This document provides the i3 forum’s perspective on IMS services development, focusing on the role of and the impact on the International Carriers. It does not intend to duplicate other existing specifications or documents on the same issue, but to complement these documents with the perspective of the International Carrier members of i3 forum.
EXECUTIVE SUMMARY

The rise of LTE technology in mobile networks together with the increasing FTTx deployment in the access section of fixed networks have been pushing a strong interest for IMS based services at the international level. These technological developments are paired at the service level by the widespread growth of LTE data services and by the first deployments in Asia and in the USA of Voice over LTE (VoLTE) services with HD voice capabilities.

In the wake of this trend, i3 forum has considered it a priority to deliver a document devoted to describing the current strategic environment, the architectures, the interfaces, the protocols and the related business models to be adopted for the support of International IMS services between two IMS Service Providers or between an IMS Service Provider and non IMS Service Provider adopting, in line with previous deliverables, an IPX model at the transport level.

Among the wide set of IMS-based services, in this first release, the focus is given to Voice over IMS (encompassing both Voice over IP originated by a fixed network and Voice over LTE) covering both the basic international call and the roaming cases.

As a result, the document addresses:
1) a description of the basic technologies (e.g. LTE) which support the evolution towards IMS together with a reference to the OTT services;
2) the state of the art of emerging technologies/services such as HD Voice, LTE, IMS/ VoLTE in terms of current and projected adoption;
3) a discussion of the Voice over IMS business model, reconfirming the existing Sending-Party- Pays business model with a charging scheme based on call duration/destination;
4) the architectural framework based on IPX, outlining the role of the International Carrier / IPX Provider together with the adopted signalling protocols (specifically the 3GPP standard TS 29.165 [1]) and codecs (e.g.G711, G.729, WB-AMR);
5) the basic principles for call routing and user addressing, quality of service control and monitoring;
6) an analysis of the impacts on Carriers’ / IPX Providers’ networks in terms of call types to be supported, physical interconnection, signalling interworking, transcoding and call routing for 4 major interworking scenarios:
   - Case A) from IMS to IMS (with no fixed/mobile interworking);
   - Case B) from IMS to IMS (with fixed/mobile interworking);
   - Case C) from legacy networks to IMS and vice versa;
   - Case D) from IMS to VoIP and vice versa.
7) the discussion of the business and technical impacts of the roaming scenarios recently approved by GSMA.

The ultimate objective of the document is to provide a unique analysis of the impact on Carriers’ / IPX Providers’ platforms of the provisioning of IMS based services, giving priority to the Voice over IMS service. The focus is given not only to the selection of the proper standard(s) to be adopted within a comprehensive IPX architectural and commercial model, but also to the discussion of the various alternatives to be dealt with and their related results with respect to the end-to-end service.
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1. **Scope and objective of the document**

Over the last two to three years, the rise of LTE technology in mobile networks together with the increasing FTTx deployment in the access section of fixed networks have been pushing a strong interest for IMS-based services at the international level.

The mentioned technological development is matched at the service level by the wide-spread growth of LTE data services and by the first deployments in Asia and in the USA of Voice over LTE (VoLTE) services with HD voice capabilities.

In the wake of this trend, i3 forum has considered it a priority to deliver a document devoted to describing the current strategic environment, the architectures, the interfaces, the protocols and the related business models to be adopted for the support of International IMS services between two IMS Service Providers or between an IMS Service Provider and non IMS Service Provider adopting, in line with previous deliverables, an IPX model at the transport level.

Among the wide set of IMS-based services, in this first release the focus is given to voice over IMS (encompassing with this terminology both voice over IP originated by a fixed network and voice over LTE), leaving to next releases the coverage of videocall / video over LTE (ViLTE) and Rich Communication Suite (RCS). In terms of call type, both the basic international call and the roaming cases are dealt with.

As a result, the scope of this document is the following:

1) to provide a description in Sec. 4 of the basic technologies (e.g. LTE) which support the evolution towards IMS together with a reference to the OTT’s services;

2) to describe the state of the art of emerging technologies/services (Sec. 4) in order to provide a solid commercial background to the discussion of the related business model (Sec. 6);

3) to discuss the architectural framework based on IPX (Sec. 5), together with the adopted signalling protocols (Sec. 7) and codecs (Sec. 8) based on the recognized international specifications from 3GPP and GSMA (a specific section is devoted to a comparison with IR.95 “SIP-SDP Inter-IMS NNI Profile”);

4) to discuss the basic principles for Routing and Addressing (Sec. 9), and quality of service control and monitoring (Sec. 11), as well as to provide an analysis of the impacts on Carriers’ / IPX Providers’ networks for 4 major interworking scenarios (Sec. 12) in terms of call types to be supported, physical interconnection, signalling interworking, transcoding and call routing;

5) to discuss the business and technical impacts of the roaming scenarios recently approved by GSMA.

*The final objective of the document is to provide a unique analysis of the impact on Carriers’ / IPX Providers’ platforms of the provisioning of IMS-based services, giving priority to the voice over IMS service. The focus is given not only to the selection of the proper standard(s) to be adopted within a comprehensive IPX architectural and commercial model, but also to the discussion of the various alternatives to be faced and their related results with respect to the end-to-end service.*

In this document, though the interconnection between two IMS-based Service Providers can always be provided by a generic International Carrier, since IPX is the recommended model by i3 forum and GSMA for supporting such interconnection, from Sec. 5 onwards, the terminology IPX Provider is always used for identifying an International Carrier.
## 2. Symbols and Acronyms

<table>
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<tr>
<th>Acronym</th>
<th>Description</th>
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<tbody>
<tr>
<td>3GPP</td>
<td>3rd Generation Partnership Project</td>
</tr>
<tr>
<td>ACELP</td>
<td>Algebraic Code Excited Linear Prediction Code</td>
</tr>
<tr>
<td>ALOC</td>
<td>Average Length of Call</td>
</tr>
<tr>
<td>AMR-NB</td>
<td>Adaptive Multi-Rate Narrow Band</td>
</tr>
<tr>
<td>AMR-WB</td>
<td>Adaptive Multi-Rate Wide Band</td>
</tr>
<tr>
<td>ASR</td>
<td>Answer Seizure Ratio</td>
</tr>
<tr>
<td>HD</td>
<td>High Definition</td>
</tr>
<tr>
<td>IETF</td>
<td>Internet Engineering Task Force</td>
</tr>
<tr>
<td>KPI</td>
<td>Key Performance Indicator</td>
</tr>
<tr>
<td>LTE</td>
<td>Long Term Evolution</td>
</tr>
<tr>
<td>MNO</td>
<td>Mobile Network Operator</td>
</tr>
<tr>
<td>NB</td>
<td>Narrowband</td>
</tr>
<tr>
<td>OTT</td>
<td>Over The TOP</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public switched Telephone Network</td>
</tr>
<tr>
<td>CHF</td>
<td>Call Handling Function</td>
</tr>
<tr>
<td>CIN</td>
<td>Calling Party’s Number</td>
</tr>
<tr>
<td>CLI</td>
<td>Calling Line Identification</td>
</tr>
<tr>
<td>CN</td>
<td>Comfort Noise</td>
</tr>
<tr>
<td>CSMA/CD</td>
<td>Carrier Sense Multiple Access/Collision Detect</td>
</tr>
<tr>
<td>CUG</td>
<td>Closed User Group</td>
</tr>
<tr>
<td>DES</td>
<td>Data Encryption Standard</td>
</tr>
<tr>
<td>Diffserv</td>
<td>Differentiated Services</td>
</tr>
<tr>
<td>DNS</td>
<td>Domain Name Service</td>
</tr>
<tr>
<td>DSCP</td>
<td>Differentiated Services Code Point</td>
</tr>
<tr>
<td>DTMF</td>
<td>Dual-Tone Multi-Frequency</td>
</tr>
<tr>
<td>DTX</td>
<td>Discontinuous Transmission</td>
</tr>
<tr>
<td>EF</td>
<td>Expedited Forwarding</td>
</tr>
<tr>
<td>EG</td>
<td>ETSI Guide</td>
</tr>
<tr>
<td>ENUM</td>
<td>E.164 NUmber Mapping</td>
</tr>
<tr>
<td>ETSI</td>
<td>European Telecommunications Standards Institute</td>
</tr>
<tr>
<td>FNO</td>
<td>Fixed Network Operator</td>
</tr>
<tr>
<td>FoIP</td>
<td>Fax over IP</td>
</tr>
<tr>
<td>FTtX</td>
<td>Fiber To The “x” (n=network, c=curb, b=building, h=home)</td>
</tr>
<tr>
<td>GIC</td>
<td>Group Identification Code</td>
</tr>
<tr>
<td>GPRS</td>
<td>General Packet Radio Service</td>
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<tr>
<td>GRX</td>
<td>GPRS Roaming eXchange</td>
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<tr>
<td>GSM</td>
<td>Global System for Mobile Communications</td>
</tr>
<tr>
<td>GSMA</td>
<td>GSM Association</td>
</tr>
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<td>GSN</td>
<td>Global Subscriber Number</td>
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<td>HD</td>
<td>High Definition</td>
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<tr>
<td>IAM</td>
<td>Initial Address Message</td>
</tr>
<tr>
<td>IANA</td>
<td>Internet Assigned Numbers Authority</td>
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<tr>
<td>IBCF</td>
<td>Interconnection Border Control Function</td>
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<tr>
<td>I-BGF</td>
<td>Interconnection Border Gateway Function</td>
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<tr>
<td>I-CSCF</td>
<td>Interrogating CSCF</td>
</tr>
<tr>
<td>Abbreviation</td>
<td>Description</td>
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<td>--------------</td>
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<tr>
<td>IC</td>
<td>Identification Code</td>
</tr>
<tr>
<td>IEEE</td>
<td>Institute of Electrical and Electronic Engineers</td>
</tr>
<tr>
<td>IETF</td>
<td>Internet Engineering Task Force</td>
</tr>
<tr>
<td>IFT</td>
<td>Internet Facsimile Transfer</td>
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<tr>
<td>II-NNI</td>
<td>Inter-IMS NNI</td>
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<tr>
<td>IM-CN</td>
<td>IP Multimedia Core Networks</td>
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<tr>
<td>IMS</td>
<td>IP Multimedia Subsystem</td>
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<tr>
<td>IMS-ALG</td>
<td>IMS Application Level Gateway</td>
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<td>IPIA</td>
<td>IP Interworking Alliance</td>
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<td>IPPM</td>
<td>IP Performance Metrics</td>
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<td>IP Security</td>
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<td>IP-SM-GW</td>
<td>IP Short Message Gateway</td>
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<td>IPv4</td>
<td>Internet Protocol version 4</td>
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<tr>
<td>IPv6</td>
<td>Internet Protocol version 6</td>
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<td>IPX</td>
<td>IP eXchange</td>
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<tr>
<td>IPX P</td>
<td>IPX Provider</td>
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<tr>
<td>ISUP</td>
<td>ISDN User Part</td>
</tr>
<tr>
<td>ITU</td>
<td>International Telecommunications Union</td>
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<tr>
<td>KPI</td>
<td>Key Performance Indicator</td>
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<td>LTE</td>
<td>Long Term Evolution</td>
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<tr>
<td>MAP</td>
<td>Mobile Application Part</td>
</tr>
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<td>MGCF</td>
<td>Media Gateway Control Function</td>
</tr>
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<td>MGF</td>
<td>Media Gateway Function</td>
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<td>MIME</td>
<td>Multipurpose Internet Mail Extensions</td>
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<td>MNO</td>
<td>Mobile Network Operator</td>
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<td>MoIP</td>
<td>Modern over IP</td>
</tr>
<tr>
<td>MOS</td>
<td>Mean Opinion Score</td>
</tr>
<tr>
<td>MOS_{CQE}</td>
<td>Mean Opinion Score, Communication Quality Estimated</td>
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<td>MSC</td>
<td>Mobile Switching Center</td>
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<td>NB</td>
<td>Narrowband</td>
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<td>NDC</td>
<td>National Destination Code</td>
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<td>NER</td>
<td>Network Efficiency Ratio</td>
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<td>NGN</td>
<td>Next Generation Network</td>
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<td>NNI</td>
<td>Network to Network Interface</td>
</tr>
<tr>
<td>OSS</td>
<td>Operations Support System</td>
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<td>OTT</td>
<td>Over The Top</td>
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<td>P-CSCF</td>
<td>Proxy-CSCF</td>
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<td>PE-router</td>
<td>Provider Edge router</td>
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<td>PGAD</td>
<td>Post Gateway Answer Delay</td>
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<tr>
<td>PGRD</td>
<td>Post Gateway Ringing Delay</td>
</tr>
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<td>PHB</td>
<td>Per-Hop Behavior</td>
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<td>PLMN</td>
<td>Public Land Mobile Network</td>
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<td>P-router</td>
<td>Provider router</td>
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<td>PSTN</td>
<td>Public Switched Telephone Network</td>
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<td>PT</td>
<td>Payload Type</td>
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<td>QoS</td>
<td>Quality of Service</td>
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<td>RCS</td>
<td>Rich Communication Suite</td>
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<td>REL</td>
<td>Release</td>
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<td>R-Factor</td>
<td>Rating-Factor</td>
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<td>RFC</td>
<td>Request For Comments</td>
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<td>RR</td>
<td>Receiver Report</td>
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<tr>
<td>RTC</td>
<td>Real Time Communication</td>
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<tr>
<td>RTCP</td>
<td>Real Time Control Protocol</td>
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<table>
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<th>Acronym</th>
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<td>RTCP XR</td>
<td>Real Time Control Protocol eXtended Reports</td>
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<td>RTD</td>
<td>Round Trip Delay</td>
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<tr>
<td>RTP</td>
<td>Real-Time Protocol</td>
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<td>SBC</td>
<td>Session Border Controller</td>
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<td>S-CSCF</td>
<td>Serving-CSCF</td>
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<td>SCTP</td>
<td>Stream Control Transmission Protocol</td>
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<td>SDES</td>
<td>Source DEScription</td>
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<td>SDH</td>
<td>Synchronous Digital Hierarchy</td>
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<td>SDP</td>
<td>Session Description Protocol</td>
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<td>Signaling Gateway Function</td>
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<td>Session Initiation Protocol</td>
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<td>SIP protocol URI</td>
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<td>SIP with encapsulated ISUP</td>
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<td>SIP-T</td>
<td>SIP for Telephones</td>
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<td>SLA</td>
<td>Service Level Agreement</td>
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<td>Short Message System</td>
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<td>SONET</td>
<td>Synchronous Optical Network</td>
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<td>SP</td>
<td>Service Provider</td>
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<td>SR</td>
<td>Sender Report</td>
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<td>SRTP</td>
<td>Secure Real Time Protocol</td>
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<td>Single Radio Voice Call Continuity</td>
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<td>Signalling System 7</td>
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<td>STQ</td>
<td>Speed processing Transmission and Quality aspects</td>
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<td>TCP</td>
<td>Transmission Control Protocol</td>
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<td>TDM</td>
<td>Time Division Multiplexing</td>
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<td>tel-URI</td>
<td>Telephone URI</td>
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<tr>
<td>TRF</td>
<td>Transit and Roaming Function</td>
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<tr>
<td>TrFO</td>
<td>Transcoder Free Operation</td>
</tr>
<tr>
<td>TrGW</td>
<td>Transition Gateway</td>
</tr>
<tr>
<td>TLS</td>
<td>Transport Layer Security</td>
</tr>
<tr>
<td>TUPUA</td>
<td>Telephone User Part User Agent</td>
</tr>
<tr>
<td>UAC</td>
<td>User Agent Client</td>
</tr>
<tr>
<td>UAS</td>
<td>User Agent Server</td>
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<tr>
<td>UDP</td>
<td>User Datagram Protocol</td>
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<tr>
<td>UDPTL</td>
<td>facsimile UDP Transport Layer</td>
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<td>UMTS</td>
<td>Universal Mobile Telecommunications System</td>
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<tr>
<td>URI</td>
<td>Uniform Resource Identifier</td>
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<td>VAD</td>
<td>Voice Activity Detection</td>
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<tr>
<td>VBD</td>
<td>Voice Band Data</td>
</tr>
<tr>
<td>VLAN</td>
<td>Virtual Local Area Network</td>
</tr>
</tbody>
</table>
3. References

