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IMS-Based Services:
Network – Network Interface Definition
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This document provides the i3 forum`s perspective on the definition of the Network-to-Network Interface (NNI) between multiservice IMS-based platforms focusing on voice and video services originated from fixed as well as mobile networks. The scope covers both the Network-to-Network Interface between a Service Provider and Int. Carrier and between two Int. 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

Over the last 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 the implementation of IMS-based services at the international level.

This process started in Asia and in the USA, and more recently also in Europe, aiming to the offering of Voice over LTE (VoLTE)/Video over LTE (ViLTE) services with HD capabilities.

In the wake of this trend, i3 forum has considered a priority to deliver a set of documents devoted to describing the architectures, the interfaces, the protocols to be adopted for the support of International IMS (IP Multimedia Subsystem) services between two IMS Service Providers or between an IMS Service Provider and non IMS Service Provider adopting, in compliance with previous deliverables, an IPX model at the transport level.

In this framework this i3 forum deliverable focuses on the definition of the Network-to-Network Interface between two International Carriers (i.e. IPX Providers) or between an FNO/MNO (i.e. IMS Service Provider) and an International Carrier addressing:

1) the identification of the most common transport layer interfaces together with the endorsement of the recognised worldwide standards for signalling protocols and QoS/DSCP traffic classification;
2) a list of recommended codecs for voice and video services and a discussion of the most common security actions at the transport layer.

The document is complemented with three Annexes which aim to facilitate and reduced the implementation time of IMS NNI devoted to:

3) the engineering guidelines for audio codecs;
4) some alternatives for service configuration at NNI – considering IMS and legacy services – to be assumed as “*best implementation practices*”;
5) interconnection forms for four services Voice over IP, Voice over IMS, ViLTE and Diameter Signalling in order to provider a “*track*” to be followed by the interconnecting parties.

The final objective of the document, together with a companion i3 forum document devoted to IMS service interoperability, is to provide a unique analysis of the impact on Carriers’ / IPX Providers’ platforms of the provisioning of IMS-based services. 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.

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1. Scope and objective of the document

Over the last 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 deployments in Asia and in the USA, and more recently also in Europe, of Voice over LTE (VoLTE)/Video over LTE (ViLTE) services with HD capabilities.

In the wake of this trend, i3 forum has considered a priority to deliver a set of documents devoted to describing the architectures, the interfaces, the protocols to be adopted for the support of International IMS (IP Multimedia Subsystem) services between two IMS Service Providers or between an IMS Service Provider and non IMS Service Provider adopting, in compliance with previous deliverables, an IPX model at the transport level.

In this framework this i3 forum deliverable focuses on the definition of the Network-to-Network Interface between two International Carriers (i.e. IPX Providers) or between an FNO/MNO (i.e. IMS Service Provider) and an International Carrier addressing:

1) the identification of the most common transport layer interfaces (Sec. 4);
2) the endorsement of 3GPP specification for signalling protocols (Sec. 5) and of GSMA QoS/DSCP traffic classification (Sec. 7) as well as the recommended codecs for voice and video services (sec. 6);

The document is complemented with three Annexes which aim to facilitate and reduced the implementation time of IMS NNI devoted to:

5) the engineering guidelines for audio codecs;
6) some alternatives for service configuration at NNI – considering IMS and legacy services – to be assumed as "*best implementation practices*";
7) interconnection forms for four services Voice over IP, Voice over IMS, ViLTE and Diameter Signalling in order to provider a "*track*" to be followed by the interconnecting parties.

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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. 4 onwards, the terminology IPX Provider is always used for identifying an International Carrier.

