Accept-Contact and Contact header fields containing feature tag +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.session",
- Contact header field with "isfocus" feature tag.

In case of CPM based chat sessions rejoin or restart, the Contact header field in SIP INVITE, SUBSCRIBE, and REFER methods will include the Request URI set to a Group Chat session identity, besides the use of the "isfocus" parameter.

When using OMA SIMPLE IM the header fields in SIP INVITE, SUBSCRIBE, and REFER methods will be populated in the following manner:

- Accept-Contact and Contact header fields containing feature tag +g.oma.sip-im and without SDP containing a=file-selector,
- Contact header field with "isfocus" feature tag.

In case of SIMPLE IM based chat sessions rejoin or restart, the Contact header field in the SIP INVITE request will include the Request URI set to a Group Chat session identity, besides the use of the "isfocus" parameter.

The disposition notifications are carried in SIP MESSAGE requests in the same manner as for One to one messaging with the feature tag included, and mapped by the terminating Messaging Server according to the messaging technology used before relaying to the end-point devices.

4.5.2.1.2 Addressing & Routing

Messaging services such as Chat conform to addressing guidelines specified for IMS based multiparty services defined in GSMA PRD IR.65, which refers to procedures is defined in 3GPP TS 23.228 and address formats defined in 3GPP TS 23.003.

Chat clients also conform to routing guidelines specified in GSMA PRD IR.65 requirements for routing between IMS networks as specified in 3GPP TS 29.165, and DNS query handling specified in GSMA PRD IR.67 to ensure end-to-end service interoperability.

Inter-operator aspects for Messaging services are specified in GSMA PRD IR.90 and are applicable to both an individual implementation or as part of the RCS framework.

4.5.2.1.3 Media Handling

According to IR.90 if a chat message being sent is larger than the negotiated message size between the conference focus and the recipient, it is also limited by the maximum message size specified by each side. Consequently, the conference focus will generate a delivery failure report to the message sender, provided that the sender subscribes to the delivery notification. As such, the message size incompatibility may cause a delivery failure of a chat message to some participants across Service Providers. However, the receiver is currently not notified of the failed message delivery due to size incompatibility.

A similar situation occurs when multimedia content is exchanged at the Group Chat session. If the terminating network or the terminating end-points do not support multimedia content, the Messaging Server will not deliver the multimedia content to those participants. As such, the multimedia content incompatibility may cause a delivery failure of a chat message to some participants across Service Providers.

As indicated in section 5.3.2.1.3 MSRP message chunking requirements specified in GSMA PRD IR.90 are also applicable for Chat services.

4.6 File Transfer

4.6.1 Use Case

The following diagram illustrates a successful File Transfer during a chat session applying the MSRP mechanism.

