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Work stream "Technical Aspects", "Service Requirements"

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Enabling HD voice continuity in international calls

(Release 1.0) 2013-2014

Disclaimer: This document 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.

i3 forum recognizes that high definition customer experience is related not only to codecs but also to the HD capabilities of devices and other components.

For the purpose of this document the term "*HD Voice*" relates only to the transport of international HD voice communications.

Revision history

Date	Rel.	Subject/Comment
May 11th, 2014	1.0	First release of Enabling HD voice continuity in international calls

Executive summary

The trend towards an all-IP communication pattern will enable an enrichment of the user communication experience through the usage of larger frequency bandwidth (wideband, superwideband and up to fullband) codecs. These codecs substantially improve the quality of VoIP calls; therefore their adoption by the Service Providers is likely to grow during the next years.

In addition, subscribers of some Service Providers or some OTT Providers are already experiencing the quality improvement enabled by the usage of such codecs.

Being a technology whose deployment is expected to become common in all kinds of VoIP Service Providers, it is worth considering how HD voice may affect the international VoIP traffic and more specifically the IPX paradigm.

This document describes the different scenarios that a Carrier or an IPX Provider might encounter when connecting VoIP sessions between two Service Providers, taking into account that the Carrier/IPX Provider might be:

1. The only Carrier/IPX Provider present in the call path.
2. The first in a chain of Carriers/two IPX Providers.
3. The last in a chain of Carriers/two IPX Providers.

A Carrier can offer high quality[1] Voice services either by its own private/public VoIP platform or via its IPX based platform. It is a Carrier decision whether to pursue one or both alternatives. In this document, considering the vast specification activity on IPX, and the market reality which pushes for an adoption of an IPX platform for the high quality Voice service, the IPX platform is considered as the reference technical model.

When no common codec(s) can be agreed between the initiating and terminating Service Providers, transcoding is required for a successful communication. The i3forum documents Technical Interconnection Model for International Voice Services [1] and Voice over IPX Service Schedule [2] include some basic guidelines about who is responsible of performing this transcoding operation. Adding to the analysis the wideband codec(s) optional capability of Service Providers and Carriers/IPX Providers largely increases the number of cases to study; therefore this document takes those basic guidelines into account and makes some additional recommendations.

The meaning of “wideband codec support” differs slightly when applied to a Service Provider as to when it’s applied to a Carrier/IPX Provider: a Service Provider supports wideband codecs when it is able to offer a wideband codec session to another entity, or accept a wideband offer from another entity. A Carrier/IPX Provider supports wideband codecs when it is able to:

- Relay downstream a SIP INVITE containing a wideband offer without removing the wideband part.
- Insert a media relay device in the Voice Path of a call where a wideband codec has been negotiated.

1 In this document "high quality codec" refers to a codec with "wideband" frequency bandwidth [50Hz to 7,000Hz] or larger (up to superwideband [14Khz] or fullband [20Khz]). It has to be noted that this document addresses a broad set of high quality codecs including but not limited to the codecs dealt with by GSMA with the HD Voice specification

A Carrier/IPX Provider can optionally be able to transcode between two legs that have independently negotiated different codecs. The presence or absence or this capability is an important factor to consider when determining its VoIP traffic media policy.

GSMA recommendations for HD Voice in transport networks, as described in [3] and [4] are also acknowledged, and with regards to the codec recommendations specifically, they can be seen as a subset of the recommendations provided in this document

This document addresses mostly technical items, meaning that its main objective is to describe the action (or absence of action sometimes) that Carriers/IPX Providers can take to optimize the audio quality in each case, given the technical capabilities of all the elements (Service Providers and Carriers/IPX Providers), and the nature of the session offered by the originating Service Provider. A Carrier/IPX Provider has also to consider business factors and commercial agreements in order to decide the actual policy to implement.

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1 Scope of the document

The objective of this document is to provide the set of technical and networking features, as well as relevant service description and business models, that Carriers/IPX Providers will be required to implement in order to provide HD voice interconnections.

As per section 8 from [1] and section 8.4 from [2], for those Carriers/IPX Providers that support HD voice, two codecs are mandatory: G.722 for fixed networks and G.722.2 (AMR-WB) for mobile networks. Other high quality codecs providing superwideband or fullband are specified and in use, especially in the Over-The-Top (OTT) Providers domain.

In the next sections of this document "high definition codec" refers to a codec whose frequency lies in the wideband range (50Hz to 7,000Hz) or larger (up to superwideband [14kHz] or fullband [20kHz]). It has to be noted that this document addresses a broad set of high quality codecs including but not limited to the codecs dealt with by GSMA with the HD Voice specification.

A Carrier can offer high quality Voice services either by its own private/public VoIP platform or via its IPX based platform. It is a Carrier decision whether to pursue one or both alternatives. In this document, considering the vast specification activity on IPX, and the market reality which pushes for an adoption of an IPX platform for the high quality Voice service, the IPX platform is considered as the reference technical model.

2 Acronyms

3GPP	3rd Generation Partnership Project
ACELP	Algebraic Code Excited Linear Prediction Code
ALOC	Average Length of Call
AMR-NB	Adaptive Multi-Rate Narrow Band
AMR-WB	Adaptive Multi-Rate Wide Band
ASR	Average Seizure Ratio
HD	High Definition
IETF	Internet Engineering Task Force
KPI	Key Performance Indicator
LTE	Long Term Evolution
MNO	Mobile Network Operator
NB	Narrowband
OTT	Over The TOP
PSTN	Public switched Telephone Network
RCS	Rich Communication Suite
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SP	Service Provider
TrFO	Transcoder Free Operation
UAC	User Agent Client
UAS	User Agent Server
WB	Wideband

3 References

[1] i3forum, «Technical Interconnection Model for International Voice Services».

[2] i3forum, «Voice over IPX Service Schedule».

[3] GSMA, «ANNEXC: Minimum Requirements for Mobile Networks and Terminals for the usage of the 'HD voice' logo».

[4] GSMA, «HD Voice Annex F Minimum requirements with LTE,» April 2014.

[5] i3forum, «Voice over IPX Service Schedule».

[6] IETF, «WebRTC Audio Codec and Processing Requirements, draft-ietf-rtcweb-audio-04,» 2014.

[7] ITU-T, «G.711: Pulse Code Modulation of voice frequencies».

[8] ITU-T, «G.729 : Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction».

[9] ITU-T, «P.800: Methods for Subjective Determination of Transmission Quality».

[10] ITU-R, «BS.1534-1 : Method for the subjective assessment of intermediate quality level of coding systems,» 2003.

[11] ITU-T, «P.862: Perceptual evaluation of speech quality,» 2001.

[12] ITU-T, «P.862.2: WIdeband extension to recommendation P.862,» 2007.

[13] i3forum, «Voice Path Engineering in International IP-based Networks».

[14] ITU-T, «P.863 : Perceptual objective listening quality assessment».

[15] ITU-T, «G.107: The E-model: a computational model for use in transmission planning».

[16] ITU-T, «G.113: Transmission impairments due to speech processing».

[17] ITU-T, «G.107.1: Wideband E-model».