[1] 3GPP, „TS 29.165; Inter-IMS Network to Network Interface“.

[2] 3GPP, „TS 23.228; IP Multimedia Subsystem (IMS); Stage 2“.

[3] GSMA, „IR.34; Guidelines for IPX Provider networks (Previously Inter-Service Provider IP Backbone Guidelines)“.

[4] GSMA, „IR.65; IMS Roaming and Interworking Guidelines“.

[5] i3forum, „Common functionalities and capabilities of an IPX platform, Release 2, May 2014“.

[6] i3forum, „Voice over IPX Service Schedule“.

[7] i3forum, „Enabling HD voice continuity in international calls, Release 1.0, May 2014“.

[8] i3forum, „LTE Data Roaming over IPX Service Schedule, Release 1, May 2014“.

[9] GSMA, „IR.92; IMS Profile for Voice and SMS“.

[10] 3GPP, „TS 24.341; Support of SMS over IP networks; Stage 3“.


[12] i3forum, „Interconnection & Roaming IMS Signaling Profile, Release 2 (May 2013)“.

[13] GSMA, „IR.95; SIP-SDP Inter-IMS NNI Profile“.

[14] 3GPP, „TS 26.114; IP Multimedia Subsystem (IMS) – Multimedia telephony – Media handling and interaction“.

[15] GSMA, „IR.94; IMS Profile for Conversational Video Service“.


[17] 3GPP, „TS 26.441; Codec for Enhanced Voice Services (EVS)“.

[18] GSMA, „IR.36; Adaptive Multirate Wide Band“.

[19] IETF, „RFC 4733; RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals, December 2006“.

[20] ITU-T, „H.264 (04/2013); Advanced video coding for generic audiovisual services“.

[21] ITU-T, „Recommendation H.265 (04/2013); High efficiency video coding“.

[22] „IETF, “ RFC 6386; VP8 Data Format and Decoding Guide."

“IMS-Based Services”, Release 1.0 - Jun 11th 2015
[23] IETF, „RFC 3966; The tel URI for Telephone Numbers“.
[25] GSMA, „AA.81; Packet Voice Interconnection Service“.
[26] IETF, „RFC 2328; OSPF Version 2 (April 1998),“.
[27] IETF, „RFC 4271, A Border Gateway Protocol 4 (BGP-4), (January 2006)“.  
[29] IETF, „RFC 4855; Media Type Registration of RTP Payload Formats“.
[30] IETF, „RFC 6849; An Extension to the Session Description Protocol (SDP) and Real-time Transport Protocol (RTP) for Media Loopback“.
[31] IETF, „RFC 3264; An Offer/Answer Model with the Session Description Protocol (SDP), (June 2002)“.
[32] 3GPP, „TR 23.850; Study on roaming architecture for voice over IP Multimedia Subsystem (IMS) with local breakout“.
[33] 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“, April 2007“.
[34] ITU-T, „P.10/G.100; Telephone Transmission Quality“.
[35] ITU-T, „G.107; The E-model: a computational model for use in transmission planning“.
[36] ITU-T, „G.107.1; Wideband E-model“.
[37] ITU-T, „G.107 Annex B; Quality measures derived from the transmission rating factor R“.
[38] ITU-T, „E.437; Comparative metrics for network performance management“.
[40] ITU-T, „E.425; Internal automatic observations“.

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4. Business & Technological considerations in 2015-‘16

This section contains market and technology dynamics surrounding IMS and its alternatives. Specifically, a discussion is carried out on the technologies that enable IMS (fixed and mobile), on the IMS services (e.g. HD Voice and VoLTE) and on possible alternative solutions provided by OTT.

4.1. Access technologies to IMS

4.1.1. Fixed access

The deployment of Fiber To The Neighborhood / Curb / Building /Home increases Fixed Broadband penetration. This creates an opportunity for Operators to converge their services onto IP and simplify their operations. From a service perspective, it facilitates the support of multi-play offers as well as Fixed-Mobile convergence. This means that the migration to IP of TV, Telephony and supplementary services can be managed from a single IMS infrastructure.

4.1.2. Radio access

Mobile Operators are deploying high speed 4G data networks designed to match fixed broadband performance and thus enabling the migration of mobile services onto IMS.

The vast majority of LTE radio access is being deployed in East Asia, Western Europe and North America. As of June 2015, nearly 400 LTE networks are commercially available in 138 countries. By end of 2020, the GSMA expects that 30% of global connections will be on LTE with over 2.5bl connections compared with 500M as of June 2015. Source – GSMA 2015.
Despite its impressive growth, the deployment of domestic homogeneous LTE coverage and performance is today a challenge:

1. Due to the current spectrum allocation in higher frequency band for LTE, the average cell coverage is lower meaning that more cells are needed than for 3G for example. LTE cell technology is evolving but it remains a key consideration with radio coverage.

2. National geographical characteristics and radio network topology will impact the level of efforts required to provide LTE coverage which matches that of the legacy technology.

3. As LTE access drives mobile data usage, high population density areas can create congestion, which greatly degrade the performance of the network and user experience.

Because of geography, congestion and costs, it may take years to deploy an LTE network with the required ubiquity and performance. In the meantime, LTE “blind spots” will have to be filled with legacy radio access, which means that ubiquitous and performing LTE-accessed services will take time to reach the same level of coverage as in 2G/3G radio access.

Wifi offload provides an opportunity to mitigate LTE coverage and performance hurdles. The wifi ecosystem is undergoing an evolution in coverage, speed and security. It is projected that 57% of LTE capable mobile traffic will be offloaded to Wifi by 2019.

![Figure 4 - Wifi global figures](image)

Wifi does come with challenges of its own. There are however, coordinated efforts underway such as those of the Wireless Broadband Association, to define what evolution path Wifi should take, in order to make the option as productive as envisioned.

Vendors are also deploying VoWifi technology such as enhanced gateways and mobile devices with VoWifi capable features (iOS8, Lollipop). This equipment will support the offloading of an estimated 53% of mobile IP voice traffic to WiFi by 2019, according to 2015 ACG and Cisco research.

### 4.2. IMS-based Services

The IP Multimedia Subsystem (IMS) is a telecommunications model enabling a standardized IP-based access to services of different networks. It was originally specified by ETSI TISPAN and then further developed by the 3rd Generation Partnership Project (3GPP), providing fixed and mobile operators with the opportunity to converge their services onto a unified core network.

IMS’ main service features are:

- Interconnection with legacy networks both mobile (PLMN with 2G/3G radio access) and fixed (PSTN, ISDN, VoIP networks);
- Integrated provisioning of secure voice services both originated by fibre access and LTE access (i.e. VoLTE) by means of the same signalling protocol (i.e. SIP);
- Integrated provisioning of secure videocall services both originated by fibre access and LTE access (i.e. VILTE);

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• Provisioning of legacy SMS service together with a suite of enhanced messaging services (e.g. Rich Communication Suite).

IMS’s convergence strategic objectives are to simplify the network, to enhance the features of existing services and to create the opportunity to deploy new services. As mobile IMS-based services are entering the market, the following challenges will have to be tackled:

1. A complete IP migration with related legacy network platform decommissioning will take time to achieve due to the LTE coverage hurdles mentioned above, as well as the need to support non-IMS capable devices for the foreseeable future.

2. Though an all IP service environment will provide MNOs with more options, ease of deploying new services, and will enable them to get closer to the OTTs’ agility, they will not be able to equal it, as they cannot operate their services independently of considerations related to maintaining service continuity, interoperability and infrastructure, as OTTs do.

3. The success of IMS-based services in competing with OTTs remains to be assessed by the market, even if it is worth underlining that IMS provides a full set of telephony services, including the supplementary ones, and those related to Rich Communications Suite.