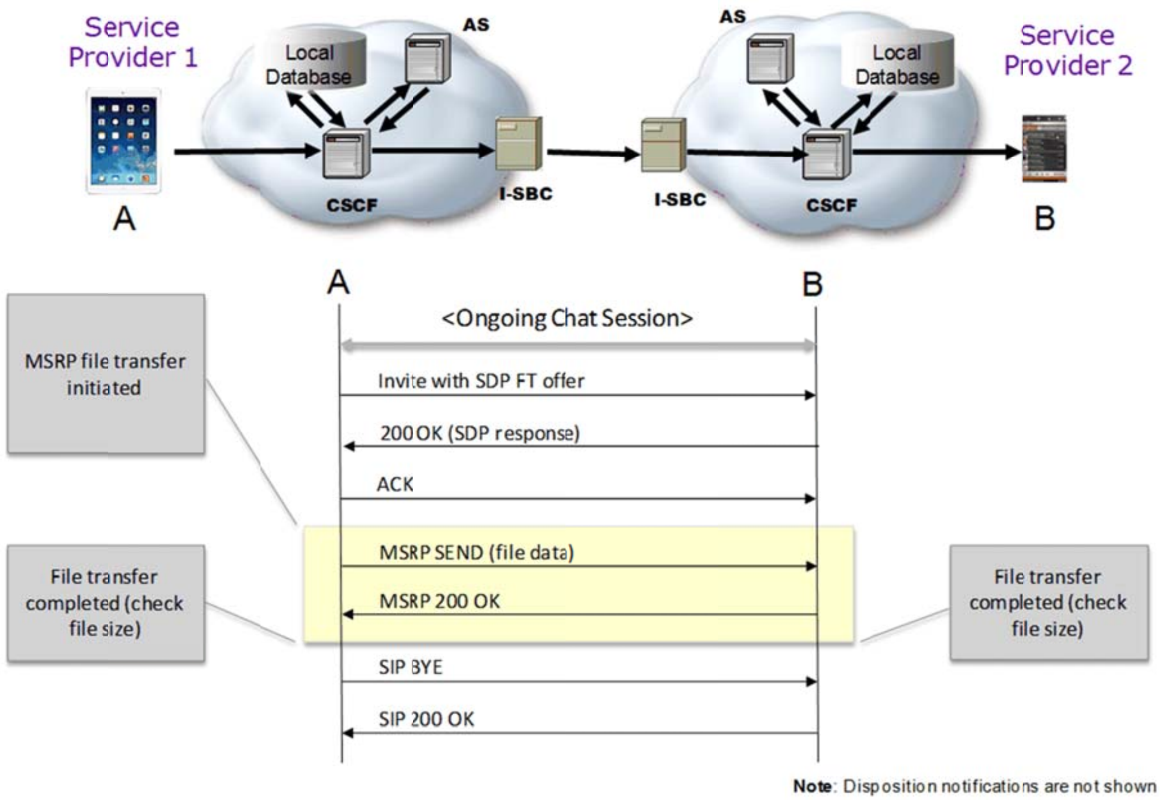

Figure 4. 14 - File Transfer in a Chat Session

Besides MSRP, RCS implements HTTP based File Transfer. Additional File Transfer scenarios and options are described in GSMA PRD RCC.07.

4.6.2 SDO Assessment/Gap Analysis

4.6.2.1 GSMA

4.6.2.1.1 Session Establishment & Signaling

The feature tag of the request shall be mapped at the terminating network according to the underlying messaging technology and respective geolocation capability before relaying to the endpoint. File transfer feature tags are defined in GSMA PRD RCC.07.

In case a file transfer is established during a One to one chat session, the SIP INVITE used will include the respective feature tag of the message technology selected (OMA CPM or OMA SIMPLE IM) as described in GSMA PRD IR.90.

In case of OMA CPM based file transfer, the following header fields need to be set as follows.

- P-Asserted-Service: urn:urn-7:3gpp-service.ims.icsi.oma.cpm.filetransfer.
- Accept-Contact and Contact header fields containing the feature tag +g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.oma.cpm.filetransfer".

When using OMA SIMPLE based file transfer the SDP to be negotiated will contain the parameter a=file-selector.

For the transfer of Geolocation information consult section 5.8 of this document.

4.6.2.1.2 Addressing & Routing

File transfer associated with messaging services conforms to addressing guidelines specified for IMS based multiparty services defined in GSMA PRD IR.65, which refers to procedures defined in 3GPP TS 23.228 and address formats defined in 3GPP TS 23.003.

File transfer clients also conform to routing guidelines specified in GSMA PRD IR.65 requirements for routing between IMS networks as specified in 3GPP TS 29.165, and DNS query handling specified in GSMA PRD IR.67 to ensure end-to-end service interoperability.

Inter-operator aspects for File transfer services are specified in GSMA PRD IR.90 and are applicable to both an individual implementation or as part of the RCS framework.

4.6.2.1.3 Media Handling

The media transfer technology (based on MSRP) is the same in the OMA IM and CPM technologies. As such, the same considerations made for Standalone messaging; One-to-one chat and Group chat apply to the delivery of the information that enables the receiver to fetch the file and the delivery notifications.

GSMA PRD RCC.07 also specifies File Transfer via HTTP as alternative mechanism to transfer the files, either uploading a file to a server in the originating network using a HTTPS POST procedure, or sending the link to the file and, when applicable, the thumbnail correspondent to the file together with some additional data (validity and size) to the receiver using a messaging service. File delivery notifications also rely on the mechanism used for delivery and display notifications associated to the messaging service used. Once the receiver gets notified of the link where the file resides, downloading the content takes place using a HTTPS GET procedure.

4.7 Content Sharing

4.7.1 IMS based Content Sharing

The following diagram illustrates video sharing within a voice session.