2. Symbols and Acronyms

3GPP	3rd Generation Partnership Project
AMR-NB	Adaptive Multi-Rate Narrow Band
AMR-WB	Adaptive Multi-Rate Wide Band
BGCF	Breakout Gateway Control Function
BSS	Business Support System
CSCF	Call Switching Control Function
Diffserv	Differentiated Services
DNS	Domain Name Service
DSCP	Differentiated Services Code Point
DTMF	Dual-Tone Multi-Frequency
DTX	Discontinuous Transmission
DWDM	Dense Wavelength Division Multiplexing
ENUM	E.164 NUmber Mapping
EPC	Evolved Packet Core
ETSI	European Telecommunications Standards Institute
FTTx	Fiber To The “x” (n=network, c=curb, b=building, h=home)
GPRS	General Packet Radio Service
GRX	GPRS Roaming eXchange
GSM	Global System for Mobile Communications
HD	High Definition
HSS	Home Subscriber Service
IBCF	Interconnection Border Control Function
Ici	Reference Point between an IBCF and another IBCF belonging to a different IM CN subsystem network
I-CSCF	Interrogating CSCF
IEEE	Institute of Electrical and Electronic Engineers
IETF	Internet Engineering Task Force
II-NNI	Inter-IMS NNI
IMS	IP Multimedia Subsystem
IPLC	International Private Leased Circuit
IPSec	IP Security
IPv4 / v6	Internet Protocol version 4 / version 6
IPX	IP eXchange
ISUP	ISDN User Part
ITU	International Telecommunications Union
Izi	Reference Point between a TrGW and another TrGW or media handling node belonging to a different IM CN subsystem network
LBO	Local Break Out
LBO	Long Term Evolution
MNO	Mobile Network Operator
MPLS	Multi Protocol Label Switching
NAT	Network Address Translation

NB	Narrowband
NFV	Network Function Virtualisation
NGN	Next Generation Network
NNI	Network to Network Interface
OSS	Operations Support System
PAT	Port Address Translation
PCRF	Policy and Charging Rules Function
P-CSCF	Proxy-CSCF
PDH	Plesiochronous Digital Hierarchy
PHB	Per-Hop Behavior
PLMN	Public Land Mobile Network
PSTN	Public Switched Telephone Network
PT	Payload Type
QCI	Quality Coded Indicator
QoS	Quality of Service
RCS	Rich Communication Suite
RFC	Request For Comments
RTC	Real Time Communication
RTP	Real-Time Protocol
S8HR	S8 Home Routing
SBC	Session Border Controller
S-CSCF	Serving – CSCF
SD	Standard Definition
SDH	Synchronous Digital Hierarchy
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SIP URI	SIP protocol URI
SMS	Short Message System
SP	Service Provider
TCP	Transmission Control Protocol
tel-URI	Telephone URI
TRF	Transit and Roaming Function
TrFO	Transcoder Free Operation
TrGW	Transition Gateway
UDP	User Datagram Protocol
UMTS	Universal Mobile Telecommunications System
URI	Uniform Resource Identifier
ViLTE	Video over LTE
VLAN	Virtual Local Area Network
VoLTE	Voice over LTE
VPN	Virtual Private Network
WebRTC	Web Real Time Communication

3. References

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[27] ANSI T1.105 – 1995 American National Standard for Telecommunications, Synchronous Optical Network (SONET)

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[29] ITU-T Rec. G.671; Transmission characteristics of optical components and subsystems

[30] IETF, „RFC 1918; Address Allocation for Private Internets" .

[31] IETF, „RFC 4193, Unique Local IPv6 Unicast Addresses"

[32] IETF, „RFC 3550; RTP: A Transport Protocol for Real-Time Applications".

[33] IETF, „RFC 4855; Media Type Registration of RTP Payload Formats".

[34] IETF, „RFC 6733; Diameter Base Protocol".

[35] IETF, „RFC 3264; An Offer/Answer Model with the Session Description Protocol (SDP), (June 2002)".

4. Transport layer

4.1 Physical interconnection alternatives

The physical interface of the interconnection can be either SDH POS – based, Ethernet-based (i.e. fast-Ethernet, gigabit-Ethernet or 10 gigabit-Ethernet) or DWDM-based.

4.1.1 SDH-based transport systems

The ITU-T Recommendations G. Series shall be considered as reference documents: ITU T Rec. G.707. For North America another reference document is ANSI T1.105.

4.1.2 Ethernet-based transport systems

The IEEE recommendations 802.3 for Ethernet communication together with enhanced Ethernet technologies such as fast-Ethernet, gigabit-Ethernet and 10 gigabit-Ethernet have to be considered (e.g. ISO/CIE 8802-3). This includes MEF standards for Carrier Ethernet connections.

4.1.3 DWDM-based transport systems

For the public interconnection configurations, a DWDM channel can be provisioned for interconnecting two carries. The ITU-T Recommendations G. Series shall be considered as reference documents: ITU-T Rec. G.671 [29]

4.2 Interconnection redundancy

The level of redundancy of a specific interconnection can be enhanced by increasing the number of involved Border Functions, by increasing the number of involved PE routers using geographical separation or by increasing the number of diverse network links involved.