4.2.1. Voice over LTE

Voice is the first service MNOs are migrating onto IMS. As of April 2015, 89 operators have announced their commitment to VoLTE and are at various testing or deployment stages. 16 MNOs have launched a commercial VoLTE offer, predominately in East Asia and these services are limited to the domestic market without, for the time being, including roaming scenarios. The vast majority of VoLTE capable mobile subscribers are in South Korea, Singapore, Japan and Hong Kong.

![Operator VoLTE status regional distribution - March 2015](image)

**Figure 5 - Operator VoLTE status regional distribution - March 2015**

Infonetics (2014) suggests that 328M subscribers will be VoLTE enabled by 2018. These subscribers are expected to be concentrated in East Asia and North America.
While the level of commitment and progress seems promising for VoLTE, the actual availability of VoLTE on networks that have launched is limited. In the critical path are LTE coverage limitations, national interoperability and device availability, which pose the following challenges:

- **Commercial challenges:**
  the definition of a proper IMS VoLTE profile taking into account the market conditions; the set of supplementary services already provided via the 2G/3G networks and what has to be deleted / added in the new IMS VoLTE profile;

- **Technical challenges**
  the integration of the fully IP-based IMS and 4G radio access networks with the legacy TDM-based radio and switching networks; the support of SRVCC (Single Radio Voice Call Continuity) capability and the related specification is a key milestone of this technical integration in order to achieve a smooth hand-over between 4G radio cells and the 2G/3G ones during a call.

- **Devices Availability**
  Devices which support VoLTE are currently scarce and typically high end. All major device vendors and mobile OS manufacturers however, are committed to rollout VoLTE-enabled devices.

Operators who do deploy VoLTE will need to also consider off-net SIP interoperability, as well as breakout to circuit-switched networks. This creates another layer of complexity which IPX Providers are well positioned to address. This off-net VoIP enablement is however faced with the reality that MNOs’ Voice international gateways are often TDM-based. This creates another item of friction for the VoLTE ecosystem growth and productivity.

Voice over LTE brings the benefit of HD voice which is one of the features which will increase the value of mobile voice, and may put MNOs on the path to mitigating the impact of OTTs. Mobile HD Voice users do express their satisfaction with the quality and experience of HD Voice.

Because of the above mentioned challenges, VoLTE penetration forecasts are questionable. We may also see a fragmented ecosystem: MNOs, case by case or market by market, will have to weigh the challenges of moving to VoLTE IMS services vs. the new benefits of services and spectrum efficiency.

### 4.2.2. HD Voice service

Successful off-net HD Voice termination requires the call path to be end-to-end-IP, and all devices within the path to be HD compliant. These requirements mean that delivering international HD Voice creates additional costs for direct routing, number portability correction and transcoding. For instance, because of the lack of national number portability database available to carriers, such as in South Korea and UK, ensuring HD compliant routing may be technically and economically more challenging in some cases.

Despite 132 HD Voice commercial networks launched in 81 countries, including 16 via VoLTE (source: GSA, April 2015), HD Voice complexity in delivering actual end-to-end international HD Voice availability will result in a lower growth rate than the adoption of domestic VoLTE. This implies that
international carriers, in particular IPX Providers, can play a key role in enabling and accelerating HD voice growth.

The HD Voice ecosystem is fragmented along the lines of which HD codec is used. This fragmentation produces interoperability issues which have to be addressed with specific transcoding capabilities and codec negotiation arrangements between carriers. In the context of VoLTE, this means that international VoLTE has more technical challenges and therefore could be more costly to terminate than non-VoLTE traffic.

In the longer run, as technologies and standards incorporating high definition codecs gain traction, high definition communications will eventually become ubiquitous and a default feature of all voice services. The growth of VoLTE and WebRTC adoption, together with the growing number of HD-enabled devices and networks, will contribute to simplify the continuity of international HD Voice.

OTTs 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 (HD Voice codecs are described in section 8.2). As an example, the Opus codec is endorsed in the current IETF draft for WebRTC codec requirements 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.

4.2.3. RCS Services

IMS can support enhanced messaging services and specifically Rich Communication Services as proposed by the GSMA. The goal for RCS is to complement MNOs’ traditional services by providing services similar to or better than those which have made OTTs successful. RCS’ latest version includes:

- Enhanced Phonebook
- Enhanced Messaging
- Enriched Calls
- Social presence information
- Group Chat
- Content sharing

40 operators have commercially launched RCS in 33 countries. 87 operators are committed to launch RCS in 2015. (source: GSA and GSMA, end of 2014). Despite some commercial launches, the actual number of RCS users has not yet picked up as expected . One of the hurdles impacting RCS implementation is the standards and interoperability complexity (e.g. RCS hubbing would need substantial interoperability support which IPX Providers may find difficult to justify if the RCS market doesn’t support it).

4.3. Alternative Solutions to IMS

OTT Providers have already offered HD voice and advanced calling services as key drivers on their application, with enhanced services. HD Voice Call or Video Call is embedded with other services beyond the media such as directory management tools, status and presence features, instant messages services, or conference call tools with multi calls connection or document sharing.

According to Telegeography 2015 studies:

- In 2014, two messaging applications, Facebook Messenger and Whatsapp have been downloaded more than 500 million times.
- KakaoTalk has been installed more than 100 million times.
- Skype is the first international provider with nearly 1/3 of the total voice international traffic and 40% of Video Calls in 2014.

WebRTC may further confirm the trend by facilitating interoperability between Web-based applications and the PSTN.

WebRTC has been gradually adopted by the most commonly used browsers - between them accounting for more than half of the world’s browsers. In October of 2014, Microsoft committed to

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including a version of WebRTC on its Internet Explorer browser, leaving only Apple as the main holdout.

WebRTC will be a market worth $4.7 billion by 2018 predicts consulting firm Smith Point Analytics, while U.K based telecom/mobile industry consultant Dean Bubley calculates that over 2 bln people, approx. 60% of the likely internet population, will be using WebRTC by 2019

IPX Providers may choose to deploy WebRTC gateways, thus taking over some of the interoperability functions and maximizing the continuity of WebRTC-based features. To date however, the majority of WebRTC instances are enterprise related, and typically will breakout to SIP or TDM destinations.
5. Architectural Framework based on IPX

Being independent from the access technologies, the IMS framework has been designed with the purpose of serving a set of multimedia applications and enhanced messaging in a full-IP environment. IMS is therefore, inherently a multi-service framework, encompassing services that are built over the IMS infrastructure like VoLTE. Video over LTE and RCS, as well as others that may be carried over IMS as an option, such as a WebRTC-based service.

By deploying an infrastructure capable of supporting IMS-based services, International Carriers can enlarge their commercial offers by allowing IMS-capable Service Providers (Fixed, Mobile, OTT) to interconnect and extend the domestic customer experience to the international domain via interworking and roaming services.

With reference to the voice services over an IMS platform, it is worth underlining that it encompasses both voice originating from mobile networks (i.e. VoLTE) and voice originating from fixed networks (e.g. FTTx and Wi-Fi customers connected to an IMS platform). In all cases, for Service Providers and International Carriers, interoperability between IMS-based networks and non-IMS legacy networks (e.g. PSTN and NGN VoIP), is a primary objective. The issues related to various interworking scenarios are extensively addressed in Section 12.

5.1. IMS Functional blocks for International Carriers

The IP Multimedia Subsystem encompasses a wide set of functional blocks (about 50) and interfaces covering access networks as well as core networks, whose specification and characteristics are thoroughly described in 3GPP specifications, starting from TS 23.228 [2].

The deployment of a complete IMS Core solution is in the scope of Service Providers’ networks which, in any case, have to deploy any or all the following functional blocks in order to meet the interconnection and roaming requirements specified by 3GPP:

- S/P/I-CSCF (Serving/Proxy/Interrogating - Call Switching Control Function)
- HSS (Home Subscriber Server)
- PCRF (Policy and Charging Rules Function)
- IBCF (Interconnect Border Control Function)
- TrGW (Transition Gateway)
- BGCF (Breakout Gateway Control Function)
- TRF (Transit and Roaming Function)

From a Carrier perspective, the focus is to allow IMS networks to connect with each other at a service level, supporting the specified protocols, interfaces and profiles at the IMS Network-to-Network Interface. Hence, the Carriers’ IMS implementations are simpler, involving the deployment of a more limited number of functional blocks.

Considering the scope of this document, two connectivity options are described in the following sections:

a) Service aware; i.e. based on the management of specific IMS protocols;
b) Service unaware; i.e. the Carrier’s network does not manage any specific IMS protocol.

5.1.1. Options for IMS NNI – Service aware

In compliance with the 3GPP IMS specification [2] and documents referenced therein, the service aware IMS interconnection configuration between Service Providers is described in Figure 7. The Lci interface provides signaling connectivity for the Control Plane based on SIP signaling. The Izi interface provides connectivity for session-based media (e.g. User Plane for the voice service).

Note: The SIP signaling across the Lci interface can also provide session-based services within the RCS suite.

The same logical interfaces, Lci & Izi, are applicable to both Service Provider-to-Carrier interconnections, as well as to Carrier-to-Carrier interconnections.

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In addition to the connectivity interfaces, there are network and service level requirements that can be fulfilled by a Carrier in the scope of IMS. Some of these requirements do overlap with features developed under the IPX framework, such as global reach, network security, QoS, and others. Additional requirements like the handling of IMS signaling have to be deployed in new functional nodes that have to act as Proxies. Some of the capabilities that can be supported by an IMS proxy are:

- support of new addressing schemes and routing mechanisms used in the IMS (e.g. SIP URI, Route Header)
- screening of application information and parameters (e.g. SIP screening)
- interworking of different application implementations, manipulation of headers
- access control at both the network and application levels
- NAT/PAT or Application Level Gateway (ALG)
- User Plane adaptation (e.g. transcoding / transrating)
- support of DNS/ENUM
- break-in and break-out mechanisms to non-IMS networks.

The 3GPP TS 23.228 [2] describes the case of an IMS Transit Network/Function in the chain between two IMS Core networks, providing inter-connectivity and transit/routing of IMS services, along with break-in/out to non-IMS Packet Switched and Circuit Switched domains.

To endorse and extend this concept, the GSMA has recognized the possibility for an IPX Provider to enable global interconnections of IMS networks and provide the mentioned features, by means of an “IPX Proxy” (ref. IR.34 [3]) encompassing the features of a subset of 3GPP logical nodes. The i3f recognizes the following list as the main logical components of an IMS enabled IPX Proxy:

- IBCF
- TrGW
- BGCF

From a technological standpoint, the deployment of the mentioned functional blocks can be achieved in a “single box”, or as standalone equipment. In the first case, i3f recognizes that the technical evolution of Session Border Controllers (SBC) can possibly address a subset of the required features, in particular, those of IBCF/TrGW related to security and access control matters.

5.1.2. Options for IMS NNI – Service unaware

Carriers may also provide a service-unaware NNI to their Customers in bilateral or multilateral mode (see Figure 8 from IR.65 [4]), which includes transport capabilities, as well as QoS control and monitoring and security. The IPX Transport service is an example of how this NNI can be implemented between two IMS Core networks.
In the 3GPP reference model, the Mw interface is used for the exchange of SIP signaling messages between CSCF resources, while the Gi/Sgi interfaces connect data network nodes and are dedicated to the transport of the user plane (making use of GRE tunnels). The Border Gateways (BG) shown in figure 6 are SIP-unaware and provide control and filtering of incoming and outgoing IP traffic.

It has to be underlined that this service unaware option only supports IMS Core to IMS Core interworking; therefore, in such a Transport mode, the Carrier is not able to provide any break-in/out mechanisms or those features previously described above that require service-awareness.

5.2. Technical and commercial reference model for the international IP interconnection

5.2.1. Use of the IPX model

As stated in GSMA IR.65 [4], the use of the global IPX framework is not a mandatory requirement for implementing the IMS Interconnect and Roaming network environment. Different interconnection schemes may be used at the NNI level, such as Public Internet or physical / virtual private networks (e.g. leased line or VPN/IPSec.), provided that the requirements of IMS-based services are implemented over the mentioned platforms. These requirements call for specific technical and commercial features and capabilities such as openness, QoS control and monitoring, security, multi-protocol and multi-attachment support and DNS/ENUM resolution.