[18] ITU-T, «G.722: 7Khz audio-coding within 64 Kbit/s».

[19] ETSI, «TS 102 527-1: New Generation DECT; Part1: Wideband Speech».

[20] ITU-T, «G722.2: Wideband coding of speech ar around kbit/s using Adaptive Multi-Rate Wideband (AMR-WB)».

[21] IETF, «RFC 6716: Definition of the Opus Audio Codec».

[22] ITU-T, «G.719: Low-complexity, full-band audio coding for high qulity, conversational applications».

[23] GSMA, «IR.39 V1.0: IMS Profile for High Definition Voice Conference,» 2013.

[24] IETF, «RFC 3264: An Offer/Anwer Model with the Session Description Protocol».

[25] 3GPP, «TS 23.153: Out of band transcoder control».

[26] IETF, «RFC 4032: Sip Preconditions Framework».

[27] IETF, «RFC 5898: Connectivity Preconditions for SDP media streams».

[28] IETF, «RFC 3312: Integration of Resource Management and SIP».

[29] IETF, «RFC 5027: Security Preconditions for SDP media streams».

[30] GSMA, «IR.34 Inter-Service Provider IP Backbone Guidelines».

[31] 3GPP, «TS 28.062: In-band Tandem Free Operation (TFO) of Speech Codecs».

[32] 3GPP, «TS 23.231: SIP-I based circuit-switched core network».

[33] 3GPP, «TS 29.231: Application of SIP-I protocols to Circuit-switched (CS) core network architecture».

[34] IETF, «RFC 4867: RTP Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs».

[35] IETF, «RFC 3551: RTP Profile for Audio and Video Conferences».

[36] «www.opus-codec.org,» [En línea].

[37] GSMA, «IMS Profile for Voice and SMS».

[38] GSMA, «IR.36 Adaptive Multirate Wide Band, V2.0,» 2013.

[39] H. T. Anssi Ramo, «Voice Quality Characterization of IETF Opus Codec,» Tampere, Finland, 2011.

[40] J. V. K. V. J. S. C. Hoene, «draft-ietf-codec-results-03,» 2013.

[41] ITU-P, «Methods for objective and subjective assessment of quality,» 1996.

[42] i3fourm, «Voice Path Engineering in International IP-based Networks».

[43] 3GPP, «TS 26.201: Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; Frame structure».

[44] i3forum, «Common Functionalities and capabilities of an IPX platform,» 2013.

[45] 3GPP, «TS 26.090: AMR Speech Codec; Speech Transcoding Functions».

4 Market Scenario and HD voice service advantages

As of today the Wideband/HD Voice ecosystem is quite fragmented being characterized by:

a) Multiple codecs

There are high definition codecs defined by international standardization organisms (ITU, IETF..) and there are also proprietary high definition codecs .The diversity of codecs used amongst Service Providers (even between fixed and mobile networks) and OTTs can generate a transcoding challenge which, in principle, can happen anywhere within the end-to-end path.

b) Multiple service markets

It is possible to identify the two markets being involved and interested in the development of wideband/HD Voice services:

- *Retail:* involving fixed and mobile Service Providers and OTT Providers for voice services
- *Business / Enterprises:* involving fixed Service Providers and OTT Providers for voice services as well as other applications (e.g. conference calls)

OTT Providers have been successful in providing their users with on-net (within a single OTT Provider's ecosystem) high definition voice using codecs not initially specified in the recognized international standardization bodies.

In this scenario, two basic forces can be identified as drivers for the take-off of high definition voice services: the graceful migration to IP of the fixed and mobile Service Providers together with the OTT-generated traffic to be terminated onto the Service Providers' networks.

As a result, it is expected that the need to interconnect OTT/Service Providers offering high definition codecs will be of increasing importance in the future. Consequently the policies that a Carrier/IPX Provider has to implement with regards to its role in the media negotiation of HD voice calls become relevant. As an example, Opus codec is endorsed in the current IETF draft for WebRTC codec requirements [6] as the mandatory-to-implement high definition codec (while G.711 is the narrowband mandatory codec): it is a marketing and technical decision of Carriers/IPX Providers whether or not to implement such a capability.

It should be noted that offering high definition voice over a managed network, where voice packets are less susceptible to degradation than in a public environment, gives end users a consistent better experience.

In the medium term, as technologies and standards incorporating high definition codecs gain traction, high definition will eventually become ubiquitous and an expected feature of voice services. The development of VoLTE, WebRTC, the growing number of HD-enabled devices and networks will contribute to make high quality voice ubiquitous.

The usage of wideband/HD Voice has proved the following benefits for the industry:

- *Higher quality user experience*

With Wideband/HD Voice, it is easier to understand specific pronunciations and to recognize certain accents and voices. A better speech quality allows the user a more relaxed phone conversation. The voice and the environmental background noise sound more natural which increases the intelligibility and the effect of presence and closeness.

- *Call duration increase*

Higher in-call quality tends to improve call duration. Low quality degrades in-call user experience to the point where the user will terminate the session prematurely, or will keep it as short as possible

- *Call productivity increase*

Clearer speech conveys subtle differences between some letters and numbers, which can be lost in lower definition session; “m” and “n”, “15” and “50” for example. In general, the greater ease of communicating in a seamless fashion reduces fatigue and increases productivity.

- *Voice Usage increase*

Increased satisfaction can result in an increased usage of the voice service vs. other data services (e.g. SMS, email).

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5 HD codec definition

Traditional TDM fixed lines use the G.711 codec - described in ITU-T “G711: Pulse Code Modulation of voice frequencies” [7]- that was designed for the telephony speech bandwidth range of 300-3400 Hz. As VoIP first developed, most VoIP sessions had to interact at some point with the PSTN; therefore the most common voice codecs for VoIP, such as G.711 and G.729 [8], were developed with TDM interworking in mind and thus with that same frequency range. End-to-end IP voice paths are becoming more common, which has given rise to the development of new codecs for higher speech bandwidth. These codecs are known as wideband codecs. Wideband codecs operate in a wider frequency range, between 50 Hz and 7,000 Hz. Even higher bandwidth codecs (superwideband codecs) have been developed in the frequency range 50-14,000 Hz, and there are also fullband codecs (up to 20 Khz).

Usage of high quality codecs (wideband, super wideband or full band encoded frequency bandwidth) enriches the user experience enabling deep clarity. The voice and the environmental background noise sound more natural, which increases the intelligibility and the effect of presence and closeness.

There are several methods for measuring and evaluating a speech signal, both subjective and objective. The most frequently used quality scale is ITU-T P.800 [9] MOS (Mean Opinion Score): listeners assign a value from 1 (lowest quality) to 5 (highest) to a speech sample. Another scale for subjective evaluation is MUSHRA (from MUltiple Stimuli with Hidden Reference and Anchor, described in ITU-R BS.1534-1 [10]), where the quality scale goes from 0 to 100. The de-facto standard procedure for objective evaluation is PESQ (Perceptual Evaluation of Speech Quality), described in ITU-T P.862 [11], extended in P.862.2 [12] for wideband signals. Additional details of these different methods can be found in section 7 of [4]. PESQ has now been updated by the new ITU-T P.863 [14] standard named "POLQA", which contains the implementation guide that shall be considered as the standard to be used for objective voice quality evaluation for conversational codecs and services.