4.3 Interconnection Points

IPX Providers can implement both direct (i.e. bilateral) interconnections and shared (i.e. multilateral) interconnections. A shared multilateral interconnection can be implemented in private and/or public locations where IPX Providers can meet.

The private locations would be those set up by a group of IPX Providers and the public ones will be those created by a third party with open access to IPX Providers. The GSMA has identified/set-up three public locations (Peering Points) in AMS-IX Amsterdam, Equinix Ashburn and Equinix Singapore for IPX services.

4.4 Internet Protocol Versions

Bilateral international VoIP interconnections may occur using either IPv4 or IPv6 network protocols; in the context of this document IP refers to both IPv4 and IPv6 addressing versions. IPv4 refers to the commonly deployed protocol version using 32 bit addressing and IPv6 to the protocol version using 128 bit addressing.

The introduction of the IPv6 addresses, partitions the Public Internet into two separate networks, the IPv4 Public Internet and the IPv6 Public Internet. In the scope of bilateral international interconnections, the introduction of this addressing scheme requires carriers to be capable of managing both schemes for private as well as public interconnections.

Private addresses discussed in this section refer to either RFC 1918 [30] addresses for IPv4 or RFC 4193 **Error! Reference source not found.** for IPv6.

5. IMS Signalling Options

5.1 Diameter Signalling

For Diameter Signalling (see RFC 6733 [**34**]) related to IMS services, i3forum recommends no changes to what is already specified in "*LTE Data Roaming over IPX Service Schedule*"[7]. Specifically, the principles and methods related to Diameter signaling accounting, policing and charging have to be maintained.

5.2 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.

i3forum considers this type of traffic as chargeable traffic, which represents a component of the total traffic exchanged between the visited network and the home network for roaming customers. This applies for the signalling messages sent by both home and visited network (or an in-between IPX provider).

5.3 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 fulfils 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 [8]. SIP provides the primitives to be used for implementing different services, and specifies the signalling 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 [9] addresses the Inter-IMS Network to Network Interface (II-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 signalling as recommended in 3GPP TS 29.165 [9], 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 (II-NNI). A Carrier / IPX Provider has to manage SIP signalling for session based services for terminating a call between two Service Providers without any roaming party and for managing a roaming call. For both cases i3 forum endorsed the recommendations in the document "Interconnection & Roaming IMS Signaling Profile Release 3 (May 2016) [10]. The document provides an operational specification of the 3GPP document, whereby each table entry describing the SIP IMS Profile, has been analysed as either being endorsed, modified or marked as not applicable for an international voice service.

5.3.1 Consistency with GSMA IR.95

In PRD IR.95 [11] 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, SMSoIP and RCS services as described in the relevant PRDs. In the document i3F – Interconnection & Roaming IMS Signalling Profile Rel. 3 [10] the i3 Forum has defined a minimum profile for the basic (i.e. excluding supplementary services) voice and video service, SMS and RCS.

Both profiles are based on 3GPP TS 29.165 Rel. 11 [9] 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 7 of this document);
- For routing of outgoing SIP sessions over the II-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 8 of this document)

i3 forum intends to endorse the SIP/SDP profile as defined in the GSMA IR.95 document [11],however it is recognized that small discrepancies could exist which will be reviewed in the future.

6. Codecs

Many different coding schemes have been defined, implemented and used for international voice and video services. 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 [12] for both mobile and fixed access to the IMS core platform and GSMA confirmed this specification with regard to VoLTE (IR. 92 [13]) and Video over LTE (IR.94 [14]).

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.

6.1 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 4 below and shall allow the negotiation of these codecs between both originating and terminating Service Providers.

Group 1. Mandatory Narrow band codecs	**Group 2. Optional Narrow band codecs**
G.711 A-law, µ-law 64 kbit/s (Mandatory in network for interworking between IMS networks and other IMS or non IMS networks)	AMR-NB (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)
G.729, G.729a, G.729b, G.729ab 8kbit/s (For interworking with existing VoIP networks and support of existing VoIP terminals)	

Figure 4 - 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 Annex 1 of this document and in the i3 forum deliverable "*Technical Interconnection Model for International Voice Services", Release 6, May 2014"*[15].