Nevertheless, it is recognized that there is no reason for Service Providers and IPX Providers to define and deploy a new technical architecture and commercial model specifically for IMS Interconnect & Roaming, different than IPX. A global IPX can already ensure most of the required commercial and technical features listed above; such as the need to achieve a global reach/coverage, and the need to segregate the IP addresses assigned to User Equipment, from the transport Carriers’ networks.

As a further consideration even for the IMS services, the general principle of service community separation can be supported, but a joint careful analysis has to be carried out in order to define the set of optimal service configuration schemes at the NNI.

On the basis of the above rationale and for the sake of simplicity, only the IPX option is described in the following sections. The i3 forum reference IPX technical architecture is described in chapter 5.2 of i3f “Common functionalities and capabilities of an IPX platform” [5]. Figure 9 depicts the IPX Reference Model, with explicit indication of IMS-capable Service Providers that are interconnected to the IPX domain.
5.2.2. Connectivity options

As specified by GSMA and i3 forum for the general IPX model [3], [5], the available connectivity options for IMS-based services are the following:

- Bilateral – Transport Only (transport without service awareness)
- Bilateral – Service Transit (transport with service awareness)
- Multilateral – Hubbing (transport and hubbing with service awareness)

5.2.3. Break-in / break-out

Break-in and break-out options, that are generally possible in the IPX with certain constraints, can be applied to IMS services taking into account, case by case, proper rules and limitations.

It is recognized that Voice over IMS has to allow all possible interworking scenarios from/to legacy voice technologies, such as TDM PSTN/PLMN, Fixed and Mobile VoIP, OTT VoIP, etc. Hence, break-in and break-out are allowed with the same features and limitations as in the VoIPX (see Sec. 5.2 of the i3f “Voice over IPX Service Schedule” [6]).

5.3. Service schedule for Voice over IMS

In the Service Providers’ (fixed and mobile) access and core networks, the implementation of Voice over IMS (VoIMS) service has to be considered a technological step forward for its IP-based intrinsic nature. In the Carriers’ networks, since the migration process towards IP has already started, some of the features and capabilities implemented over the years at the transport layer (e.g. IPX model) can be exploited to offer an effective and efficient management of the VoIMS service as well.

In addition to the connectivity and break-in/out options described above, the deployment of VoIMS should follow some of the basic principles described in the “Voice over IPX Service Schedule” [6]. For Routing and Traffic Management, in line with general IPX requirements (see Sec. 12.2 of [6]), IPX Providers have to implement routing criteria for VoIMS in order to minimize QoS impairments due to network traversals, limiting as much as possible the number of networks to be crossed end-to-end.

Specific importance has to be given also to the setting and management of proper Class of Service (through IP Packet Marking, see Sec. 6.2 of [6]) onto all the networks in the call chain, so as to
guarantee the End-to-End QoS expected in VoIMS service. Further details can be found in Section 11 of this document.

As described in Sec. 7 of this document, in the IMS context there are specific SIP signaling features and profiles for the Voice/Video services. Therefore, among all Signaling Functions described in Sec. 7 of Voice over IPX service schedule [6], the SIP IMS profile shall be supported in the scope of the services described in this document.

Other SIP profiles (e.g. IETF SIP) are deemed not suitable for this purpose.

With regards to Security requirements, all the principles applicable for the IPX environment shall be kept for VoIMS. Future releases will address specific requirements applicable to IMS networks and services.
6. Business Models for IMS-based services

Within the IMS framework, different business models can be identified depending on the specific service to be offered (e.g. voice, video, messaging, signaling).

6.1. Voice over IMS service

For voice over IMS service, 3forum recommends a model that promotes and facilitates a quick adoption. In order to make the transition easier in terms of counting, rating, invoicing, auditing and reconciliation, it is desirable for the IMS voice service to be compatible with the existing voice business model widely used today in TDM and IP networks.

Of course IMS Service Providers and market forces could make possible other business models as well, but to promote IMS transition and adoption, it shall be possible, at least initially, for all players to use an existing standard default business model.

As a result, 3f recommends the IMS voice (HD or SD voice) service to be compatible initially with the current business model based on Calling Party Pays with cascading of the termination price among all involved Service Providers and Carriers/IPX Providers. Similarly, the pricing scheme based on call duration and destination shall be retained.

In addition, the following considerations / remarks have to be taken into account with regard to:

a) Call Routing: whenever possible, the VoLTE service requires direct routing to the destination network, either for continuity of HD voice or other associated services;

b) Number portability resolution: number portability management could be incurred by IPX Providers. It is the responsibility and choice of IPX Providers to negotiate with Service Providers, when these features are required and how the associated costs are charged.

c) Transcoding: in a number of call scenarios it could be the responsibility of an IPX Provider(s) to transcode media from an originating codec to a terminating codec (e.g. from mobile WB-AMR to fixed G.722). It is the responsibility and choice of the IPX Providers to negotiate with Service Providers when these features are required, and determine how the associated costs are charged. For more details refer to 3forum document: “Enabling HD voice continuity in international calls (Release 1.0) May 2014 [7] and Sec. 8 of this document.

6.2. Video over IMS (videocall) service

To be developed in future releases.

6.3. Signalling for IMS-based services

In this section the signaling services which enable IMS services encompass Diameter and SIP signaling.

Diameter signaling controls the transport layer allowing the registration, policy and charging over LTE networks, while SIP runs at the service layer allowing all session-based IMS services.

The H.248 protocol, being internal in Service Provider / IPX provider networks, is out of scope.

6.3.1. Diameter Signalling

For Diameter Signalling over IMS service, 3forum recommends no major changes to what is already specified in “LTE Data Roaming over IPX Service Schedule” [8]. Specifically, the principles and methods related to Diameter signaling accounting, policing and charging have to be maintained.

In terms of business models, it is reconfirmed what was previously recommended in [8]:

Options for signalling service between Service Provider and IPX Provider:

- flat fee (with tiered levels based on usage)
- per-transaction fee

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6.3.2. SIP Signalling for IMS registration

For the time being, there is neither a specification nor market experience examples of how to deal commercially with the flow of signalling information for IMS registration generated by roaming end-users, since this traffic is not present in 2G/3G roaming cases.

i3 forum considers this traffic as chargeable traffic, which represents a component of the total traffic exchanged between the visited network and the home network for roaming customers.

6.3.3. SIP Signalling for session-based services

A Carrier / IPX Provider has to manage SIP signalling for session based services for 2 different call scenarios:

a) for terminating a call between two Service Providers without any roaming party;

b) for managing a roaming call.

In the first case, the signalling information is associated with a media flow which is charged according to the principles given in Sec. 6.1 above and no charge scheme applies to the signalling information.

6.4. IMS messaging services

6.4.1. SMS over IMS

For Short Messaging Services over IMS, i3forum recommends for Service Providers and Carriers/IPX Providers to support GSMA PRD IR.92 "IMS Profile for Voice and SMS" [9] and related 3GPP standards. More specifically a MNO can choose either to fall back to the existing platform or to put in operation an IP Short Message Gateway (IP-SM-GW) as specified in 3GPP TS 24.341 [10].

At the international level, Carriers/IPX Providers provide SMS Transit and SMS Hubbing, and, with reference to the two scenarios mentioned above:

- in the case of a fall back SMS implementation, the existing business model and charging schemes apply; i.e. the charging of MSUs within a flow of SS7 legacy signalling or charging per event;
- in the case of the implementation of a IP Short Message Gateway, given a lack of market trend, it is likely that the same principles in 6.3.2 apply; namely, charging the SIP method “Message”.

6.4.2. Rich Communication Suite over IMS

To be developed in future releases.

6.5. IPX Transport services for IMS

Even if there are no pure data services within the IMS framework, IPX Providers can offer service unaware IPX Transport solutions to Service Providers for the interconnection of IMS-based services. For more details, refer to i3forum document “Whitepaper on IPX Transport Service (Release 1.0, May 2014)”

As an example of such a service, IPX Providers may offer transport solutions connecting:

a) a couple of Service Providers;

b) one Service Provider to any other Service Provider (much like GRX today);

c) one Service Provider and a selected group of one or more Service Providers.
In any case, the service offered will be an IP service unaware transport, with guaranteed QoS and Class of Services.

The business models can vary, but to ensure an easy adoption and facilitate IPX Providers and SPs in terms of counting, rating, invoicing, auditing and reconciliation, it would be preferable for the IMS data transport services to be compatible with the existing IP business widely used today for the GRX service. The service should be counted and charged based on:

- Bandwidth: Mbits-per-second (and not volume MBytes) with a 95th percentile or average or a maximum capacity allowed,
- Class of Service: the transport service(s) can be provided and sold with different Classes of Service that would guarantee different QoS commitments,
- Destination: the transport service can be sold without a price difference per network of destination (like today with GRX), or it could have a variable price per destination component.

For the latter, this would bring complexity to the network as well as IT requirements, and it can only be expected to be an option rather than the default model initially. However, in the medium term, due to a significant cost variance for the transport of 1Mbps between different regions (e.g. between WE Europe and East Coast in USA vs. between Europe and Far East), it is expected that the cost per destination will gradually be implemented.

A different counting per destination methodology can also enable other new business models that would require the cascading of a data termination fee by the terminating network. It is to be noted that a data termination fee would most likely be based in Volume and not in Bandwidth, which would require new network and IT developments.

6.6. Other IMS service-aware services

To be developed in future releases.
7. IMS Signalling Protocols

7.1. Session signalling protocol for voice/video call over IMS

Given that IMS is a platform for an all-IP suite of services, interconnection, interworking and interoperability with legacy signalling protocols as well as with other Internet services has been one of the guiding principles of its specification. Another IMS core concept is “independence”, in the sense that all the services deployed rely on a single and unique protocol for session establishment and control.

The protocol that fulfills these two objectives is SIP (Session Initiation Protocol), an IETF standard widely deployed among Service Providers and IPX Providers. In this respect, it is worth quoting the 3GPP specification TS 23.228 [2]: “In order to achieve access independence and to maintain a smooth interoperation with wireline terminals across the Internet, the IP multimedia subsystem attempts to be conformant to IETF "Internet standards". Therefore, the interfaces specified conform as far as possible to IETF "Internet standards" for the cases where an IETF protocol has been selected, e.g. SIP.”

The SIP protocol specification document is IETF RFC 3261 [11]. SIP provides the primitives to be used for implementing different services, and specifies the signaling message, its format, and the related content needed in order to achieve:

- Device registration onto an IMS platform;
- Session Establishment, including media negotiation;
- Session Modification;
- In-session notifications and messages;
- Session Termination.

For the exchange of content type of the session (voice, video, chat messages...), although SIP itself does not mandate a specific protocol, Session Description Protocol (SDP) is the de facto standard. Every SIP compliant device is able to negotiate the attributes of a session through the exchange of SDP content embedded in regular SIP messages. The 3GPP specification TS 29.165 [1] addresses the Inter-IMS Network to Network Interface (I-NNI) in order to support end-to-end service interoperability through a detailed SIP Profile. The recommendation specifies which SIP methods and headers should be supported by the entities present at that interface.

The SIP IMS signaling as recommended in 3GPP TS 29.165 [1], together with SDP for media information exchange, shall therefore be supported by all the IMS Service Providers, both within their networks and in the Network to Network Interface (I-NNI). For international voice interconnection and for all voice roaming scenarios, i3 forum endorsed that recommendation in the document “Interconnection & Roaming IMS Signaling Profile Release 2 (May 2013).” [12] The document provides an operational specification of the 3GPP document, whereby each table entry describing the SIP IMS Profile, has been analyzed as either being endorsed, modified or marked as not applicable for an international voice service.

7.1.1. Consistency with GSMA IR.95

In PRD IR.95 [13] GSMA has defined a SIP/SDP profile for the interconnection and roaming NNI between IMS networks. This profile covers several IMS based services: VoLTE, Video Call, SMS/IP and RCS services as described in the relevant PRDs. In the document i3F – Interconnection & Roaming IMS Signalling Profile Rel 2 [12] the i3 Forum has defined a minimum profile only for the basic voice service.