In the i3Forum document *Voice Path Engineering in International IP-based Networks* [13], the E-model, described in ITU-T G.107 [15] is thoroughly explained. The E-model was developed for narrowband codecs in the band 300-3400 Hz, defines the scalar quantity R (Transmission rating factor) as an objective measure of the quality of a call. It works by subtracting from a nominal factor Ro (basic signal-to-noise ratio) different impairment factors. Details of each of these impairment factors can be found in ITU-T G.113 [16].

ITU-T G.107.1 [17] provides an adapted version of the E-model for a very limited set of wideband codecs. It is still a work in progress because for some parameter combinations (e.g. the effects of delay in conjunction with other impairments), wideband model predictions have been questioned and are currently under study. G.107.1 defines an extended R-factor that resizes the scale of the E-model to reflect the quality improvement when migrating from narrowband to wideband. This extended R-scale is a universal scale in the sense that it can be both applied to narrowband and wideband transmission channels. The new scale goes from 0 to 129. The MOS, as in the narrowband E-model, can be obtained mathematically from R. The E-model does not apply to high quality codecs with bandwidth larger than wideband.

In addition to regular impairment sources (delay, jitter, packet loss …), transcoding is another factor that can impact severely the audio quality of a voice call. Transcoding between narrowband codecs, especially if performed in more than one point in the call path, might lead to an unacceptable quality in an international call.

If transcoding is performed between two different high quality codecs operating in the same bandwidth, since the original sampling can be maintained, the quality is expected to be higher than if the negotiated codec were narrowband end-to-end.

Presently (i.e. 2014) there is no published analysis from a neutral source with a comparison between a call where transcoding wideband-narrowband is performed and a pure end-to-end narrowband call. For the analysis in further sections it is assumed that no quality improvement is obtained by transcoding in this scenario.

It has to be kept in mind that this scenario is very unlikely since it implies that a Service Provider exclusively supports wideband codec(s), something unknown presently (i.e. 2014). The only scenario where this situation is conceivable nowadays is a call where one of the endpoints has to fall back to a narrowband mode for some network related reasons. For instance, two mobile subscribers in different Service Providers negotiate an AMR-WB session under 3G coverage, and in the middle of the call one of them moves to a 2G area where AMR-WB is not supported, thus falling back to a narrowband mode. In that case transcoding between AMR-WB and AMR may occur. However, given that it is technically possible, this scenario is covered in section 9.

6 High Definition codec types

There are different high definition codecs in the market offering wideband or even larger bandwidth. They use different technologies and were conceived for various purposes.

G.722 and AMR-WB are the mandatory-to-implement wideband codecs for a Carrier/IPX Provider. Also, as per [1], a Carrier/IPX Provider must support two narrowband codecs: G.729 and G.711.

Opus has been specified by IETF as the mandatory-to-implement high definition codec for WebRTC as stated in [6]. It evolved for voice from the Silk codec developed by Skype; and for music and higher-than-wideband speech, from CELT, developed by the Xiph.org foundation.

A codec can be described in terms of several attributes. The main ones are the following:

- Encoding/decoding scheme: the algorithm used for coding speech.
- Frequency: the frequency range of the original voice signal that is sampled. This is what determines whether or not a codec is narrowband or wideband/super wideband/fullband.
- Sampling frequency: the rate at which voice samples are taken from the original voice signal.
- Bitrate

In the following section some of the high quality codecs are characterized in terms of these attributes.

6.1 Mandatory-to-implement wideband codecs for a Carrier/IPX Provider

6.1.1 G.722

G.722 is an ITU-T standard ([17]).

Encoding scheme: sub-band Adaptive differential pulse-code modulation (ADPCM)

Frequency: 50-7000 Hz

Sample frequency: 16000 Hz

Bitrate: 48, 56, 64 kbps.

License: Free

Applications: Speech, Voice recording. It delivers wideband quality at the same bandwidth used by G.711. G.722 is specified by ETSI in [19] as the mandatory wideband codec for New Generation DECT (e.g. CAT-iq compliant terminals)

6.1.2 G.722.2 (AMR-WB)

G.722.2 as defined by ITU in [20] is the same codec as AMR-WB, the name 3GPP gives to this codec.

Encoding scheme: Algebraic Code Excited Linear Prediction Coder (ACELP)

Frequency: 50-7000 Hz

Sample frequency: 16000 Hz

Bitrate: Variable 6.60, 8.85, 12.65, 14.25, 15.85, 18.25, 19.85, 23.05 or 23.85 kbps

License: VoiceAge (www.voiceage.com)

Application: Audio, Voice recording, as G.722. Having different modes allows the codec to interoperate with existing GSM and 3GPP wireless systems.

6.2 Other high quality codecs

There are other high quality codecs in use nowadays. Among them:

Opus

Described in IETF RFC 6716 [21]. It has been derived from Skype SILK but it has been substantially modified and they are no longer interoperable.

Encoding scheme: Linear Prediction (Voice) and Modified Discrete Cosine Transform (music, or super wideband/full band speech)

Frequency: 50-20000 Hz

Sample frequency: Variable 8000-48000 Hz

Bitrate: Variable 6-510 kbps

Other features: Support for speech and music, support for mono and stereo, support for up to 255 channels (multistream frames)

Applications: Voice, music

License: BSD (Berkeley Software Distribution)

G.719

Low complexity codec described in [22]. GSMA in IR.39 [23] specifies that a UE should support it for high definition (full-band) video conference.

AAC-(E)LD

Used in Apple Facetime. AAC, a full band codec, is supported in most mobile terminals.

SILK

Developed by Microsoft/Skype.

G.722.1

Partial implementation of codec Siren 7 developed by Polycom.

Speex

Codec from Xiph.Org. Obsoleted by Opus but still in use.

G.729.1

Wideband version of ITU G.729; it has a variable bitrate and it interoperates with narrowband G.729.

iSAC

Used in Google Talk. Originally developed by Global Ip Solutions. It was one of the codecs present in the first WebRTC stack open sourced by Google.

BV32.

Developed by Broadcom. It is the wideband version of BV16, used in PacketCable.

7 Technical reference model

A Carrier can offer high definition services either by its own private/public VoIP platform or via its IPX based platform. It is a Carrier decision whether to pursue one or both alternatives.

The pursuit of a consistent higher quality call experience by using a high definition voice codec should also rely on the use of a high quality network transport layer. The high quality IP transport layer can be achieved with a private managed QoS network or via an interconnected platform network like the IPX. In this document the IPX platform is considered as the reference technical model.

This document is an extension of the i3Forum documents [1] and [2]. A detailed analysis is provided of VoIP interconnections where some of the involved actors support wideband audio codecs, but the reference model subject to analysis will still apply.

In agreement with that reference model, only one type of connectivity option will be considered, namely the Multilateral – Hubbing option.

Figure 1 depicts the different stakeholders under consideration.