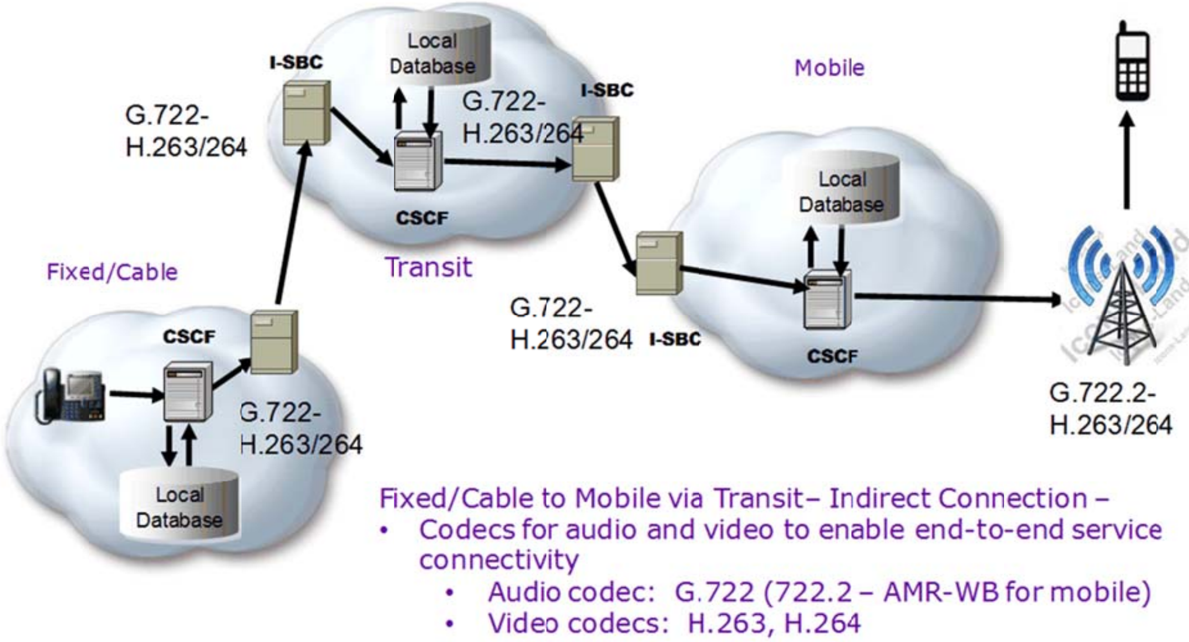

Figure 4. 15 - Content Sharing – Video with Voice - Indirect Connection

The following diagram illustrates video sharing outside of a voice session.

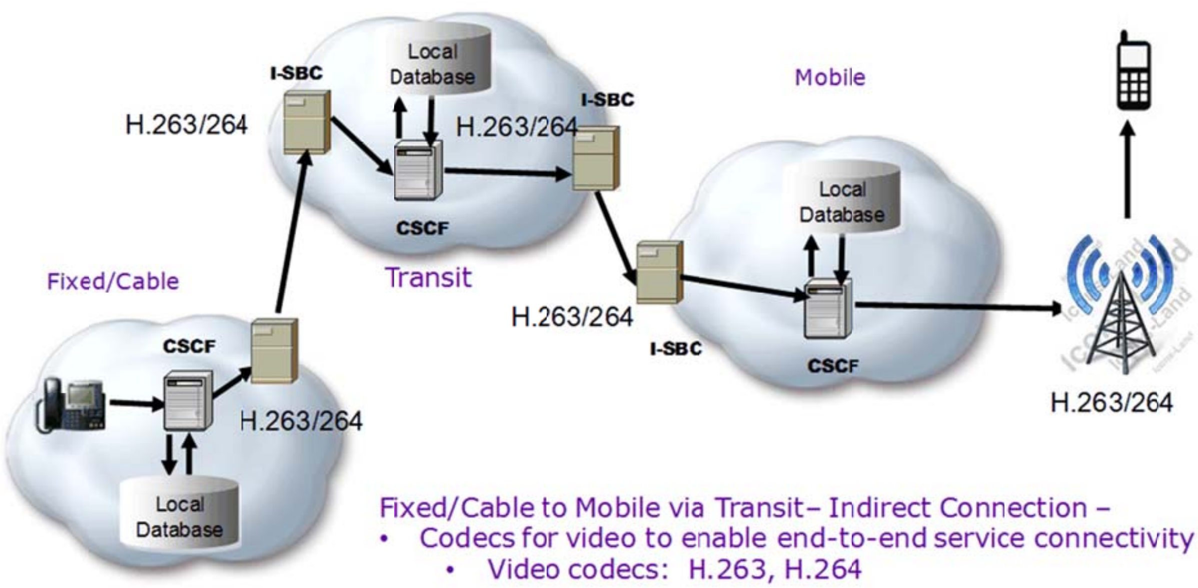

Figure 4. 16 - Content Sharing – Video w/o Voice - Indirect Connection

The following diagram illustrates sharing of a still image within a voice session.