Both profiles are based on 3GPP TS 29.165 Rel. 11 [1] and most of the differences between the two profiles can be explained by the difference of services in scope. Two examples of current differences between the two profiles are:

- IR.95 mandates for the UE as audio codecs AMR, AMR-WB, and for interoperability with non-3GPP, G.711. Other codecs (e.g. G.722 and G.729) are for bilateral agreement and out of scope, while i3 Forum defines AMR-WB, G.722, G.711 and G.729 as mandatory and AMR as optional (see also section 8 of this document).
- For routing of outgoing SIP sessions over the I-NNI, IR.95 mentions three scenarios based on SIP URIs, while the Tel URI is classified as ‘not applicable’. i3 forum takes both Tel URI and SIP URI into account (see also Section 9 of this document).

i3 forum intends to endorse the SIP/SDP profile as defined in the GSMA IR.95 document [13], however it is recognized that small discrepancies could exist which will be reviewed in the future.

7.2. Session signalling protocol for RCS
To be developed in future releases.

7.3. Other IMS signalling protocols
The same guidelines that drove the migration to SIP from traditional protocols like ISUP guided the substitution of other legacy protocols such as MAP and CAMEL by Diameter, for mobility, call control and charging. i3 forum already released its own view on Diameter protocol in the deliverable “LTE Data Roaming over IPX Rel. 1 May 2014” [8] where a detailed application of this protocol for international data roaming service is given.

For the sake of completeness, it is worth mentioning the protocol ITU-T H.248 for media control being used within Carriers’ networks, and only in specific implementations, is out-of-scope of this document.
8. Codecs

Many different coding schemes have been defined, implemented and used for international voice service. In an IMS framework, a reduced set of codecs has been standardized in order to guarantee the end to end interoperability and quality of service. More specifically, the media requirements have been specified by 3GPP in TS 26.114 [14] for both mobile and fixed access to the IMS core platform and GSMA confirmed this specification with regard to VoLTE (IR. 92 [9]) and Video over LTE (IR.94 [15]).

In the scope of this document these codecs are divided into 2 categories: narrowband codecs for standard quality communication and wideband codecs for high definition communications. Codecs used in OTT Apps are also shortly discussed.

8.1. Narrowband voice codecs

Narrowband codecs reproduce the audio bandwidth of the PSTN and it is expected that they will be used in IMS-based voice networks for some time.

IPX Providers shall be able to carry all voice media flows encoded as per any of the i3 forum narrowband recommended codecs, to be considered mandatory in this context, listed in the Figure 8 below and shall allow the negotiation of these codecs between both originating and terminating Service Providers.

<table>
<thead>
<tr>
<th>Group 1. Mandatory Narrow band codecs</th>
<th>Group 2. Optional Narrow band codecs</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711 A-law, μ-law 64 kbit/s</td>
<td>AMR-NB</td>
</tr>
<tr>
<td>(Mandatory in network for interworking between IMS networks and other IMS or non IMS networks)</td>
<td>(Mandatory in terminals using 3GPP access to the IMS and in IMS Media Gateways. It is likely that the usage of this codec will spread with the development of VoLTE)</td>
</tr>
<tr>
<td>G.729, G.729a, G.729b, G.729ab 8kbit/s</td>
<td></td>
</tr>
<tr>
<td>(For interworking with existing VoIP networks and support of existing VoIP terminals)</td>
<td></td>
</tr>
</tbody>
</table>

Figure 10 - Mandatory and Optional Narrow Band Codecs for Voice

Note: as far as the conversion between G.711 A-law and G.711 μ-law is concerned, the existing conventions apply (i.e. conversion will be done by the countries using the μ-law).

Provided that at least one of the mandatory codecs is present in the session description protocol (SDP) offer, and provided that at least one of the mandatory codecs is supported by both originating and terminating Service Providers, then codec negotiation is guaranteed to be successful. For destinations where one of the mandatory codecs is not available by the IPX P, these destinations shall be disclosed to the SP.

Specific engineering guidelines for the usage of these codecs are given in Section 15 Annex 2 of this document and in the i3 forum deliverable “Technical Interconnection Model for International Voice Services”, Release 6, May 2014” [16].

8.2. Wideband and super wideband/full band voice codecs

There is a general trend towards the increased use of wideband codecs. They provide superior voice quality and this can reduce voice quality degradation due to transcoding.

Support of wideband codecs by carriers is optional. However, when a carrier supports wideband codecs, this section applies and specifies what needs to be supported. The codecs that shall be supported for wideband transmission are:

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<table>
<thead>
<tr>
<th>Group 1. Mandatory Wideband codecs</th>
<th>Group 2. Optional Wideband codecs</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.722 (generally used by fixed network operators)</td>
<td></td>
</tr>
<tr>
<td>(Mandatory in network for interworking between IMS networks and other IMS or non IMS networks)</td>
<td></td>
</tr>
<tr>
<td>G.722.2 (AMR-WB, generally used by mobile network operators)</td>
<td></td>
</tr>
<tr>
<td>(Mandatory in wideband terminals using 3GPP access to the IMS and in IMS Media Gateways supporting wideband)</td>
<td></td>
</tr>
<tr>
<td>(Mandatory for VoLTE in GSMA IR.92 [9])</td>
<td></td>
</tr>
</tbody>
</table>

Figure 11 - Mandatory and Optional Wideband Codecs for Voice

Note: The mandatory status is conditional on the support of wideband voice interconnection: if wideband voice interconnection is supported, then the Group 1 codecs in Figure 9 are mandatory.

Specific engineering guidelines for the use of these codecs are given in Section 15 Annex 2 of this document and in the i3 forum deliverable “Technical Interconnection Model for International Voice Services” [16]. Additional information about wideband voice codecs and their usages can be found in i3 Forum white paper “Enabling HD Voice continuity in International calls” [7].

With regard to superwideband/full band codecs for voice, it is worth mentioning that the codec EVS (Enhanced Voice Service), has recently been standardized by 3GPP to provide Enhanced Voice Services over IMS [17], and has been included in GSMA VoLTE IR.92 [9] as a mandatory codec if a superwideband / full band voice service is offered.

The first EVS trials are expected in 2016’-17 and consequently, it has to be considered as an optional codec for international IMS services, though its implementation is required to enable an end to end high quality transcoding free superwideband/full band voice quality service.

8.3. Codecs supported for low bit rate transmission

The usage of low bit rate transmission codecs is not foreseen in an IMS environment given that these codecs are typically used on legacy transmission platforms such as satellite links.

8.4. Codecs supported by OTTs

Notwithstanding that the voice traffic generated by OTTs exploits the Public Internet, there is today the delivery of this traffic towards PSTN/PLMN networks and, in the near future, the delivery towards IMS fixed/mobile networks.

For this reason, it is worth mentioning the most important codec used by OTTs: Opus. It evolved for voice from the Silk codec developed by Skype, but it has been substantially modified and they are no longer interoperable. It has been specified by the IETF as the mandatory-to-implement high definition codec for WebRTC. Its characteristics are listed below:

- **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

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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)

8.5. Basic Transcoding guidelines

The control and monitoring of end-to-end QoS is one of the main objectives for an IPX Provider participating in a voice IMS session over IPX. As transcoding adversely affects the quality of the communications, the following guidelines should be applied in all cases (note 1) for minimizing transcoding quality impairments:

1. Transcoding should be avoided as 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/3GPP docs[9], [17], [18]. The same principle applies between two wideband codecs of the same family; for example AMR-WB Robust Sorting and AMR-WB Bandwidth Efficient.
5. No significant quality improvements are expected if a call, in some segments, is converted to wideband versus an end-to-end narrowband quality.
6. If both narrowband and wideband codecs are offered in a voice IMS session, the wideband ones should be placed in the top priority positions in the SDP offer.
7. The order of codec/packetization period preference is determined by the originating terminal and should be honored wherever possible;
8. 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.

Note 1: It should be noted that high quality codecs (e.g. super wideband or full band) with bandwidth larger than wideband may be handled as wideband codecs. The above transcoding principles may be applicable to high quality codecs as well.

8.6. Codecs for Fax and DTMF transmission

Fax: for Fax transmission over IP, ITU-T Rec. T.38 shall be used.


8.7. Management of “early media” information

The P-early media header shall be supported across the II-NNI together with all relevant values as specified in GSMA IR.92 [9] and IR.95 [13].

8.8. Video codecs

8.8.1. Mandatory video codecs

The below table provides the mandatory video codecs for IMS services.
<table>
<thead>
<tr>
<th>Group 1. Mandatory Video codecs (note)</th>
<th>Group 2. Optional Video codecs</th>
</tr>
</thead>
<tbody>
<tr>
<td>H.264 (Mandatory in IMS networks as per in [20] specification)</td>
<td>H.265 (HEVC) (Recommended by 3GPP for Multimedia Telephony Service over IMS [21])</td>
</tr>
</tbody>
</table>

**Figure 12 - Mandatory and Optional Video Codecs**

The mandatory configuration to be supported is the Constrained Baseline Profile (CBP) Level 1.2 and support of level 3.0 is recommended.

It should be noted that the H.264 codec has been specified by IETF as mandatory codec to be implemented in WebRTC clients (browsers and non-browsers) in addition to the VP8 codec [22]. The support of VP8 is consequently not needed for interoperability with WebRTC video services and remains optional for a Carrier/IPX Provider.
9. Addressing and Routing

As services migrate away from a circuit switched environment to IP, the user identification number starts to differentiate from the PSTN numbering scheme. This process generates opportunities as well as challenges to manage.

The opportunities lie with the fact that more granular domains can be created, each with no limitation on the services it can cover. The challenges come from the impact on existing network platforms and related OSS/BSS chain and how to implement this transition phase.

9.1. Addressing

Two basic addressing schemes can be identified:

a) Tel URI [23] which endorses the traditional ITU-T E.164 addressing scheme (see [Ref. i3f Tech doc Rel. 6] for additional information);

b) SIP URI [24] which links the user identification with his network domain, (sip:+14085551212@domain.com;user=phone or sip:abcdef@domain.com). For mobile networks GSMA in IR.67 further specified the user identification string as: sip:+14085551212@ims.mnc<MNC>.mcc<MCC>.3gppnetwork.org;user=phone

The migration of Tel URI to SIP URI has a profound implication on how IPX Providers mediate and terminate calls:

- From the technical perspective: routing by domain will increasingly differ from routing by dial code ranges (E.164) in that the addressing scheme becomes decoupled from the PSTN structure. This will impact existing already complex routing engines.

- From the operational perspective: traditionally, Service Providers and IPX Providers provide international voice termination pricing by dial code ranges. Pricing by domain will increasingly replace the legacy scheme implying a new way of processing and managing the pricing data.

- From the business/commercial standpoint: IPX Providers termination pricing is typically, and in part, a function of the destination network's cost. As costing by domain becomes necessary, so will pricing by domain.

Because of this evolution, the routing, costing and pricing systems will have to be adapted, which implies that IPX Providers will be facing substantial system investments. In the short run however, most of the impact will be on the technical side as it relates to the support of the new addressing scheme.

9.2. Routing

The goal of routing engines is to achieve the call path compliance relative to the service objectives. In the case of IMS-based services, call path compliance is key to delivering on HD voice for instance, or Video. More specifically, when IMS-based service features require the continuity of specific encoding or protocol, the routing over a compliant network path is key.

The routing Carrier / IPX Provider needs advanced knowledge of key information required for the appropriate routing to be applied:

- Network on which the device is currently anchored: was the number ported? Is the user roaming and visiting another network?
- Capability of the network on which the device is registered: is the current network IMS-capable?
- Capability of the called device: is it IMS-service capable? Which service is it capable of?

Number portability database and solution are widely available in the market. They however have some key shortcomings:

1. Coverage is limited – some VoLTE markets do not have a NP database available to carriers

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2. Where there is National NP support, the costs to access the information are often substantial. An ENUM resolution model could enable this discovery. It is envisioned to be structured along the lines of a cascading resolution process:

- Tier 0 authoritative resolution of Tier 1 index. It determines which country Tier 1 database to lookup.
- Tier 1 query returns the national operator Tier 2 database to query.
- Tier 2 query returns the information on the user device: it confirms whether it is indeed assigned to the network and its service capabilities.