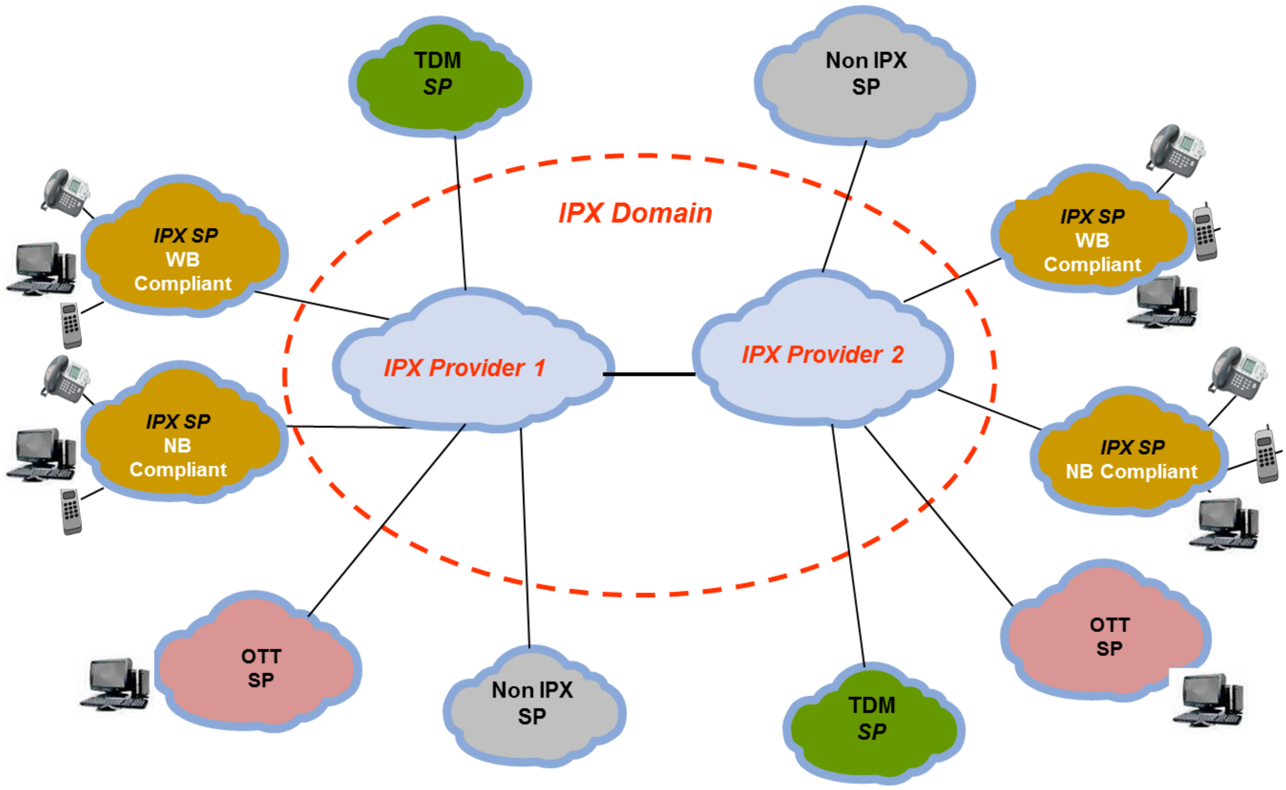

Figure 1: Reference model

A brief description of the elements in the previous figure follows:

- IPX Provider: A VoIP Provider (Carrier/IPX Provider) that conforms to the guidelines provided by [1] and [2].
- IPX SP NB compliant: a Service Provider supporting G.729 or G.711 (or both).
- IPX SP WB compliant: a Service Provider supporting G722 or G.722.2 (or both)

- SP TDM. Service Provider connecting its own legacy PSTN network to an IPX domain.
- OTT SP. Over-The-Top Service Provider.

All these types of Service Providers may have VoIP interconnections across an IPX domain. In the following sections the different cases will be considered. The analysis will focus on the media part of a VoIP call; however, in order to apply its media policy, a Carrier/IPX Provider has to implement some signaling procedures that are the subject of the following section.

8 Signalling

The media negotiation in a SIP session relies on well established procedures. These procedures, known as the offer-answer model, are described in RFC 3264 [24]. This specification provides the mechanisms for establishing, accepting, or rejecting a media session.

This section describes the SIP signaling procedures involved in the cases when the Carrier/IPX Provider has to decide whether or not to take care of the transcoding of the call.

In the most frequent case, the entity that initiates the signaling session also creates the initial media offer; that is, 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. However, other combinations are possible: for instance, the initial INVITE may lack the SDP body; in that case the entity receiving the call is mandated by RFC 3264 to send the offer within the 200 OK message. The answer is then present in the ACK sent from the originating side.

SIP entities can also implement the following mechanisms for setting up a media session or to modify an existing one.

- *Offer in reliable response.* The initial INVITE has no SDP body, the UAS can include the media offer in a provisional response sent in a reliable manner (including a Require: “100rel” header and a “RSeq” header). The answer will be contained in the SDP body of the PRACK request.
- *SIP UPDATE Method.* If the initial INVITE contained an SDP offer and the UAS sent a reliable provisional response with the answer (e.g., a 183 response), the UAC may modify the initial offer and include the modified SDP body in an UPDATE request. The SIP UPDATE can also be used by the UAS after the call has been answered, but this is not common.
- *SIP Re-INVITE*. Both UAC and UAS can modify the parameters of a confirmed ─answered─ media session by sending an INVITE message with a new offer. The new offer contained in a Re-INVITE can be completely different than the initial one. For instance, given a narrowband session, one of the Service Providers might try to upgrade the session to a wideband codec by sending a Re-INVITE to its intermediate Carrier/IPX Provider. This could be the situation when a MNO subscriber moves to an area where AMR-WB is available. Re-INVITE messages can also be used as a keepalive mechanism, in which case the media session remains unaltered. Also, a Re-INVITE can contain a new media type not present in the original offer (e.g., a RCS subscriber wanting to add a video stream to the voice session).

For clarity, in the next section only the case when the offer-answer is carried respectively in the INVITE-OK messages will be considered.

8.1 Signalling modes[2]

In the multilateral hubbing model under consideration in this document, a Carrier/IPX Provider (or a chain of at most two Carriers/IPX Providers) is in charge of relaying the SIP signaling between two Service Providers. As far as the media negotiation is concerned, the application layer of the Carrier/IPX Provider proxy function should have a clear policy for every type of scenario. Assuming that the Carrier/IPX Provider supports

2 in the next sections the high quality codecs with bandwidth larger than wideband may be handled as wideband codecs transcoding, it may behave in two different modes: proactive and reactive. These two modes are described in the following sections.

8.1.1 Proactive mode

In proactive mode, a Carrier/IPX Provider modifies the offer contained in the initial INVITE before delivering it to the terminating Service Provider or to the next Carrier/IPX Provider. Let's call C1 the set of codecs present in the offer contained in the initial INVITE, and C2 the set of codecs added by the Carrier/IPX Provider. By definition, sets C1 and C2 share no common codec. C2 can contain one or more codecs, either narrowband or wideband[3]. The priority assigned to each codec in the new offer should be based on the general rules outlined in section 9. If C1 contains one of the mandatory wideband codecs, a Carrier/IPX Provider should place in C2 the other mandatory wideband codec.