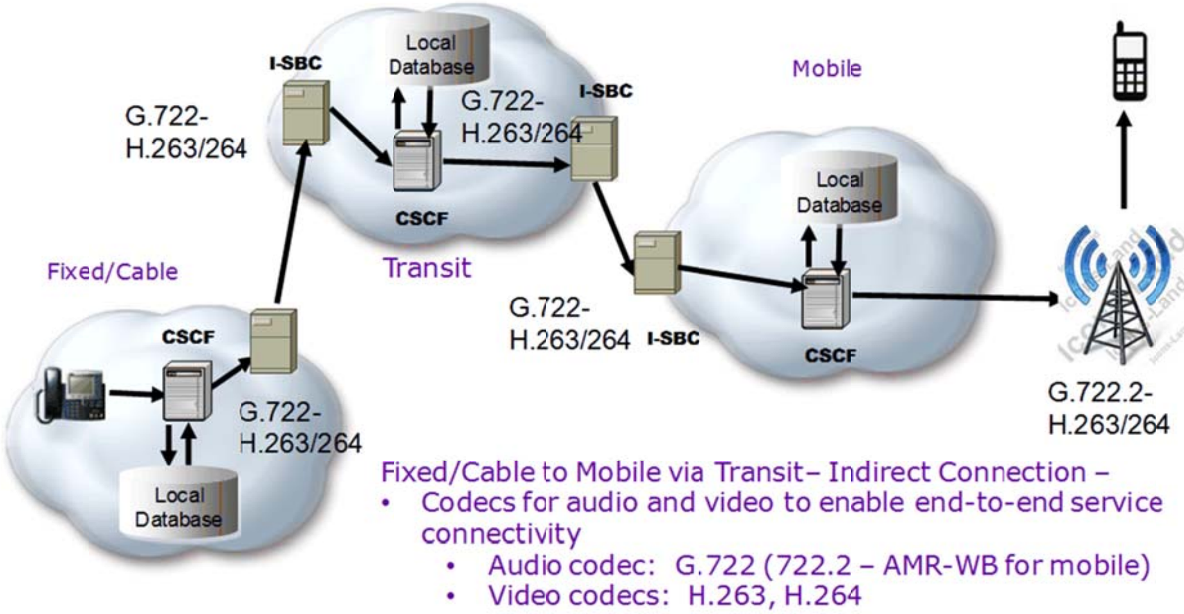

Figure 4. 17 - Content Sharing – Still Image with Voice - Indirect Connection

4.7.2 SDO Assessment/Gap Analysis

4.7.2.1 GSMA

4.7.2.1.1 Session Establishment & Signaling

The content share services as specified in GSMA provide the capability to share videos and pictures (images) in near real-time.

Video content sharing services are specified in GSMA PRDs IR.74 and IR.84; differentiation is based on service capabilities, i.e., video sharing done within a call or outside of a call. Image content sharing service is specified in GSMA PRD IR.79 and allows sharing of images during a call.

Content share services can be used in stand-alone mode or as part of RCS as defined in GSMA PRD RCC.07.

For the content sharing during a call, a user needs to be able to recognize if the conversation partner can support or has activated required content sharing services (i.e., Video and Image Share) based on the exchange of capabilities between conversation partners or queried from a server.

When sharing content during a voice call according to GSMA PRDs IR.74 and IR.79, the service is linked to the call and it will also automatically terminate when the call ends.

Conversely, it is also possible to use share video content according to GSMA PRD IR.84 with the conversation partner during a voice call. As such, Video Share session is established independently of the voice call, and the sharing session may continue after the voice call ends, and explicitly terminated.

The 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.

Setting up a content sharing session is very similar with or without a voice call. The main difference for Video Share is that in the latter case the INVITE request includes instead of the complementary services the feature tag “+g.3gpp.cs-voice” as specified in GSMA PRD IR.74, the ICSI feature tag with the MMTel ICSI and the IARI feature tag with the Video Share IARI (according to GSMA PRD IR.84), i.e., +g.3gpp.icsi-ref=”urn%3Aurn-7: %3A3gpp-service.ims.icsi.mmtel” and +g.3gpp.iari-ref=”urn%3Aurn-7%3A3gppapplication. ims.iari.gsma-vs".

For Video or Image Share with a voice call, the +g.3gpp.cs-voice feature tag is added to the Accept-Contact and Contact headers of the SIP INVITE request. For Image Share according to GSMA PRD IR.79, clients shall not include the MMTel ICSI urn:urn-7:3gpp-service.ims.icsi.mmtel but add the Image Share IARI urn:urn-7:3gpp-application.ims.iari.gsma-is in these requests.

4.7.2.1.2 Addressing & Routing

Content sharing services conform to addressing guidelines for IMS based multiparty services mentioned in GSMA PRD IR.65, which refers to procedures defined in 3GPP TS 23.228 and address formats defined in 3GPP TS 23.003.

Conversational video clients also conform to routing guidelines specified in GSMA PRD IR.65 requirements for routing between IMS networks as specified in 3GPP TS 29.165, and DNS query handling specified in GSMA PRD IR.67 to ensure end-to-end service interoperability.

4.7.2.1.3 Media Handling

To guarantee the interoperability of clients during Video Share scenarios, all devices supporting the Video Share service (during or without a call) shall, at least, support the following video format:

- Video format: H.264/MPEG-4 (Moving Pictures Experts Group) Part 10 // AVC (Advanced Video Codec).
- H.264 Profile: Baseline Profile (BP).
- H.264 Level: 1b (shall be encoded using the profile-level-id set to 0x42900B and the H.264 Constrained Baseline Profile 1b is 0x42D00B).

NOTE: Please note that including this, it is highly recommended to support also the H.263-2000 codec with profile 0 Level 45, which is mandatory in RCS Release 1-4 Video Share that is based on [PRD-IR.74].

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

Video formats must be listed in order of preference in the SDP media description as specified in RFC3264. Hence, additional codecs providing better quality than these mandatory ones should be listed in the SDP before the mandatory codecs.

In any case for the encoding of the actual stream should be adapted to the currently available bandwidth and might therefore use bit rates lower than the maximum negotiated during session setup. To support this RTCP Receiver Reports (RR) shall be sent at least at a rate of one RR per second.

Note that in H.264, support of a certain level implicitly requires support for all lower levels, so a client supporting other H.264 levels should only indicate the highest level per profile that it supports.

RTP payload handling shall be as described in section 7.4.3 of [3GPP TS 26.114] for the H.264 (AVC) video codec.

The receiving clients should preserve the aspect ratio of the incoming video stream, avoiding that video is stretched to fit the UI. The sending client may redefine the aspect ratio when supporting a flexible handling interface that could alternate between landscape and portrait (e.g., from 4:3 to 3:4 after the sending device has been rotated).