Pathfinder, a GSMA initiative contracted to Neustar, is one of the available options. It has been designed as a global root (T0), with nationally administered T1s, but as of today it doesn’t seem to be taking shape. There is currently no global solution for such cascading resolution process and there are gaps in the critical path to the complete implementation, for example:

- Number portability is not integrated in the ENUM model. This means that there is no current network discovery mechanism prior to querying T2.
- Also, not all countries maintain a T1 national database.
- Similarly, not all operators maintain and make available their T2 database for resolution.

An alternative to ENUM would rely on the routing IPX Provider’s ability to register the requested features based on SIP/SDP signalling and route on the basis of its NNI and destination network capability. In the case of HD voice for example, the routing carrier registers that the seized call requests a wideband codec as a first choice. The carrier routes the call to its HD compliant route if the destination network, corrected for number portability when possible, is HD capable. In this case, codec-aware routing enables compliant routing but the relevance and predictability of the outcome is limited by the lack of prior T2 resolution.
10. Security
To be developed in future releases.
11. QoS control and monitoring for VolMS

A key requirement for the offering of telecommunications services is the capability to monitor and guarantee predefined levels of quality of service. Indeed, in the legacy TDM networks, a number of parameters have been defined and implemented at the service layer; (e.g. Network Efficiency Ratio, NER; Average Length of Conversation, ALOC) and the usage of these parameters is widely spread in the Telcos’ Operational departments as well as in commercial negotiations.

In an IP domain, in addition to the well-known service layer parameters, quality parameters related to the IP layer also have to be taken into account, aiming to a local significance as well as to an end-to-end measure.

This section, after having listed the set of QoS parameters to be considered, deals with the issues related to the monitoring of these parameters both across a single IPX Provider’s network and across the chain of two IPX Providers’ networks, taking into account the available technical methodologies, their in-field implementation and the related QoS commercial negotiations.

The analysis is carried out considering that QoS parameters are defined for the purpose of:

- Monitoring (supervision) against pre-set thresholds;
- SLA compliance and QoS reporting for IPX Provider to IPX Provider interconnections and IPX Provider to SP interconnections.

Note 1: SLAs only apply provided that the load over the originating and terminating SP interconnections do not exceed agreed upon threshold as recommended by the GSMA.

Note 2: The definition of any specific commercial agreement associated with SLAs and/or QoS reporting is outside the scope of this document.

Note 3: In the following when a specific measure of a QoS parameter is discussed, the terminology “Key Performance Indicator” (KPI) is used.

11.1. QoS parameter identification

On the basis of international standards and operational / commercial practices, the following QoS parameters are considered as the most relevant and they are divided into two sets: one pertaining to the transport layer, and the other to the service layer, as follows:

Transport parameters

- round-trip delay
- jitter
- packet loss

Service parameters

- MOSCOE / R-factor
- ALOC
- ASR
- NER
- PGRD

Note 4: PGRD is preferred over PGAD (Post Gateway Answer Delay) because the latter depends on the end-user behavior.

Additional parameters can be measured by IPX Providers for the support of VolMS service.

Service Availability

*The definitions of the above parameters are given in Annex 2 “QoS parameters definition” (Sec. 15).*

11.1.1. CLI Management

For VolMS service the management of the Caller Line Identification (CLI) is a key requirement for roaming, charging and operational practices.
In an international IMS environment, it is mandatory that international IPX Providers will pass on the CLI unaltered. IPX Providers, under normal operational conditions, are not expected to check CLI validity. Even for VoIMS service, which is considered a high quality service, an IPX Provider cannot guarantee that:

- the CLI will be transmitted by the originating Service Provider;
- the CLI received from the originating Service Provider is a valid value, i.e. a value of a CLI owned or ported to Service Provider, and indeed, is the correct CLI for the calling party;
- the CLI forwarded to an interconnecting IPX Provider will be delivered to the terminating user, or delivered without any error being introduced beyond the interconnecting IPX Provider.

IPX Providers can ensure that a CLI received is always passed on unmodified across their own domain except in the case to change CLI from national format to international format. A CLI in SIP would normally be in the format specified in section 9.1 of this document, so no change of format would be necessary. IPX Providers can also have specific agreements with other interconnecting IPX Providers in order to guarantee CLI transparency.

11.2. Reference QoS scenario in an IPX environment for VoIMS service

In compliance with the content of Sec. 5, for VoIMS service, an IPX model is assumed. Hence, it is necessary to analyze the GSMA QoS requirements for voice service over an IPX platform, listed in AA.81 [25] which call for:

- the measurement and reporting of transport-level KPIs for packet loss, packet delay and packet jitter; for the whole IPX Provider domain, i.e. from the first piece of equipment in the IPX Provider’s network facing the originating Service Provider, to the last piece of equipment in the Carrier’s network facing the terminating Service Provider.
- the measurement and reporting of the service-level KPIs

Note: SLA may or may not also include the local tail., This is a commercial decision for the IPX Provider – Service Provider relationship.

- the carrying out of the above measurements follows the forwarded path (dictated by the service) and not the shortest path driven by OSPF / BGP / other IP routing protocols; [26], [27].

The figure below describes the reference configuration for QoS measurement.

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**Figure 13 - Reference configuration for QoS measurement**
11.3. Technical implementation of GSMA quality requirements

11.3.1. Traffic class transparency

In compliance with GSMA IR.34 Sec. 6.2 [3], IPX Providers are committed to managing the VoIMS traffic, ensuring Traffic Classes according to the QCI value received from the Service Providers.

Specifically, for the voice service, this implies the management of the voice packet traffic as Conversational Service applying the Diff. Serv. PHB code “EF” (Expedite Forwarding) equivalent to the DSCP code “46” (in decimal base).

11.3.2. Measurement of Transport parameters

The above described scenario and requirements call for the ability to measure the identified transport parameters across two specific network domains (see Figure 13 above). It should also be possible to analyze the call flow in order to locate and isolate faults.

**Measurement via RTCP**

On the basis of the extensive analysis carried out by i3 forum jointly with other bodies and vendors, there is only one protocol, RTP Control Protocol (i.e.: RTCP) per call which reports back the quality information of the downstream networks, but:

- the RTCP stream is generated by the RTP endpoint and it propagates back across all border functions in the path. Since no information is available in the RTCP reports indicating where the actual RTCP end-point is located in the downstream networks, there is uncertainty on the segment actually being measured;
- transcoding functions generate a new RTP/RTCP stream, thus making the measurement unreliable;
- the solution assumes that the carrier network elements fully support IETF RFC 3550 [28] and IETF RFC 4855 [29] and generate RTCP reports.

As a result, the RTCP measurement scheme is not suitable to meet the above GSMA requirements of an end-to-end measurement process. More specifically, it is not possible to have via RTCP a direct, reliable and accurate measure of transport QoS parameters from the originating Service Provider edge to the terminating Service Provider edge.

**Measurement via Media Loopback**

An approach available in the standard (see IETF RFC 6849 [30]) is the active measurement methodology based on media loopback.

The establishment of the requested loopback type is initiated by a “loopback source” using new SDP media attributes, thereby providing the capability to monitor the quality of the media in an active session using the offer/answer model [31] to establish a loopback connection. Hence, this methodology is based on dummy calls generated by the ingress Border Functions of the 1st Carrier / Service Provider up to the egress Border Functions of the last Carrier / Service Provider.

The downside of this methodology, to be carefully considered, is the number of required testing calls, which significantly increases when the number of routes to measure increases. For the sake of information, assuming a conservative approach where all IPX Providers are fully meshed and all routes of each Carrier / IPX Provider are used by all other IPX Providers, for a domain with 20 Carriers / IPX Providers, each with 8 POPs generating 2 calls / h , call duration 30 sec, each IPX Provider has to generate nearly 916k calls / month.

As of today, there is no information that the Media Loopback standard has been implemented or is going to be implemented in any Service Provider / IPX Provider network.

**Usage of ICMP**

Today in IP networks, Internet Control Message Protocol (ICMP), is widely used in the provisioning phase and in the fault management process, but it is not recommended for the purpose of QoS Control and Monitoring since it:

- does not guarantee reliable delay measurements being subject to low priority treatment and filtering actions.

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**Aggregation methodology**

A different approach has been proposed by i3 forum in “Common functionalities and capabilities of an IPX Core Platform” Rel. 2 May 2014 [5]. It assumes that each IPX Provider measures the performance of its own network, and these measures are subsequently aggregated by the IPX Provider connected to the originating Service Provider to assess the end-to-end performance.

As an example, in the figure below two IPX Providers are connected and IPX Provider A is in charge of producing an end-to-end report for originating SP-A across IPX Providers A and B.

![Figure 14 - Aggregation based approach](image)

In this document the delay on the NNI between two IPX Providers is assumed to be negligible because in the vast majority of the cases, they interconnect in Tele Houses / Carrier Hotels. If this is not the case, then transmission delay has to be added and considered an offset.

The performance across two domains is estimated by aggregating the performance across each domain. In [5] the computation of delay, packet loss and jitter is extensively dealt with.

Consensus is required from the involved IPX Providers in order to report the requested QoS data to the originating SP. As of today, in the market, there is no implementation of the aggregation scheme as well as of any of the previous mentioned QoS measurement methodology. IPX Providers, assuming a pragmatic approach, meet Service Providers requirements of predefined QoS levels in SLA by assuming historical data of the downstream network performance.

**11.3.3. Measurement of Service Layer parameters**

As far as the measurement of the service parameters is concerned, following the consolidated market trends and technological capabilities, the Service Providers’ requirements can be satisfied by existing methodologies and systems already implemented by Carriers with the exception of MOS\textsubscript{CQE}.

Since some parameters (e.g. NER, ALOC) have a significance from the IPX Provider networks down to the end-user in the terminating Service Provider network, the above statement implies that the quality level of the these parameters can be affected by the quality of the terminating Service Provider network.

The parameter MOS\textsubscript{CQE} can be computed only for a specific network domain, i.e. each IPX Provider can estimate MOS\textsubscript{CQE} for its own domain on selected routes. In general this computation is made by means of external probes located in the PoPs at the border of the network.

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11.4. KPI computation for SLA / QoS reporting

Whatever the definition of a specific QoS parameter and its measurement process, the KPI of this parameter has to be estimated at the operational level by means of a series of measures that generate statistical samples. These samples, properly computed in accordance to a selected statistical function, give the requested KPI.

The following measurement scheme is proposed. Let:
- T be the reporting period (e.g. T = one month)
- i be the index of the suite of measurements by the Border Function and/or probes and/or Call Handling Function (as applicable)
- KPIi be the measured value of the i-th sample for the considered KPI (e.g. RTD)
- N be the number of measurements over the period T (i=1..N)

The generic KPI is computed as a function of all the measured “N” samples \( KPI_i = f(KPI_1, KPI_2, ..., KPI_N) \) over a time period, the length of which is outside the scope of this document.

The following functions are suggested:
- RTD: 95 / 99 % percentile or average
- Packet loss: 95 / 99 % percentile or average
- Jitter: 95 / 99 % percentile or average

As far as the market practices are concerned, it has to be noted that, from a commercial perspective, the statistical function “average” is the preferred option in most of the cases.

With regard to the measurement of the Service Availability, in general, it is not possible to assess it neither analytically (due to the intrinsic complexity of an end-to-end path involving multiple network equipment and multiple paths) nor operationally (due to the overload it could generate in the network by means of continuous injection of traffic).

With reference to the equipment and systems to be used for carrying out these measurements, a number of technical options are available on the market, encompassing external probes as well as internal testing routines to be launched by network elements. Additional Business Support Systems are required for the statistical post-processing computation.

11.5. Managing QoS

11.5.1. Managing QoS at the commercial level

As a general principle, each IPX Provider can offer KPIs of QoS parameters according to its own commercial policy. As a result, each IPX Provider is free to select the QoS parameters subject to QoS Control and Monitoring as well as the related configuration parameters of the operational process for collecting data (statistical samples) in order to produce the KPIs.