If the call is successful, the answer in the 200 OK message may contain:

- One or more of the codecs belonging to C1. In this case it is as though the set C2 had not been added to the SDP; the terminating SP (or another intermediate IPX Provider) prefers to establish the session using the codecs in C1; therefore no transcoding will be necessary and the Carrier/IPX Provider can relay the answer unaltered.
- One or more codecs from C1, and one or more codecs from C2. In that case, and following the guidelines outlined in section 9.1, the behavior of the Carrier/IPX Provider will depend on the nature of the codecs present in C1 and C2, together with the codecs present in the response:
    1. C1 has wideband codecs, and so does the answer: this is the optimal scenario; a wideband voice call with no transcoding will be established. The answer can be relayed upstream unaltered.
    2. C1 includes one mandatory wideband codec; the answer also contains a wideband codec, but the one present in C2, not the one in C1. An end-to-end wideband call is possible if the Carrier/IPX Provider transcodes the session. In that case the answer to the originating side should only contain the wideband codec present in C1.
    3. C1 has no wideband codecs. A common narrowband codec exists between endpoints, therefore transcoding is not necessary. The answer to the originating side should only contain the narrowband codec(s) in the answer that are also present in C1.
- One or mode codecs from C2. No common codec has been found for the UAC and the UAS; therefore transcoding will be needed for a successful call establishment. Provided that the Carrier/IPX Provider had decided to transcode the call, it will strip the set C2 from the 200 OK message and send it to the UAC with an answer that includes one or more of the codecs from C1. If wideband codecs are available in both legs of the call, that's the combination the Carrier/IPX Provider should honor for an optimal quality. In that case, the 200 OK message sent to the originating side should only contain the wideband codec(s) present in C1.

If the Carrier/IPX Provider is directly connected to the terminating SP, the proactive mode may be useful if the Carrier/IPX Provider knows in advance that the SP will be unable to complete the call for the media offer received, based on an agreed codec list. Extending the offer with the set of codecs known to be accepted by the SP may lead to a faster call setup.

On the contrary, If the Carrier/IPX Provider is directly connected to the originating SP and the call is delivered to a second Carrier/IPX Provider, proactive mode is likely to be avoided because the media

3 The Carrier/IPX Provider under consideration is not mandated to support higher-than-wideband codecs capabilities of the terminating SP are unknown. However, specific agreements enforcing this mode of operation may exist between the originating SP and the Carrier/IPX Provider.

8.1.2 Reactive mode

In reactive mode, the Carrier/IPX Provider does not alter the media attributes of the initial offer coming from the initiating SP or from an intermediate Carrier/IPX Provider when sending it downstream. It may decide to react to an unsuccessful media negotiation by sending downstream a subsequent INVITE with a supplementary offer. If this second offer is successful, a transcoding session is then established. The upstream elements in this session are unaware of this second INVITE. Business and technical factors will determine in each case the reactive policy to be applied in the scenarios that will be addressed in section 9.

8.2 SIP Procedures in a transcoding scenario

This section describes the signalling procedures involved in a transcoding scenario.

8.2.1 Procedures in proactive mode

The policy implemented in the Carrier/IPX Provider network may force it to perform transcoding in proactive mode for a specific traffic pattern. For instance, if the terminating Service Provider is known by the Carrier/IPX Provider to support only AMR-WB, but the incoming INVITE contains exclusively narrowband codecs, proactive mode can enable a faster call set up. The Carrier/IPX Provider will alter the offer in the initial INVITE by adding the codec (or set of codecs) known to be accepted by the terminating SP. It may also completely replace the original offer and create a new one containing only the set of codecs supported by the SP. After the call completes, RTP will flow using one codec in the A-Leg and a different one in B-Leg with a transcoding device present in the media chain.

Even in proactive mode, for some reason the terminating SP may still reject the media offer. The Carrier/IPX Provider will consider a media session offer as rejected in three cases:

1. The INVITE request is rejected with a SIP message with code 488 “Not Acceptable here”.
2. The INVITE request is rejected with a SIP message (or SIP-I with missing ISUP MIME) with a code different than 488, but containing a Reason header with this format: **Reason: Q.850 ;cause=65 ;text=”*<text>*"**
3. Assuming a SIP-I INVITE, the request is rejected with a message that includes an ISUP body encapsulating a REL message with a Release Cause 65.

In this case, the Carrier/IPX Provider most likely will consider the SP unable to terminate the current session. According to its routing policy it may attempt to deliver the call to a backup SP, if applicable, or to relay the rejection message upstream.

8.2.2 Procedures in reactive mode

In reactive mode, the Carrier/IPX Provider will relay downstream the media offer it received unaltered. This statement refers to the codec attributes; other parameters (c= line if it wants to remain in the media path; the s= line, etc.) can be modified.

This is consistent with the guidelines outlined in section 9.1.

In reactive mode, the terminating Service Provider (or a transit Carrier/IPX Provider) may reject the media session. Once a Carrier/IPX Provider receives a message indicating a rejection due to codecs not matching (code 488, or cause=65 as explained in the previous section), it will decide how to react:

- If it is directly connected to the originating SP, and uses a second Carrier/IPX Provider for termination, the action upon rejection of the call has to be dictated by the agreement between the Carrier/IPX Provider and this SP. If no specific agreement exists, the Carrier/IPX Provider will apply the aforementioned rule from *[2]*: "*in the first instance it is the responsibility of Service Providers to support transcoding "*, that is, it will relay the rejection message to the SP, who can decide whether or not to provide a new INVITE with an expanded list of codecs and then to transcode the call. Otherwise, the Carrier/IPX Provider may itself build a new offer with a different set of codecs. If the Carrier/IPX Provider ignores the set of codecs accepted by the terminating SP, it should include in the new offer all the codecs (narrowband and wideband, the latter with higher priority) that it supports.

- If it is directly connected to the terminating SP, the action upon rejection of the call has to be dictated by the agreement between this Carrier/IPX Provider and the Service Provider. According to the nature of this agreement, the Carrier/IPX Provider may decide to send a new offer and transcode the call; especially if the initial offer contains wideband codec(s). Otherwise it will relay the rejection message upstream.

8.3 Signalling requirements

The ability to transcode between different codecs has some implications in the signalling part of a VoIP Session. A wideband codec compliant Carrier/IPX Provider should support some signalling features that may alter its default media policy.