6.2 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:

Group 1. Mandatory Wideband codecs	Group 2. Optional Wideband codecs
G.722 (generally used by fixed network operators) (Mandatory in network for interworking between IMS networks and other IMS or non IMS networks)	EVS (Enhanced Voice Services) codec
G.722.2 (AMR-WB, generally used by mobile network operators) (Mandatory in wideband terminals using 3GPP access to the IMS and in IMS Media Gateways supporting wideband) (Mandatory for VoLTE in GSMA IR.92 [13])	

Figure 5 - 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 5 are mandatory.

Specific engineering guidelines for the use of these codecs are given in Annex 1 of this document and in the i3 forum deliverable "*Technical Interconnection Model for International Voice Services" [15].* Additional information about wideband voice codecs and their usages can be found in i3 Forum white paper "*Enabling HD Voice continuity in International calls"* [16]

With regard to superwideband/full band codecs for voice, it is worth mentioning that the codec EVS (Enhanced Voice Services), has been standardized by 3GPP to provide Enhanced Voice Services over IMS [17], and has been included in GSMA VoLTE IR.92 [13] as a mandatory codec if a superwideband / full band voice service is offered. First EVS trials have been carried out in 2016.

6.3 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.

6.4 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 is an evolution of the SILK voice 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

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)

6.5 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* [13], [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.

6.6 Codecs for Fax and DTMF transmission

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

DTMF: for transmitting DTMF digits, [IETF RFC 4733 [19]]] shall be used as specified in Annex G of 3GPP TS 26.114 [12].

6.7 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 [13] and IR.95 [11].

6.8 Video codecs

6.8.1 Video codecs for display resolutions up to SD

The below table provides the mandatory and the optional video codec for IMS services in case such a service is presented on a display that supports a maximum number of 576 lines.

Group 1. Mandatory Video codecs (note)	Group 2. Optional Video codecs
H.264 [20] (Recommended by 3GPP for Multimedia Telephony Service over IMS [12])	H.265 (HEVC) [21] (Recommended by 3GPP for Multimedia Telephony Service over IMS [12]}

Figure 6 - Mandatory and Optional Video Codecs

Note: The mandatory configuration to be supported is the Constrained Baseline Profile (CBP) Level 1.2 as specified in section 3.3 of IR.94 [14] and support of level 3.1 is recommended as specified in [12].Examples for frame sizes and their assigned maximum frame rates are given in Table A-6 of the H.264 specification [20].

The granularity of profiles and levels of the optional H.265 (HEVC) codec is not the same as for the H.264 codec. In order to cover those frame sizes and frame rates as supported by the H.264 codec the following shall apply; when the H.265 codec is supported, it is recommended to support the Main Profile Level 3.1 as specified in [12]. Examples for frame sizes and their assigned maximum frame rates are given in Table A.7 of the H.265 specification [21].

It should be noted that there are other optional codecs, e.g. VP8, which is commonly used in case of WebRTC services. However the H.264 codec has recently been specified by IETF as mandatory codec to be implemented in WebRTC clients (browsers and non-browsers) in addition to the VP8 codec [28] Therefore support of VP8 is not needed for interoperability with WebRTC video services and remains optional for a Carrier / IPX Provider, unless the vast majority of the market solutions adopts this codec.

6.8.2 Video codecs for display resolutions up to HD

The table below provides the mandatory and the optional video codec for IMS services in case such a service is presented on a display that supports a maximum of 1088 lines.

Group 1. Mandatory Video codecs (note)	Group 2. Optional Video codecs
H.264 [20]	H.265 (HEVC) [21]

Figure 7 - Mandatory and Optional Video Codecs

Note: The mandatory configuration to be supported is the Main Profile Level 4.2. Examples for frame sizes and their assigned maximum frame rates are given in Table A-6 of the H.264 specification [20].

The granularity of profiles and levels of the optional H.265 (HEVC) codec is not the same as for the H.264 codec. In order to cover those frame sizes and frame rates as supported by the H.264 codec the following shall apply. When the H.265 codec is supported, it is recommended to support the Main Profile Level 4.1. Examples for frame sizes and their assigned maximum frame rates are given in Table A.7 of the H.265 specification [21].

7. QOS / DSCP Traffic Classifications

Following GSMA IR.34 [4], to ensure consistent QoS parameters across the network, *MNOs (Service Providers) have to mark their packets to the correct traffic classes in accordance with Table 1 below. Header information should remain unchanged over the NNI (and thus end-to-end), unless agreed otherwise between Service Provider and IPX Provider.*

If the DSCP marking from the Service Provider cannot be trusted, the IPX Provider has to correct the marking to a pre-agreed default value before sending the packets to the correct service VLAN. On the NNI between IPX Providers, it is the responsibility of the sending IPX Provider to ensure that the DSCP marking can be trusted.