11.5.2. Managing QoS at the operational level

There are two possible general methods for QoS control and related SLA enforcement:

a) **SLA enforcement through fault management**

A QoS problem is raised upon SP Customer request claiming a QoS degradation by opening a trouble ticket with its serving IPX Provider. The IPX Provider (IPX \( P_A \)) and the SP will then work together to verify if there is an end-to-end QoS fault. In such a case, IPX \( P_A \) will start troubleshooting within its own network and, in the event that no cause of degradation is detected, it passes downstream (cascading) to the interconnected IPX \( P \) (IPX \( P_B \)) the task of solving the problem. If the problem is identified, and if the repair duration is above the limits set in the SLAs, then the IPX Ps must pay the penalties negotiated in the contract.

b) **Enforcement through constant monitoring and reporting of KPI values**

In this option, an IPX Provider constantly measures the QoS in its network (e.g. RTD, NER) and reports these values to its customers; for example, on a monthly basis. This last option can be
very difficult to manage and not fully scalable for hundreds of routes. This solution is only optional and it is up to each IPX Provider to decide to offer it for one or several routes.

In reality, regarding both operational and commercial QoS, an IPX Provider acts both according to methodology 1 and 2 above, aiming to maximize QoS performance, while optimizing operational efforts.

An SLA can foresee penalties in case that the agreed QoS levels are not met by the contracted IPX Provider.
12. Interworking cases for voice services

As described above, the technology for voice telephony is currently in transition from a TDM-based legacy platform towards IP-based solutions both as basic VoIP service (today widespread globally) and as a service of an IMS platform (currently limited to small geographical areas).

The traditional voice service is also in the process of being enhanced with features like high quality voice (HD Voice), adding pictures or video to a voice call, and other means that enrich a final customer experience. These features call for a different set of supported services and new requirements in terms of signaling, interworking / interoperability, routing and transcoding.

This section is focused on analyzing and discussing the basic functions to be performed by Carriers/IPX Providers in a wide variety of transmission scenarios between IMS networks and legacy networks, and between fixed and mobile IMS networks. In this respect, the following four cases have been identified covering all voice call scenarios:

Case A): Calls originated from a fixed IMS to be terminated to a fixed IMS and, in an equivalent way, all originated from a mobile IMS 4G to be terminated to a mobile IMS 4G

Case B): Calls originated from a fixed IMS to be terminated to a mobile IMS 4G and vice versa

Case C): Calls originated from a non-IMS (fixed TDM or mobile 2G/3G) network to be terminated to an IMS (fixed IMS or mobile IMS 4G) network and vice versa

Case D): Calls originated from an IMS (fixed IMS or mobile IMS 4G) network to be terminated to a VoIP legacy network (including OTTs) and vice versa.

12.1. Case A) from IMS to IMS (with no fixed/mobile interworking)

![Figure 15 – From IMS to IMS (with no fixed/mobile interworking)](image)

**Services:** In this use case, voice services can be available in all variants, e.g. voice SD and HD, video, ready to future enhancements. Due to the different characteristics of fixed and mobile devices, additional and supplementary services are limited by the functionalities of these devices and the related network capabilities.

**Physical Interconnection:** standard IP interconnection over multiple transmission systems (see. “Technical Interconnection Model for International Voice Services”, Rel. 6, May 2014, [16])

**Signalling:** Support of the standard IMS Signalling as per 3GPP specification TS 29.165 complemented by i3 forum deliverables [1]. No interworking / interoperability is required.

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Transcoding: Codec transparency is guaranteed. For fixed networks no transcoding is needed in this scenario assuming the endpoint devices are HD enabled, whereas for mobile networks because of the diversity of AMR-WB codecs (e.g. bandwidth efficient or octet-aligned) transcoding may be required ensuring in any case the end-to-end HD quality.

Routing: in addition of the basic Voice over IPX requirement of max two IPX Providers end-to-end (see [5]) two addressing schemes can be used:
   a) Tel-URI or SIP-URI user=Phone: no impact for the Carrier / IPX Providers networks;
   b) SIP-URI Alphanumeric: this option requires important changes in voice service platform and in its relationship with the OSS/BSS systems.

12.2. Case B) from IMS to IMS (with fixed/mobile interworking)

![Figure 16 – From IMS to IMS (with fixed/mobile interworking)](image)

Services: In this use case, voice services can be available in all variants, e.g. voice SD and HD, video, ready to future enhancements. Due to the different characteristics of fixed and mobile devices, additional and supplementary services are limited by the functionalities of these devices and the related network capabilities.

Physical Interconnection: standard IP interconnection over multiple transmission systems (see. “Technical Interconnection Model for International Voice Services", Rel. 6, May 2014, [16])

Signalling: Support of the standard IMS Signalling as per 3GPP specification TS 29.165 complemented by i3 forum deliverables [1]. No interworking / interoperability is required.

Transcoding: If the call cannot successfully negotiate a common wideband codec on each side (e.g. for fixed handset G.722 and for mobile handset AMR-WB), then the transcoding between these codecs can be done by one of the two SP on either side or by the IPX Provider in between or in case of multiple IPX Providers, by one of these IPX Providers.

However, it is common practice in the market that the originating SP takes care of transcoding. In any case, there is the certainty to set-up the call using the G.711 codec.

Routing: in addition of the basic Voice over IPX requirement of max two IPX Providers end-to-end (see [5]) two addressing schemes can be used:
   a) Tel-URI or SIP-URI user=Phone: no impact for the Carrier / IPX Providers networks;
   b) SIP-URI Alphanumeric: this option requires important changes in voice service platform and in its relationship with the OSS/BSS systems.
12.3. Case C) from legacy networks to IMS and vice versa

![Figure 17 – From legacy networks to IMS](image)

**Services:** for this use case, voice is the basic service to be provided. The expansion to additional and supplementary services is limited by the functionalities of both legacy and IMS networks and the interworking capabilities between these two types of networks.

In mobile 3G networks, HD voice can be offered by MNOs provided that they support transcoder free operation (TrFO). If Carriers / IPX Providers can manage HD codecs or better, are TrFO enabled, even in this use case HD voice can be offered end-to-end.

**Physical Interconnection:** standard IP interconnection over multiple transmission systems (see. “Technical Interconnection Model for International Voice Services”, Rel. 6, May 2014, [16])

**Signalling (i.e. from ISUP to SIP / IMS):** The signaling interworking and interoperability is typically performed by the 1st IPX Provider between the calling and called party.

**Transcoding:** If different codecs are declared from the originating and terminating party, then the transcoding between these codecs can be performed by one of the two Service Providers or by the 1st IPX Provider (i.e. the closest to the originating party) or by the 2nd IPX Provider (i.e. the closest to the terminating party).

However, it is common practice in the market that the originating Service Provider takes care of the transcoding. It is also common practice, that if no better codecs can be selected, then the G.711 codec is selected on both sides as the codec to be used for the call. As a result, in this scenario, the end-to-end communication is mainly implemented by means of the G.711 codec. With regards to the support of HD Voice:

a) for fixed PSTN networks: this service is not available;
b) for mobile 3G networks: this service is possible based on TrFO and IP based backhauling.

**Routing:** in addition to the basic Voice over IPX requirement of max two IPX Providers end-to-end (see [5]) only the addressing schemes based on E.164 apply (i.e. Tel-URI or SIP-URI user=Phone see section 9.1 of this document).

No specific requirements to the Carrier/IPX Provider apply.

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12.4. Case D) from IMS to VoIP and vice versa

Services: In this use case, the transport layer is fully IP but interoperability between different signaling protocols and codecs (e.g. transcoding an OTT proprietary codec) is needed, resulting in a related impact on quality of service (e.g. part of the call path may be over Public Internet).

The voice service can be offered in the SD and HD variants. Interoperability of supplementary services is not natively ensured and, if needed, may require additional implementations for SPs and IPX Ps.

Physical Interconnection: standard IP interconnection over multiple transmission systems (see. "Technical Interconnection Model for International Voice Services", Rel. 6, May 2014, [16])

Signalling: The signaling interworking and interoperability is typically performed by the 1st IPX Provider between the calling and called party.

Transcoding: If different codecs are declared from the originated and terminating party, then the transcoding between these codecs can be performed by one of the two Service Providers or by the 1st IPX Provider (i.e. the closest to the originating party) or by the 2nd IPX Provider (i.e. the closest to the terminating party).

In this case, in addition to standard telecom-originated codecs, also codecs typically used by OTT Providers (e.g. Opus) have to be considered. Currently, OTTs, in addition to their proprietary codecs, also support the codecs standardized in the telecom world (e.g. G.711 and G.729).

Routing: in this case in addition to the two addressing schemes specified in the telecom industry (Tel-URI and SIP-URI) already mentioned in the other cases; the proprietary addressing schemes of the various OTT Providers has to be taken into account. As of today, it is a widely solid market trend that the mapping from the telco addressing scheme to other addressing scheme is carried out in the OTT Providers domain.
13. VoLTE roaming scenarios

In existing circuit switching (CS) mobile systems, two basic roaming schemes have been adopted (OMR scheme, though specified, has presently no implementation in the market):

a) *Via Visited PMN routing* (or local break-out): the visited network routes voice calls for inbound roamers directly to the requested destination network. The VPMN is responsible for all wholesale costs associated with the termination charges for routing those calls. The HPMN is then charged per minute for the usage of the VPMN network including the wholesale termination costs when applicable;

b) *Via Home PMN routing*: in this case the visited network has an agreement with HPMN to use CAMEL triggers hence the HPMN has the possibility to retrieve the call (signaling and media) on its own network and then terminate the call.

Since the early beginning of mobile communications, the mobile community has been adopting these roaming schemes for all CS mobile systems and the reasons for this wide adoption are:

- they are service aware: all involved networks (HPMN, VPMN and Int. Carrier(s) know via signalling that a voice call has to be terminated);
- the related signalling protocols (SS7 and CAMEL) are worldwide accepted standards;
- the standard voice business model is retained: the calling party pays with a deterministic charging scheme based on destination and call duration;
- for most call scenarios, the call routing can be optimised to follow the shorter path to the destination network.

It is worth outlining that in the CS environment, from the Int. carrier/IPX Provider perspective; two different and distinct services are offered to MNOs: an international voice service for terminating the call and a signaling service.

In a VoLTE IP-based scenario, the above well-established paradigm has been put under question in the GSMA and, in the weeks when this deliverable is released, two basic alternatives are still under discussion:

a) **Service aware option**: based on SIP IMS signaling being managed by all networks in the call chain and assuming the same business model of the CS scenario; and

b) **Service unaware option**: based on a data connection, exploiting the data layer of the LTE networks (S8 interface of Evolved Packet Core) from the Visited up to Home network where the IMS Core is located.

The remainder of section 13 shortly discusses the two options as well as offers a statement which aims to present the i3 forum position on this debate.

13.1. Service aware roaming option

This model has been already specified by GSMA in IR.65 “IMS Roaming and Interworking Guidelines Version 15.0” October 2014 [4] and it is based on the SIP IMS signaling for allowing the exchange of information between VPMN, Int. Carrier/IPX Provider, HPMN both on the originating side and the terminating side.

The technical specification has been developed by 3GPP in the document TR 23.850 [32] which has been conceived and designed with the objective of replicating the 2G/3G business model and related charging scheme. Considering that in IMS, the call control is performed by Serving – CSCF and the service policies are located in the Telephony Application Server, 3GPP worked out the RAVEL model which encompasses two cases:

a) **Service aware VoLTE model via HPMN routing**

b) **Service aware VoLTE model via VPMN routing**
From the Int. Carrier/IPX Provider perspective, the support of this VoLTE roaming model implies the need to support the SIP IMS signaling profile (see Sec. 7) and specifically, the support of the SIP Route header and OMR parameters in order to correctly forward the signaling information towards the Home Serving-CSCF according to the RAVEL scenario.