8.3.1 Transcoder Free Operation (TrFO)

The TrFO mechanism is described in 3GPP specification TS 23.153 [25] for MNOs. A MNO initiating a session to another MNO traversing an IPX domain (or a chain of Carriers compliant with [1]), might want to use the TrFO procedure in order to try to set up a transcoder free operation. The TrFO definition provided by the specification is: *"configuration of a speech or multimedia call for which no transcoder device is physically present in the communication path and hence no control or conversion or other functions can be associated with it."*

Using SIP-I as transit protocol, TrFO slightly changes the semantics of the offer-answer model from RFC 3264 [24], although it is still compliant with its rules. The main idea behind TrFO is to try to establish an end-to-end session with no transcoders in the media path and where endpoints are free to negotiate each other a change of codec during the call itself. For instance, an AMR-WB session could be set up between mobile stations located in different MNO's connected through an IPX domain. The specification also describes the procedures for a break-out and break-in of the TrFO. An example of a break-out is when a mobile station in the middle of a TrFO session using AMR-WB moves into a new cell whose radio capabilities only allow regular AMR.

According to [25], wideband codecs, if available, should be given preference in the codec negotiation.

When a TrFO session establishment or modification is taking place between two MNOs, the Carrier/IPX Provider should leave the media negotiation in the hands of the Service Providers. That requires it to be able to recognize that a SIP session establishment is using the TrFO procedure. The TrFO sessions are marked in the SDP body with a distinctive attribute **a=3gOoBTC**. Whenever this attribute is present in a SIP message ─either in an offer or in an answer─, a Carrier/IPX Provider should not intervene in the media negotiation stage, unless the downstream SP is known to be unable to support the TrFO procedure.

If the Service Providers are not able to establish a TrFO operation, the Carrier/IPX Provider is free to apply its general policy.

8.3.2 SIP Preconditions

There is another situation when a Carrier/IPX Provider should not interfere on the media negotiation stage, namely, when the endpoints use the SIP Precondition framework, as defined in RFC 4032 [26]. This RFC provides a mechanism that prevents SPs from alerting the final user unless certain conditions are met. Presently (i.e. 2014), three types of preconditions are defined:

- Connectivity preconditions (IETF RFC 5898 [27]). The terminating user will not be alerted unless end-to-end connectivity has been previously ensured.
- QoS preconditions (IETF RFC 3312 [28]). The terminating user will not be alerted unless QoS parameters can be guaranteed.
- Security preconditions (IETF RFC 5027 [29]). The terminating user will not be alerted unless a secure communication between endpoints can be guaranteed.

The SIP Preconditions framework is implemented by adding new attributes to the SDP. This is a simple example of a SDP body inside of an initial offer:

```
m=audio 20000 RTP/AVP 0
a=curr:qos e2e send
a=des:qos optional e2e send
a=des:qos mandatory e2e recv
m=audio 20002 RTP/AVP 0
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos optional local sendrecv
a=des:qos mandatory remote sendrecv
```

The attribute `curr` indicates the current situation; and `des` the desired one. This particular case mandates the terminating Service Provider to reserve the media resources before alerting the final user. If the resource reservation fails, the offer must be rejected.

SIP preconditions is an end-to-end mechanism. The Carrier/IPX Provider should be able to recognize that preconditions are used in a session establishment. In that case, the policy that it would otherwise (i.e. with no preconditions) apply for that particular offer should be bypassed and, as in the TrFO case, the media negotiation should be left in the hands of the Service Providers.

If the preconditions are met and the call progresses, the Service Providers may want to re-negotiate the media parameters of the call. For instance, they can use the SIP UPDATE method to renegotiate the codecs even before the call is connected. At this point, the Carrier/IPX Provider is free to apply its general policy and act as in a regular session.

8.3.3 AMR-WB signalling

AMR-WB is, together with G.722, one of the two mandatory-to-implement codecs for a Carrier/IPX Provider that supports HD voice, or for a Carrier compliant with [1]. AMR and AMR-WB were originally designed for circuit-switched mobile radio systems. However they are also suitable for packet-switched (IP) networks. AMR-WB is a very flexible codec and can be set up in variety of configurations by the proper tuning of several parameters. Some of these parameters are:

- octet-align: determines if octet-aligned or bandwidth-efficient transmission is used.
- mode-set: list of modes supported
- channels: number of audio channels.

All 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. Therefore, as far as the signaling is concerned, a Carrier/IPX Provider must be compliant with this specification in order to fully support transcoding of this codec.

This is an example of an SDP body containing an offer of two different flavors of the AMR-WB codec.

```
m=audio 49120 RTP/AVP 98 99
a=rtpmap:98 AMR-WB/16000
a=fmtp:98 octet-align=1
a=fmtp:98 interleaving=30
a=rtpmap:99 AMR-WB/16000/2
a=fmtp:99 octet-align=1
a=maxptime:100
```

9 Call Scenarios

The reference model under consideration and described in section 7 allows several combinations of VoIP interconnections, resulting in different call scenarios. Some of them are very unlikely to occur presently, others are the general case and others are still rare but they are expected to become more frequent.

VoIP sessions using wideband voice codecs will increase as described in section 4; the challenge is to maintain this quality in sessions between different Service Providers across a Carrier/IPX domain.

The following sections describe these different scenarios. Recommendations and guidelines are offered to the Carriers/IPX Providers so that, given the media capabilities of the originating and terminating endpoints in a VoIP call, the optimal quality available - when feasible - can be attained.

9.1 Rules and guidelines

Ensuring end-to-end QoS should be one of the main objectives for a Carrier/IPX Provider participating in a VoIP session. In general, the policy to be applied in all cases will be based on the following guidelines[4]:

1. **Transcoding should be avoided when it impairs speech quality**.
2. **Wideband codec continuity with no transcoding offers the optimal quality scenario.**
3. **Transcoding to narrowband codecs must be avoided unless it is the only way for a call to be successfully established.**
4. **A call, where transcoding between two different wideband codecs takes place, has better quality than the same call using a unique narrowband codec end-to-end, as stated in GSMA docs [3] and [4]. No significant quality improvements are expected if a call, in some segments, is converted to wideband versus an end-to-end narrowband quality[5].**
5. **If both narrowband and wideband codecs are offered in a VoIP session, the wideband ones should be placed in the top priority positions in the SDP offer.**
6. **The order of codec/packetization period preference is determined by the originating terminal and should be honored wherever possible;**
7. **In the first instance it is the responsibility of Service Providers to support transcoding in order to ensure successful voice interoperability for their services. Transcoding likelihood decreases if the originating Service Provider offers a wide range of codecs.**

4 With respect to this section, high quality codecs with bandwidth larger than wideband (super wideband or full band) may be handled as wideband codecs. The following guidelines may be applicable to high quality codecs offering more than wideband (superwideband or fullband) according to the quality resulting from the transcoding scenarios considered between different high quality codecs of different sampling frequencies.

5 In a call where transcoding WB-NB takes place, one stream will be converted from narrowband to wideband and vice versa. Assuming a fixed bitrate, conversion from narrowband to wideband implies allocating fewer bits to the narrowband part of the codec; therefore an end-to-end narrowband call is expected to have better quality.

9.2 Scenarios where no TrFO is enabled

In the analysis that follows, the session type under consideration has these features:

- 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 Re-INVITE's 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 Carrier/IPX Provider has transcoding capabilities for the mandatory-to-implement wideband codecs.