The originator of the Video Share session can indicate support for both Baseline (BP) and Constrained Baseline (CBP) Profiles. The originator shall never use Flexible Macroblock Ordering (FMO), Arbitrary Slice Ordering (ASO), or Redundant Slices (RS) features of the profile no matter what the receiving client selects.

According to GSMA PRDs IR.74 and IR.84 a device shall support the H.264 video codec with baseline (and optionally Constrained Baseline Profile) profile and level 1.3 to provide 768 kilobits per second (kbps) video over an LTE bearer or over a similar high bit rate bearer.

If a second Video Share session is established in parallel, the H.264 video codec with baseline profile (and optionally Constrained Baseline Profile) and level 1.2 shall be used instead. The assumption for the use of a high bit rate bearer is that the connectivity and video parts of both terminals support it and have LTE or another high bit rate broadband access; otherwise the video bit rate will be reduced to the level 1b to assure compatibility.

4.8 Geolocation Services

4.8.1 Use Case

The following diagram illustrates a successful File Transfer of Geolocation information during a multimedia service session based on the PUSH method and application of MSRP technology.

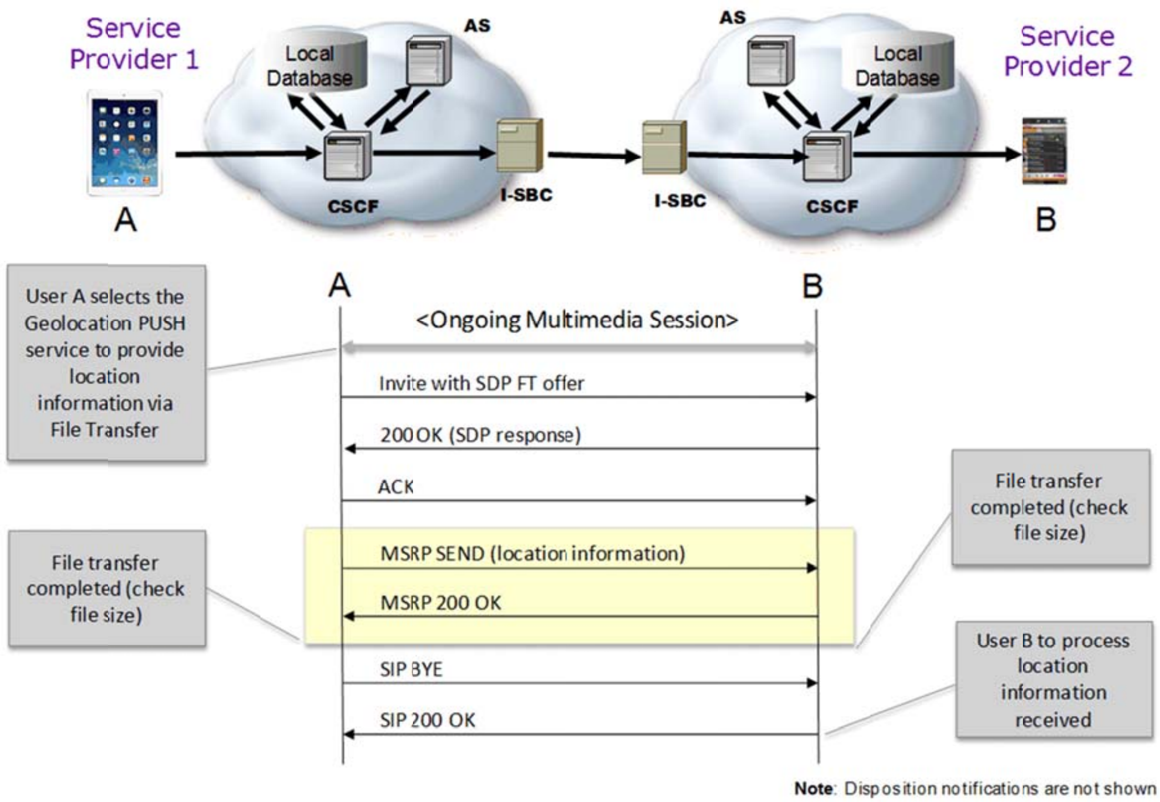

Figure 4. 18 - Push of Geolocation information using File Transfer

Additional Geolocation information delivery scenarios based on PUSH and PULL methods are described in GSMA PRD RCC.07.

4.8.2 SDO Assessment/Gap Analysis

4.8.2.1 GSMA

4.8.2.1.1 Session Establishment & Signaling

If Geolocation information needs to be provided (PUSH) during a voice call (content share) the respective SIP INVITE to start the MSRP session required for file transfer and be populated as follows.

- Accept-Contact and Contact header fields will contain the feature tag; +g.3gpp.iari-ref="urn%3Aurn-7%3A3gpp-application.ims.iari.rcs.geopush".
- The Request URI will set to a user's address and NOT to a Group Chat session identity.