EPS QoS	GPRS/UMTS QoS Parameters			IP Transport		IPX QoS	Ethernet Transport	
QCI	Traffic Class	THP	Signalling indication	Diffserv PHB	DSCP	Traffic Class	CoS	Binary
1	Conversational	N/A	N/A	EF	101110	Conversational	5	101
2								
3								
4	Streaming	N/A	N/A	AF41	100010	Streaming	4	100
5	Interactive	1	Yes (see note)	AF31	011010	Interactive	3	011
6			No	AF32	011100		3	011
7		2	No	AF21	010010		2	010
8		3	No	AF11	001010		1	001
9	Background	N/A	N/A	BE	000000	Background	0	000

Table 1 – QoS information and their mapping to CoS & DSCP values

8. Annex 1 - Codecs and 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 [33]

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 [25]

WB Codecs engineering guidelines

The requirements for AMR-WB are taken from GSMA PRD IR.36 [18] and RFC 4867 [25]. 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 kbt/s, 18.25 kbit/s, 15.85 kbit/s, 14.25 kbit/s, 12.65 kbt/s, 8.85 kbt/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 = {15.85 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 [25]

9. Annex 2 – Services' Configuration at NNI

On the basis of surveys carried out among International Carriers as well as results from GSMA studies, it has been recognized that the implementation time of IPX interconnection can last some months implying an unacceptable waste of time and resources.

One of the issue which takes time to be finalized is the service's configuration at the NNI and this issue gets even more relevance in an IMS environment due to the large set of services which can be transported over IMS.

i3 forum considers useful to propose a limited set of alternative solutions for services' configurations at NNI not aiming to a unique international standard but to identify recognized best practices making easier and quicker the convergence of the interconnecting parties towards a unique service' configuration map.

The final target is to agree, jointly with GSMA, a reasonable set of configurations, tentatively two or max three. These configurations should be considered strongly recommended at the interface between two Carriers (i.e. IPX Providers); whereas they should be considered as suggested configurations at the interface between a MNO (i.e. Service Provider) and a Carrier (i.e. IPX Provider).

9.1 Criteria for selecting the target configurations

Assuming a correct NNI design which guarantees QoS requirements and correct charging for all listed services; the selection criteria have to meet the basic requirements of:

i. minimizing the cost of implementation/configuration, in terms of man power;
ii. minimizing the time of implementation/configuration of the services
iii. maximizing the level of security.

The following criteria have been identified for selecting the target configurations (no hierarchical order is assumed):

1) Operational efficiency, in terms of configuration time and use of resources;
2) Technical capabilities of SPs to separate services;
3) Different network equipment managing the media and signalling flows;
4) Different charging schemes among the listed services, both between SPs as well as between SP and IPX P
5) Different addressing and routing schemes to be applied to the listed services;
6) Different QoS requirements among the listed services (it is considered equivalent the adoption of Traffic Classes or Quality Class Identifiers for specifying the quality requirements of various services);
7) Specific Fault Management requirements;
8) Allocation of legacy services (e.g. Sigtran, GRX, VoIP).
9) Financial value of listed service.

9.2 Service bundles

In the following a proposed services' configuration is given making bundles of services considering the criteria listed above:

a) **Voice services (CS based -> VoIP and Voice over IMS) including the related signalling information:** for voice services two possible configurations are suggested being this service is too «sensitive» and it is unavoidable to follow several options.

   The first configuration considers the aggregation of legacy VoIP and VoIMS into one VLAN; the second configuration considers the separation of the legacy VoIP and the VoIMS into two different VLANs. Rationale: the "*aggregation*" is considered since the same equipment and routing system are used together with the same charging criteria, the "*separation*" is considered due to the high level of sensitiveness of the service and when IMS platform is regarded as a separate platform.

It is worth outlining that in the above case signalling is always associated with the related media flow. More specifically for VoIP, SIP signalling is configured with the same VLAN of the VoIP media flow. For Voice over IMS, the 3 signalling types:

i. SIP signalling for IMS registration;
ii. SIP IMS Signalling for an interconnecting call;
iii. SIP IMS Signalling for a roaming call.

are always configured in the same VLAN than the IMS media flow.

b) **Videocall over IMS/LTE:** "*aggregated"* (i.e. in the same VLAN) with IMS Voice. Rationale: the two services share the same network equipment and routing system and, in principle, there is no difference between Voice (1 media stream) and Video (2 media streams including voice).

c) **Data Services (GRX, LTE Data and S8HR):** "*aggregated"* in the same VLAN: Rationale: traffic already received by a Carrier/IPX Provider aggregated and it shares the same network equipment and routing system.