9.2.1 Session initiated by a SP offering ONLY mandatory wideband/HD codecs (G.722/AMR-WB)

In this case, the initiating Service Provider ONLY includes in the offer one or both of the IPX mandatory wideband codecs: G.722 and AMR-WB.

Scenario likelihood: no case known presently.

At the time of writing (i.e. 2014) it can happen that within IPX-compliant Service Providers, the offer only contains mandatory wideband codecs. However, when the call has to traverse an international Carrier/IPX domain, the SP normally adds to the offer at least a narrowband codec, namely G.711. In that case the optimal condition (i.e. wideband continuity) is achieved by placing the narrowband codecs in the bottommost SDP positions when sending the INVITE downstream.

Anyway, since it is conceivable that eventually some Service Provider will offer only wideband codecs to its subscribers, it is worth considering the different scenarios that can arise depending on the terminating SP capabilities.

9.2.1.1 The terminating SP accepts one or both of the wideband/HD voice codecs present in the offer.

This is the optimal case, because there is end-to-end HD voice continuity. The media session will be established using the common codec(s) among the Service Providers. *No intervention in the media negotiation is expected from the Carriers/IPX Provider(s) involved in the call.*

9.2.1.2 The terminating SP only accepts narrowband codecs.

There is no common codec between Service Providers; therefore, for the call to succeed, transcoding will be required at some point. Applying general rule 7, Service Providers are the entities responsible for transcoding.

Also, as per general rule 4 in section 9.1, no quality improvement is expected if a wideband codec is only used towards the initiating SP versus a full end-to-end narrowband call. If no transcoding is applied, the session will be rejected by the terminating SP. The initiating SP, on reception of the rejection message, may decide to retry the call offering a new set of codecs.

There can be *exceptions to this guideline*:

1. The Carrier/IPX Provider connected to the originating SP, and the originating SP itself may have an agreement whereby the Carrier/IPX Provider takes care of the transcoding if necessary. In that case, the Carrier/IPX Provider may work in proactive mode by adding more codecs (narrowband and wideband) to the initial offer; or in reactive mode when a media rejection (SIP 488 or Cause=65) is captured.

2. The Carrier/IPX Provider connected to the terminating SP, and the terminating SP itself may have an agreement whereby the Carrier/IPX Provider takes care of the transcoding if necessary. In that case, the Carrier/IPX Provider may work in proactive mode by adding more codecs (narrowband and wideband) to the initial offer; or in reactive mode when a media rejection (SIP 488 or Cause=65) is captured. Working in proactive mode results in a faster call setup.

9.2.1.3 The terminating SP supports one or more wideband codecs, but different from the ones contained in the offer

As in section 9.2.1.2, there is no common codec supported by the originating and terminating Service Providers. However there is a difference. As stated in rule 4 from section 9.1, a transcoding session between different wideband codecs is supposed to render better quality than a call where there is end-to-end continuity of a narrowband codec. For that reason the Carriers/IPX Providers may act in a slightly different way depending on their place in the media negotiation chain:

- *The Carrier/IPX Provider connected to the originating Service Provider should not react upon a 488 / cause 65 rejection coming from another Carrier/IPX Provider. If this Carrier/IPX Provider is willing to transcode the call, it should act in proactive mode, which will lead to a faster call setup*. Still, specific implementations or business agreements may lead it to act in reactive mode after receiving a SIP Status-Code 488 or cause 65.

- *The Carrier/IPX Provider connected to the terminating Service Provider, provided that it is aware of the media capabilities of the terminating SP, can, in proactive mode, modify the initial offer by adding to it the wideband codec known to be supported by the SP*. If not, it may react or not to a SIP 488 / cause 65 rejection based on business rules.

Therefore, only a specific business agreement between the terminating SP and its connected Carrier/IPX Provider, may force the Carrier/IPX Provider to take care of the transcoding. If that is the case and to possibly ensure wideband transcoding, regardless whether of the Carrier/IPX Provider acting in proactive or reactive mode, the wideband codec for which transcoding is supported should be included in the new offer as the top priority codec.

The following table summarizes the content of this case:

Codecs supported by the terminating SP	End-to-end wideband possible	Carrier/IPX Provider “optimal” action	Business aspects
IPX mandatory Wideband codec(s)	Yes	• None if a common codec exists • Transcoding WB-to-WB (proactive mode enables faster setup)	Transcoding requires additional resources
Narrowband codecs	No	Not applicable	
Other wideband codecs	Yes	Transcoding	Transcoding requires additional resources

Table 1: Originating SP only offers one or both mandatory WB/HD Voice codec

9.2.2 Session initiated by a SP offering mandatory wideband/HD Voice codecs and other codecs.

In this case, the Service Provider includes in the offer G.722 and/or AMR-WB among other codecs, either narrowband or wideband.

Scenario likelihood: Low-medium, expected to become the default scenario.

As the HD voice deployment spreads, wideband codecs are likely to be offered even when interworking is necessary. Normally they will be present in the media offer along with globally supported narrowband codecs such as G.711.

In this case, transcoding is not expected in general, because the more codecs that are present in the offer, the more likely it is that the terminating Service Provider will accept one (or several) of them.

The general rule 7 should be applied by default, leaving to the Service Providers the media negotiation stage. However, specific agreements between the Carriers/IPX Providers and the Service Providers might induce the Carrier/IPX Provider to perform transcoding. One example scenario: an initial offer contains both G.722 and G.711. The terminating SP supports AMR-WB and G.711. Applying the default rule 7, that is, leaving the media negotiation to the Service Providers, would result in an end-to-end non-wideband G.711 call. *A wideband capable Carrier/IPX Provider should try to know the media capabilities of the Service Providers connected to it, so enabling wideband voice.* That way, it may in this case ─or analogous ones─ change its default behavior and participate in the media negotiation by trying to establish a wideband path for the call (even if the codecs in the two legs are different).

The following table summarizes the content of this case:

Codecs supported by the terminating SP	End-to-end wideband attainable	Carrier/IPX Provider “optimal” action	Business aspects
IPX mandatory Wideband codec(s)	Yes	• None if a common codec exists • Transcoding WB-to-WB	Transcoding requires additional resources
Narrowband codecs	No	Not applicable	
Other wideband codecs	Yes	Transcoding	None of the codecs supported by the terminating SP is mandatory for the Carrier/IPX Provider

Table 2: Originating SP offer WB codecs

9.2.3 Session initiated by a SP offering only Opus

In this case, the Service Provider includes only Opus in the initial offer, the high quality codec most likely to become mainstream for OTT providers, mostly due to its mandatory-to-implement role in WebRTC.

Scenario likelihood: unknown.

It is common among OTT voice providers to use high quality codecs for communication between two or more of their subscribers. Still, for calls towards other providers, OTTs include in their offer other codecs, G.711 being the most common.

It has to be taken into account that Opus is not a mandatory-to-implement codec for a Carrier/IPX Provider. Therefore, for a Carrier/IPX Provider the only recommendation to apply is the general rule of leaving the media negotiation in the hands of the Service Providers. However, the growth potential of Service Providers offering VoIP on Opus, makes the case study worthwhile technically and commercially.