If Geolocation information is requested (PULL) then the SIP INVITE needed to establish the MSRP involved in transfer to populate the Accept-Contact and Contact header fields with the feature tag: +g.3gpp.iari-ref="urn%3Aurn-7%3A3gpp-application.ims.iari.rcs.geopullft"

"Show us on a map" capabilities require the SIP INVITE needed to establish the MSRP session for the transfer to populate the Accept-Contact and Contact header fields with the feature tag: +g.3gpp.iari-ref="urn%3Aurn-7%3A3gpp-application.ims.iari.rcs.geopush". If the Geolocation information is requested from the Chat focus then Contact header field will include the Request URI set to the Group chat session identity. Conversely if the information is requested by the focus the Contact header will contain the "isfocus" feature tag.

4.8.2.1.2 Addressing & Routing

The transfer of Geolocation information in RCS conforms to addressing guidelines specified for IMS based multiparty services defined in GSMA PRD IR.65, which refers to procedures defined in 3GPP TS 23.228 and address formats defined in 3GPP TS 23.003.

RCS clients also conform to routing guidelines specified in GSMA PRD IR.65 requirements for routing between IMS networks as specified in 3GPP TS 29.165, and DNS query handling specified in GSMA PRD IR.67 to ensure end-to-end service interoperability.

Inter-operator aspects for Geolocation services are specified in GSMA PRD IR.90.

4.8.2.1.3 Media Handling

Transfer of Geolocation information requires file transfer capabilities (based on MSRP) defined for RCS in the same manner as applicable for OMA SIMPLE IM and OMA CPM messaging technologies. As such, the same considerations apply to the delivery of the information that enables the user to fetch or to send Geolocation information.

4.9 Presence

4.9.1 SDO Assessment/Gap Analysis

4.9.1.1 GSMA

4.9.1.1.1 Session Establishment & Signaling

The Social Presence Information is controlled and configured by a user and exchanged when establishing Social Presence relationship with a contact in the user's Enhanced Address Book. As such, Social Presence Information is visible not only to contacts but also from other places on a device, like communications log files, message folders or on the other devices of the user.

Social Presence Information allows users to set their availability status (available and willing or unavailable or busy). These states are informative but permanent (until changed) and set the stage for other RCS capabilities. Social Presence Information can be shared (or not) by either accepting or ignoring an invitation (no answer), as well as by blocking invitations. As such, a Social Presence relationship can be also revoked. The exchange of Geolocation information is also associated with Social Presence.

According to RCC.07, a Presence mechanism based on OMA SIMPLE Presence can be also used for the exchange of capability information, but contrary to the SIP OPTIONS mechanism, does not involve an end-to-end transaction, as the capabilities are queried against a server using the standard OMA SIMPLE Presence procedures. As such, the Presence server acts as a source for both Social Presence Information and capability information.

4.9.1.1.2 Addressing & Routing

The Presence services (Social Presence Information and capability information exchange) specified for RCS and defined in GSMA PRD RCC.07 conform to addressing guidelines specified for IMS based multiparty services defined in GSMA PRD IR.65, which refers to procedures defined in 3GPP TS 23.228 and address formats defined in 3GPP TS 23.003.

RCS clients also conform to routing guidelines specified in GSMA PRD IR.65 requirements for routing between IMS networks as specified in 3GPP TS 29.165, and DNS query handling specified in GSMA PRD IR.67 to ensure end-to-end service interoperability.

Inter-operator aspects for Presence services are specified in GSMA PRD IR.90.

4.9.1.1.3 Media Handling

The exchange of Social Presence Information requires use of non-anonymous SUBSCRIBE/NOTIFY with event = "presence", and the use of XCAP GET for service "presence content XDMS", as specified for the NNI in GSMA PRD IR.90. If Presence is required for the exchange of capability information makes use of the SIP PUBLISH method to announce client/device. The exchange of client/device capabilities using the Presence mechanism does not require XDM interworking between two networks.

Users may also retrieve the service capability information of contacts even if they do not have an established Social Presence relationship using an Anonymous Fetch. This operation will result in a single NOTIFY request indicating the service capabilities of that contact.

4.10 Video Conference/Collaboration

4.10.1 Use Case 1

4.10.2 SDO Assessment/Gap Analysis

4.10.2.1 GSMA

4.10.2.1.1 Session Establishment & Signaling

The HDVC service as specified in GSMA PRD IR.39 comprise point-to-point video calls and video conferences with one full duplex audio stream with tight synchronization to one main video stream and another video stream aimed for sharing content.

HDVC clients must support the SIP registration procedures specified in GSMA PRDs IR.92 and PRD IR.94.

If the HDVC device has no access to a Universal Integrated Circuit Card or is configured not to use the identity associated to a UICC then it shall follow the procedures in 3GPP TS 24.229 for IMS registration and authentication using SIP digest with or without TLS and using configured identity parameters specified in GSMA PRD IR.39.

The HDVC device does need to use signaling compression when the initial IMS registration is performed via higher-bandwidth networks such as LTE.

4.10.2.1.2 Addressing & Routing

The HDVC services conform to addressing guidelines specified for IMS based multiparty services defined in GSMA PRD IR.65, which refers to procedures is defined in 3GPP TS 23.228 and address formats defined in 3GPP TS 23.003.