With regard to the agreed specification of VoLTE roaming configuration based on the usage of data transport exploiting the S8 interface, since the Carrier (IPX Provider) will receive this traffic aggregated with the Data Services, it is unavoidable to "aggregate" in the same VLAN of Data Service also this S8 HR traffic. This aggregation implies that GRX,LTE Data and S8HR packets – though all encapsulated in GTP tunnel - all need to be tagged by the Mobile Operator with the right class of service.

Operationally this traffic is managed as an IPX Transport service.

d) **Sigtran and Diameter Signalling:** "*aggregated"* in the same VLAN or "*separated*" in two VLANs. Rationale: the two options are equivalent and it is up to parties to agree a the most suitable solution.

e) **Leased Line equivalent:** in a separate VLAN. Rationale: being service unaware, it is desirable to confine it in a separated unique logical channel. In addition, this service is intrinsically different from (GRX, LTE Data, S8HR) and has to be separated in a different VLAN.

Operationally this traffic is managed as an IPX Transport service.

f) **RCS (IM, video share, presence without voice and video services):** analysis in progress. No clear position for the time being; i3 forum welcomes GSMA guidelines for RCS interworking charging schemes.

The following tables summarises the guidelines given above:

- Configuration set "A" which requires the minimum number of VLANs (4 + RCS) aggregating both voice services and signalling services
- Configuration set "B" which requires an intermediate number of VLANs (5 + RCS) separating either voice services (B1) or signalling services (B2)
- Configuration set "C" which requires the maximum number of VLANs (6 + RCS) separating both voice services and signalling services

Configuration Set "A" (minimum VLAN number)	Configuration Set "B1" (Intermediate VLAN number)	Configuration Set "B2" (Intermediate VLAN number)	Configuration Set "C" (maximum VLAN number)
▪ **1 VLAN for VoIP+VoIMS +Video oIMS** (it includes signalling)	▪ **1 VLAN for VoIP** (it includes related signalling) ▪ **1 VLAN for VoIMS +Video oIMS** (it includes related signalling)	▪ **1 VLAN for VoIP+VoIMS +Video oIMS** (it includes signalling)	▪ **1 VLAN for** VoIP (it includes related signalling) ▪ **1 VLAN for VoIMS +Video oIMS** (it includes related signalling)
▪ **1 VLAN for (GRX+ LTE Data + S8HR)** (via an IPX Transport service)	▪ **1 VLAN for (GRX+ LTE Data + S8HR)** (via an IPX Transport service)	▪ **1 VLAN for (GRX+ LTE Data + S8HR)** (via an IPX Transport service)	▪ **1 VLAN for (GRX+ LTE Data + S8HR)** (via an IPX Transport service)
▪ **1 VLAN for (Sigtran+ Diameter)**	▪ **1 VLAN for (Sigtran+ Diameter)**	▪ **1 VLAN for Sigtran** ▪ **1 VLAN for Diameter**	▪ **1 VLAN for Sigtran** ▪ **1 VLAN for Diameter**
▪ **1 VLAN for LL equivalent (L2 / L3)** (via an IPX Transport service)	▪ **1 VLAN for LL equivalent (L2 / L3)** (via an IPX Transport service)	▪ **1 VLAN for LL equivalent (L2 / L3)** (via an IPX Transport service)	▪ **1 VLAN for LL equivalent (L2 / L3)** (via an IPX Transport service)
▪ **RCS (all services): further analysis needed**	▪ **RCS (all services): further analysis needed**	▪ **RCS (all services): further analysis needed**	▪ **RCS (all services): further analysis needed**
4 VLANs+RCS	**5 VLANs+RCS**	**5 VLANs+RCS**	**6 VLANs+RCS**

Table 2 – Proposed Service configuration maps

10. Annex 3 – IMS Interconnection forms

In the attached .xls file IMS interconnection forms are proposed for the following services:

1) Voice over IP form;
2) VoLTE/ViLTE form for Interworking and LBO roaming;
3) VoLTE/ViLTE form for S8HR;
4) Diameter Signalling form