9.2.3.1 The terminating SP accepts Opus

This is the optimal scenario: and end-to-end up to full band communication will be established between Service Providers, using Opus codec.

9.2.3.2 The terminating SP accepts only AMR-WB or G.722, not Opus

Opus operates in various modes, including narrowband (frequency sampling = 8 Khz), wideband (fs = 16 Khz), superwideband (fs = 32 Khz) and fullband (fs = 48 Khz), and with a variable bit rate (VBR). In WB SILK-mode (i.e. for voice communications), it behaves as a wideband codec.

A VoIP call where a wideband sampling frequency is kept along the full media path, even if transcoding takes place, has better quality than an end-to-end G.711 session. Thus, a transcoding between Opus (in SILK Mode) and G.722 or AMR-WB is desirable even if, as said earlier, Opus is not a mandatory codec for a Carrier/IPX Provider. There is no known analysis of the quality of calls where transcoding between Opus in other modes (e.g. fullband) and AMR-WB or G.722 takes place.

An increase in the number of Service Providers supporting Opus is expected. This may lead to business cases where transcoding between Opus (in wideband mode) and other wideband codecs, mostly the IPX mandatory wideband codecs, will be of interest for the Carriers/IPX Providers.

9.2.3.3 The terminating SP accepts other codecs

In this case the Carrier/IPX provider should not participate in the codec negotiation, because falling back to a narrowband codec will eliminate any quality improvement resulting from having Opus in one of the legs.

Codecs supported by the terminating SP	End-to-end wideband attainable	Carrier/IPX Provider "optimal" action	Business aspects
IPX mandatory Wideband codec(s)	Yes	Transcoding when Opus is in SILK-wideband mode	Opus is not mandatory, thus it's a Carrier/IPX Provider decision whether to support it or not
Narrowband codecs	No	not applicable	
Opus	Yes	None	
Other high quality codecs	Yes	Transcoding when Opus and the codecs supported by the terminating SP operate in the same frequency range	Opus and the codecs supported by the terminating SP are not mandatory for the Carrier/IPX Provider

Table 3: Originating SP only offers Opus

9.2.4 Other scenarios

In this section the rest of the possible cases are considered.

9.2.4.1 The initiating SP ONLY offers non-mandatory high quality codecs

For this case, the guidelines outlined in section 9.2.3 remain valid; Opus codec has been dealt with separately for its growth potential. Other codecs (G.722.1 G.729.1, AAC-(E)LD, G.719, etc.) are deployed in other Service Providers, and therefore transcoding between them and other wideband codecs may be worthwhile in some cases.

Codecs supported by the terminating SP	End-to-end wideband attainable	Carrier/IPX Provider "optimal" action	Business aspects
IPX mandatory Wideband codec(s)	Yes	Transcoding	None of the codecs supported by the originating SP is mandatory for the Carrier/IPX Provider
Narrowband codecs	No	not applicable	
Other high quality codecs	Yes	• None if a common codec exists • Transcoding (proactive mode enables faster setup)	None of the codecs supported by the originating and terminating SP is mandatory for the Carrier/IPX Provider

Table 4: Originating SP offers other WB codecs

9.2.4.2 The initiating SP ONLY offers non-mandatory narrowband codecs

In this case there is no way for the Carrier/IPX Provider to setup an end-to-end HD voice call. Therefore, this case is out of scope of this document.

10 Routing

In a strict narrowband environment, most of the Service Providers offering voice over IP services via international interconnections are able to exchange media packets (RTP) with no intermediate device in the communication path. G.711 - in its two variants - and G.729, are universally deployed codecs, and therefore generally speaking transcoding is performed only in special cases.

For these reasons, a Carrier/IPX Provider rarely takes into account the media capabilities of the Service Providers for the configuration of its routing policy. Other factors like delay, packet loss or jitter have a bigger influence in the QoS related KPIs, especially with reference to the ALOC (Average Length of Call). Therefore, for a Carrier/IPX Provider, the routing system implements a dynamic process whose specific inputs are Carrier/IPX Provider dependant, but the basic parameters can be bundled as follows:

- Origin and Destination of the traffic (calling/called numbers)
- Rates (offered to the clients and received from the providers)
- KPIs (ASR, ALOC, …)
- Technical factors (e.g. network congestion)
- Cost of transport towards a destination, mostly in satellite cases.
- Nature of the traffic (speech, 64k channel(s), etc.)

Wideband/HD Voice capabilities should be included in the last item; and it can give rise to new routing criteria.

Considering that a Carrier/IPX Provider has no practical way of systematically gaining advanced knowledge of the called party wideband/HD Voice capability (lack of end-point capability discovery) and the need to support end-to-end of wideband/HD Voice codecs, these requirements call for three basic routing principles:

i. the wideband/HD call needs to preferably be routed directly to the terminating network;
ii. the wideband/HD call needs to be routed to a wideband/HD Voice capable downstream network;
iii. transcoding might take place between wideband codecs.

As a result, a Carrier/IPX Provider has to be aware, if possible, of the wideband/HD Voice capabilities both of the terminating Service Providers and the other Carriers/IPX Providers involved in the end-to-end path and its routing system should be able to perform (route) accordingly.

In this respect Number Portability resolution gains relevance together with all techniques that allow a Carrier/IPX Provider to keep track of the network which owns a specific end-user customer. This knowledge may allow the Carrier/IPX Provider to make routing decisions taking into account the likelihood of the terminating SP being HD capable.

From a business perspective, in comparison with a “traditional” standard quality environment, the scenario described above (i.e. update of the routing system/logic; increased number portability awareness) generates additional costs for Carriers/IPX Providers which have to implement optimal solutions in order guarantee service profitability.

11 Recommendations and Business Model for HD Voice continuity

11.1 Recommendations

As far as HD Voice continuity is concerned, the following rules can be derived as a summary of the different cases analyzed in the previous sections.

- Transcoding should generally be avoided; Service Providers should offer a fallback narrowband codec supported universally (e.g. G711) along with its supported high quality codec(s).
- A standardized method for achieving HD Voice continuity is the implementation of Transcoder Free Operation (TrFO) as specified by 3GPP.
- If transcoding is the only way for a session to be established, Service Providers are the preferred entities to perform it.
- There are cases where through transcoding, a Carrier/IPX Provider is able to assure wideband continuity in a session that otherwise could only be setup with narrowband codec by the service provider.
- If transcoding is to be performed by a Carrier/IPX Provider, then:
  - Call setup time is minimized when the Carrier/IPX Provider acts in proactive mode as described in Section 8 and 9 above.
  - If the call is rejected by the terminating SP, the fastest call setup takes place when transcoding is performed by the Carrier/IPX Provider connected to the terminating SP.

11.2 Business model

As far as the business model of a HD voice service is concerned; the “traditional” business model based on Calling Party Pays with cascading of the termination price among all involved Service Providers and Carriers/IPX Providers does not have to change. Similarly, the pricing scheme based on call duration can be retained.

It is a Carrier/IPX Provider decision to pass on the incremental costs of HD voice either by creating a new product at a premium price for HD voice calls or by absorbing the costs into existing products relying on higher traffic volumes.