HDVC clients also conform to routing guidelines specified in GSMA PRD IR.65 requirements for routing between IMS networks as specified in 3GPP TS 29.165, and DNS query handling specified in GSMA PRD IR.67 to ensure end-to-end service interoperability.

4.10.2.1.3 Media Handling

GSMA PRD IR.39 defines the following video media requirements:

- HDVC clients fulfill voice and video media requirements defined in GSMA PRD IR.92 (VoLTE) and GSMA PRD IR.94 respectively.
- In addition, HDVC supports a wide range of video conferencing clients, such as high end dual HD-screen video conference devices connected over high speed fixed broadband IP access to downloaded HDVC apps on tablets or smart phones connected via Wi-Fi or cellular access without session based QoS control.
- HDVC clients support the Adaptive Multi-Rate (AMR) speech codec for narrow band voice and the AMR wideband (AMR-WB) codec for wideband voice. In absence of AMR or AMR-WB speech codec HDVC clients support the ITU-T G.711 and the ITU-T G.722 speech codec.
- If full band voice is supported, wideband and narrowband must also be supported. Consequently, the client should support ITU-T G.719 codec or AAC-LD codec (ISO/IEC 14496-3:2009).
- The entities in the IMS core network that terminate the user plane shall support transcoding between these codices.
- Support for H.264 main and high profiles are optional for the Access Unaware HDVC UE and the network, but when supported SD video and HD video content shall comply with ETSI TS 101 154.
- Screen sharing shall be supported in an HDVC device and in the network by a dedicated video stream to the SIP session, typically using low frame rate and high resolution, similar to what is specified for ITU-T Recommendation H.239.
- If end-points support the same profile, but offer different levels, then the end-point offering the higher level must be able to adapt to the lower level, as specified in IETF RFC 6184. Hence, asymmetry in receive and transmit capability must be supported by all end-points.
- The bandwidth SDP parameter is mandatory on media level and optional on session level.

NOTE: IR.94 compliant clients as described in sections 5.2.3 and 5.2.4 in this document do not support all HDVC specific requirements, but still may be used to participate in High Definition Video Conference.

4.11 Social Networking

Social Networking can be broadly defined as an internet or mobile-based social space where people can connect, communicate, and create and share content with others. Social networking incorporates capabilities such as Presence, Location, Messaging, File Transfer, and Content Sharing that are described in this document.

A number of the social networking services in use today are proprietary and use proprietary protocols. At this time, social networking is considered out of scope for this landscape effort.

It should be noted that the World Wide Web Consortium (W3C) has just chartered a Social Web Working Group, which will define the technical standards and APIs to facilitate access to social functionality as part of the Open Web Platform. These include a common JSON-based syntax for social data, a client-side API and a Web protocol for federating social information such as status updates.

5 Conclusions & Recommendations

While IP Services may be deployed on both softswitch based and IMS architectures, most of the analysis for the services beyond HD voice and video assumes IMS end-to-end.

The key technical areas of session establishment and signaling, addressing and routing, and media handling were assessed across each IP service. The following provides a summary assessment and the key recommendations to be addressed in the development of detailed Interconnect profiles for the identified IP services.

5.1 Session Establishment & Signaling

For both Softswitch based and IMS architectures, SIP session establishment signaling follows the IETF SDP offer/answer procedures defined in RFC 3264.

5.1.1 Early Media Support

Early media is the concept of delivering a media stream (e.g., audio and video) prior to call answer or session establishment. Support for early media is important both for interoperability with the Public Switched Telephone Network (PSTN) and for billing purposes. Varying implementations exist for the support of early media. In some cases early media is indicated by the presence of a P-Early-Media header in a 18X response.

It is recommended that the ATIS/SIP Forum NNI Task Force define the procedures for early media across the NNI as part of the IP Interconnect profile, and that the same procedures be used for other services.

5.1.2 Support of QoS preconditions

Preconditions are a set of constraints about a session that are introduced in the offer. The recipient of the offer generates an answer but does not alert the user or proceed with the session establishment until after the preconditions are met. Quality of service preconditions are included in the SDP description. PacketCable and some other industry segments do not currently support the use of preconditions.

It is recommended that the ATIS/SIP Forum NNI Task Force evaluate support in the North American Service Provider community for use of preconditions; as a prerequisite for any technical specification of their use.

5.1.3 Capability Discovery

It is sometimes desirable to be able to discover if the far end supports video calls across the NNI. Using OPTIONS for this capability discovery across the NNI is optional as indicated in IETF RFC 3840. OPTIONS messaging can be used to discover far end video support status without trying to establish a video session. Trying to establish a video session implies the far end user may be alerted which may not be appropriate for capability discovery.

Rather than incur the extra call setup signaling with SIP OPTIONS, a database lookup with local caching may be employed for capability discovery. For further details, reference the Address and Routing section.

The User Capability Exchange (UCE) mechanisms defined in GSMA PRD RCC.07 (RCS 5.1) is addressing end-to-end interoperability via Presence and SIP Options and should be considered moving forward. It is recommended that a preferred capability discovery mechanism be specified as part of the detailed IP Services NNI specification.