In addition, while in 2G/3G there might be the case where no CAMEL agreement is in place between the two MNOs and all calls are necessarily routed by VPMN, in VoLTE, since in any case the signaling information goes back to HPMN, the Home network can decide on a call-by-call basis, whether to route the call to its destination via its own network or to leave this task to the VPMN. This implies that, in the case that the HPMN of the calling party roamer opts for a VPMN routing, the managed SIP IMS signaling information has no associated media in the segment between the VPLMN and HPLMN. From a commercial standpoint, today's clear distinction between a voice service and a signaling service disappears.

13.1.1. Service aware VoLTE roaming call with HPMN routing

In this case, the Home PMN retains full control of call routing and signaling, and media follows the same path. In terms of latency, in case of calls not terminating in the HPMN or in a PMN of the home country, the delay could be significant especially when intercontinental routes are involved.

The following figure depicts the signaling and media paths, showing the main IMS functional blocks involved.

![Figure 19 – Service aware roaming model with Home PMN routing](image)

13.1.2. Service aware VoLTE roaming call with VPMN routing

In this case, the Visited PMN, after exchanging information with the HPMN, is responsible for call routing. This scenario is the closest to the present 2G/3G roaming model and requires the support of Int. Carriers/ IPX Providers in case the call has to be terminated outside the VPMN or the VPMN's country.
The media path can be optimised, and there is clear distinction of the signalling path.

The following figure depicts the signalling and media paths, showing the main IMS functional blocks involved.

![Figure 20 – Service aware roaming model with Visited PMN routing](image)

13.2. Service unaware (via S8 interface) roaming option

In the Summer of 2014, GSMA launched a task force named Revolver whose aim is to assess the feasibility and suitability of some VoLTE roaming models. In addition to the service aware models described in the previous section, Revolver recommended to consider a new scheme based on a roaming data connection (service unaware).

The basic features of this model are the following:

a) VPMN and HPMN are interconnected by a data channel (e.g. IPX Transport) where specific QoSs are provided in the Radio Access network and then mapped to the expected DSCP / Traffic Class in the Core and Transport networks; the S8 interface is used between the two LTE Evolved Packet Cores;

b) this data path encompasses both signaling and media without service awareness on the transport networks;

c) HPMN of the calling party has the full control over routing to the destination and is responsible for any interconnect fees associated with call delivery;

d) given that the transport between VPMN and HPMN is service unaware, the HPMN IMS provides a UNI interface to the calling party, thus an EPC interworking is needed between VPMN and HPMN; and as a result, the VPMN is not service aware;

e) VPMN needs to support APN/QoS based charging;

f) once signaling has reached the HPMN, it makes use of standard IMS SIP signaling when routing the call towards the destination network. In case the terminating party is in roaming, a new data path is generated between the terminating HPMN and the terminated VPMN.

The following figure depicts the signaling and media paths showing the main IMS functional blocks involved.
Figure 21 – Service unaware roaming model based on S8 Home Routing

From the commercial perspective this model has been proposed and supported by MNOs who have already launched VoLTE services and like to offer roaming services to their own subscribers without waiting for an IMS implementation in the visited network.

In terms of the business model, it clearly marks a difference with respect to the existing 2G/3G roaming scheme, moving from voice to a data service, where charging is based on volume (packets) rather than duration (minutes). There is no firm commercial guideline in the market, but technically, it requires that MNOs are capable to properly distinguish this roaming traffic (different APN and different QCI management) as well as necessitating that IPX Providers have the capability to properly charge differentiated quality IPX transport service.

*It is worth noting that, notwithstanding the 3GPP and GSMA specifications, an MNO can always offer a packet voice service to their own roamers as a data service with a Best Effort quality level, which does not offer any specific VoLTE roaming agreement with the visited MNO and exploits the existing data traffic interconnection.*

From the technical standpoint, there are still a number of open points on how to guarantee emergency calls and lawful intercept in compliance with the different regulations, and in terms of latency, the same considerations already given in sec. 13.1.1 for the service aware Home routing model apply: it could imply a longer transmission path than a VPMN routing.

13.3. i3 forum position on VoLTE roaming

In late 2013 – early 2014, i3 forum endorsed the GSMA service aware models and gave its own contribution with regard to the management of a specific SIP header (specifically the SIP Route Header).

In 2014 some liaison statements were sent to the GSMA Revolver Task Force outlining the risks of further delays in reopening the discussion on VoLTE roaming scenarios and raising some questions on the management of two scenarios.

*In June 2015, i3 forum welcomes a clear specification from GSMA on this key aspect of VoLTE services, confirming the availability of i3 forum members to support both of the two scenarios under discussion, as well as expressing their preference for the service aware (signalling based) scheme. This preference is supported by the guaranteed QoS levels and the continuity of the legacy services such as emergency calls (with location based) and lawful intercept.*
It is an i3 forum objective to release a more detailed analysis from the technical and commercial perspective of the VoLTE roaming models once a definite specification is released from GSMA.
ANNEX 1 - Codecs engineering guidelines

**NB Codecs engineering guidelines**

Packetisation period for mandatory Narrow Band codecs:

- for G.711 A-law and μ-law, the packetisation period shall be 20 ms.
- for G.729, G.729a, G.729b, G.729ab, the packetisation period shall be 20 ms.

**Payload type definition for mandatory Narrow Band codecs:**

- G.711 A-law \( PT=8 \) Static
- G.711 μ-law \( PT=0 \) Static
- G.729, G.729a \( PT=18 \) Static
- G.729b, G.729ab \( PT=18 \) Static. Optional parameter “annexb” may be used according to RFC 4855

**Packetisation period for other (optional) Narrow Band codecs:**

- for AMR-NB the packetisation period shall be 20 ms.

**Payload type definition for other Narrow Band codecs:**

- AMR-NB \( PT=\)Dynamic as defined in RFC 4867 [33]

**WB Codecs engineering guidelines**

The requirements for AMR-WB are taken from GSMA PRD IR.36 [18] and RFC 4867 [33]. The requirements for G.722 are taken from New Generation Dect-ETSI TS 102 527-1; New Generation DECT, Part 1 Wideband Speech.

AMR-WB can operate in a 9 modes at source codec bit rate of 23.85 kbit/s, 23.05 kbit/s, 18.25 kbit/s, 15.85 kbit/s, 14.25 kbit/s, 12.65 kbit/s, 8.85 kbit/s, 6.60 kbit/s. The AMR-WB configurations specified for 2G and 3G are:

WB-Set 0 = 12.65 8.85 6.60
WB-Set 2 = 14.25 12.65 8.85 6.60
WB-Set 4 = 23.85 12.65 8.85 6.60

No other combination of the 9 AMR-WB modes is allowed for voice telephony. The other modes of AMR-WB may be used for other applications.

All these 3 supported configurations are TrFO compatible. However, WB-Set 0 is the guaranteed minimum common denominator mandatory for all configurations and shall be supported. This configuration also includes DTX, i.e. WB-SID frames and no data transmission during inactive speech; support of SID frames in reception is mandatory; generation is optional. All other modes are optional. G.722 shall be supported at a bit rate of 64 kbit/s.

**Packetisation period for mandatory Wideband codecs**

- for G.722, packetisation period shall be 20 ms
- for AMR-WB, packetisation period shall be 20 ms

**Payload type definition for mandatory Wideband codecs**

- G.722 \( PT=9 \) Static
- AMR-WB Dynamic as defined in RFC 4867
ANNEX 2 - QoS parameter definitions

**Parameters relevant to the transport layer**

The below parameters are defined for the quality of service control and monitoring at the transport layer for international voice service.

*Round Trip Delay*

Round Trip Delay is defined as the time it takes for a packet to go from one point to another and return

*Jitter*

Jitter is the absolute value of differences between the delays of consecutive packets

*Packet loss*

Packet loss is the ratio between the total lost packets and the total sent packets over a given time period.

**Parameters relevant to the service layer**

The below parameters are defined for the quality of service control and monitoring at the service layer for international voice service.

*MOS / R-factor for voice calls*

MOS (Mean Opinion Score) is a subjective parameter defined in ITU-T Rec. P.10 [34] as follows

“The mean of opinion scores, i.e., of the values on a predefined scale that subjects assign to their opinion of the performance of the telephone transmission system used either for conversation or for listening to spoken material.”

ITU-T Rec. G.107 [35] defines an objective transmission rating model (the E-model) for representing voice quality as an R-Factor, accounting for transmission impairments including lost packets, delay impairments and codecs. The impairment factors of the E-model are additive, thus impairments from different network segments may be added to obtain an end-to-end value.

With regards to usage of Wideband codecs, the ITU-T G.107.1 Wideband E-model: [36] should be referred to for voice service planning purposes.

The R-Factor may be converted into an estimated MOS which is called MOS Communication Quality Estimated or MOSCQE (as defined in ITU-T Rec. P.10 [34]) using formula in ITU-T Rec G 107 Annex B [37]. As a result, MOS is thus an actual user opinion score, and all measurements done by equipment (including R-Factor and MOSCQE) are estimates, and may differ from what actual customers would perceive.

**ALOC**

Average Length of Conversation (ALOC) expresses the average time in seconds of conversations for all the calls successfully setup in a given period of time. In a TDM environment ALOC has been defined in ITU-T Recommendation E.437 [38]:

\[
\text{ALOC} = \frac{\text{Time periods between sending answer and release messages}}{\text{Total number of answers}}
\]

In a Voice over IMS environment, and for the purpose of this document, ALOC is defined as follows:

- SIP protocol: ALOC is measured from the time of SIP 200 OK (in response to an INVITE initiating a dialog) to the time of call release (SIP BYE).

ALOC depends on user behavior.

**ASR**

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Answer Seizure Ratio (ASR) expresses the ratio of the number of calls effectively answered in a given period of time against the number of call session requests in that time. In a TDM environment, ASR has been defined in ITU-T Rec. E.411 [39] with the following formula:

\[
\text{Seizures resulting in answer signal} = \frac{\text{ASR}}{\text{Total Seizures}}
\]

In a Voice over IMS environment, and for the purpose of this document, ASR is defined as follows:

- SIP protocol: ASR is the ratio between the number of received 200 OK (in response to an INVITE initiating a dialog) and the number of sent INVITE initiating a dialog.

NER

Network Efficiency Ratio (NER) expresses the ability of a network to deliver a call without taking into account user interferences (measure of network performance) in a given period of time. In a TDM environment, NER has been defined in ITU-T E.425 [40] released in 2002 with the following formula:

\[
\text{Answer message or user failure} = \frac{\text{NER}}{\text{Total Seizures}}
\]

Note: user failure includes caller abandonment.

In a Voice over IMS environment, and for the purpose of this document, NER is defined as follows:

- SIP protocol: NER is the ratio of the number of received responses amongst the following responses, with the number of sent INVITE initiating a dialog:
  - a response 200 OK to an initial INVITE or
  - a BYE response or
  - a 3xx response or
  - a 404, 406, 410, 433, 480, 483, 484, 485, 486 or 488 response or Note that 403 is not included because it is categorized as both Network and User events and 403 is not sent to international networks
  - a 600, 603 or 606 response
  - a CANCEL message (in forward direction, i.e., from the calling party)

PGRD

Post Gateway Ramping Delay (PGRD) expresses the time elapsed between a request for a call setup and the alerting signal for that call. In a VoIP environment, and for the purpose of this document, PGRD is defined as follows:

The PGRD is the elapsed time after INVITE till media is available to the remote device. It can be calculated with the average time between sending an INVITE initiating a dialog and the first received message of the following SIP Responses:

- 180 resulting in local ringing at the remote device.
- The 200 OK to the initial INVITE without preceding 180 or 183, resulting in the call/session being answered.
- 183 with SDP and if there is no 180, resulting in media being available from the far end to the remote device. The media from the far end to the remote device will typically be ringing, but there are scenarios where the media would be either a tone or an announcement.

Note: only INVITEs initiating a dialog for which an alerting response is received are taken into account.
Service availability

Service Availability is defined as the percentage of the time the IPX Provider’s network is available for the VoIMS service. This encompasses the routers’ availability as well as all service platforms involved in the VoIMS service delivery. Assuming the access link between the SP and the IPX Provider is under the responsibility of the IPX Provider, the availability of this link maybe part of the overall service availability.