5.2 Addressing and Routing

For IP services SDOs for different industry segments have defined addressing and routing standards to enable end-to-end connectivity and within each segment of the industry, e.g., wireless, packet cable, landline, various options exist. The key step in determining whether one of these IP services can be made available end-to-end are the specifications in the interconnect agreement and the ability to discover if the terminating end supports a specific service across the NNI.

A database or discovery lookup may optimize the efficiency of the network rather than incur the extra call setup signaling if the majority of these sessions would be rejected or downgraded. For example, originating networks can screen/update to exclude certain service attributes to minimize the chances of call failure due to far end device's inability to handle that particular service. Furthermore, end users can be presented with advanced call options if it is known that features, such as video or content sharing, are supported end-to-end. A standalone service capability database separate from one that also serves a routing function seems unlikely, however.

Current inter-network routing solutions such as are defined in GSMA PRD IR.67, assume that the point of interconnection at which a session request is handed off, is determined only by the called number. It is possible to make this more efficient by allowing consideration of additional factors, such as the source of the request (the network from which it originated) and the service(s) requested. This would allow for example:

- Session requests to be sent to interconnect points that have been "tailored" by the terminating network per the requirements of the originating network.
- Session requests specifying services that the terminating network provides via a 3rd party, to be routed by the originating network directly to the 3rd party or to an interconnect point specified by the terminating network for interconnection to the 3rd party.

In review of the published addressing and routing standards across the SDOs, much information exists on the use of local databases but a consensus on how routing and service discovery data is provisioned and made available across/between networks to enable addressing and routing has not been achieved. Some of the options in use across the industry segments include:

- Peering partners share each other's capabilities and/or routing information through bilateral information exchange
- A registry database is configured with routing information and/or supported services of the far end party, Such a database may provide information for real time query or download and may provide the information directly or via a pointer leading to the dynamic discovery approach below
- Dynamic discovery via ENUM or other protocol to the far end network,
- SIP OPTIONS message negotiates the desired service with the far end (capability only)

Further study is recommended on potential enhancements to inter-network routing that enable consideration of attributes such as originating network and requested service(s).

5.3 Media Handling

Subscribers are now demanding a very rich user experience due to the sophistication of smartphones and the popularity of many other communications-enabled devices. The all IP network provides the service provider community with technology to provide that experience ubiquitously and end-to-end. Selection of the best quality codec is desirable for a rich IP experience, and should be prioritized in the offer list. However, the diversity of codecs across industry segments presents challenges to end-to-end multimedia services.

Maximizing Transcoder-Free Operation (TrFO) is a common industry objective for the all IP network. The Joint ATIS/SIP Forum IP NNI taskforce is defining an IP Interconnection Profile (document IPNNI-2014-00011 at the time of writing) that encourages the avoidance of transcoding. It is recommended however that Service Providers support transcoding, as to fail to do so may lead to excessive call failures or undesirable fallback to circuit switched communication.

Because transcoding between wideband and narrowband codecs is computationally intensive it is recommended that a device supporting both wideband and narrowband codecs, that requests use of one or more wideband codecs in an SDP offer it originates, also indicate in that offer at least one supported narrowband codec. This will allow networks that must transcode the media to do so more efficiently.

To avoid the scenario where transcoding is performed several times, applicable codecs used at the NNI should be restricted as little as possible in the inter-operator agreements. Operators will generally apply codec policies to SDP offers at the egress point from their network facing an NNI, in a way that reflects their interconnect agreement with the party on the other side of the NNI and/or the destination network. The nature of those policies is to add codecs to the end of the list of codecs already in the offer when it reaches that point. Other actions, e.g., removing or re-ordering codecs, may be necessary to accommodate exceptional circumstances. Caution must be taken when implementing such policies, however, to avoid creating a need for transcoding.

It is desirable that the “burden” of transcoding be evenly borne by the parties on either side of an NNI. This avoids the overhead of accounting and charging for transcoding. If transcoding is necessary it is preferable to select the first applicable one in the offer set in order to avoid multiple transcoding.

It is recommended that the ATIS/SIP Forum joint taskforce address minimizing transcoding as part of the NNI Profile.

Video services face additional challenges in media compatibility with the requirement to handle various screen sizes and resolution. The H.264 (Advanced Video Coding) is compatible between PacketCable and 3GPP or 3GPP2 networks and therefore supports rich end-to-end video experiences. Recommendations are similar for both conversational/2-way services using video. Video sessions using different codec families across the NNI should either be transcoded by the network, similar to how audio is transcoded, or fallback to high quality audio or basic voice in that order. Similarly Group Chat (and one-on-one chat) use of multimedia should retain the text portion for any party participating on the other side of an NNI that is not compatible with the media.

As any existing or new codecs become prevalent, the industry should revisit the IP Interconnection Profiles in order to keep TrFO and service evolution in balance. In any case, subscribers have numerous options to enjoy enriched services via “over the top” applications rather than wait for end-to-end service provider media and application interoperability so scalable industry wide mechanisms for bilateral and transit interoperability should be considered for the IP NNI profile with some urgency.

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7 Participating Companies

At the time of consensus on this document, IP Services Interconnect Focus Group, which was responsible for its development, had the following leadership:

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