INTERNATIONAL INTERCONNECT FORUM FOR SERVICES OVER IP

i3 FORUM

IP international interconnections for bilateral Voice services
(V.1.0) May 2008

Scope

This document is the first sub-set of deliverables of the i3 forum Services and Requirements work stream. It covers the migration from TDM to IP of the international bilateral voice interconnections. Works and recommendations for other international services to be carried over IP will be addressed in subsequent documents.

Table of Content

1 Introduction

The i3 forum is formed by eight international wholesale carriers namely AT&T, Deutsche Telekom, France Telecom Orange, Telefonica International Wholesale Services, Telecom Italia Sparkle, Telecom Poland SA, TeliaSonera and SingTel who share the following characteristics;

- Transport over 40% of the entire international voice market
- Represent a strong retail mobile and PSTN base (over 1 billion customers in more than 80 countries in aggregate) complemented by a significant and growing retail VoIP customer base.
- Have extensive experience in providing voice over IP international interconnections solutions for several years.
- Actively participate to the standardization bodies for new generation networks.
- Engaged to the evolution of their domestic networks towards IP and NGN solutions, and already providing leading edge convergent services.

The i3 forum gathers a unique market experience that includes

- Managing thousands of international interconnections allowing for a global view on a wide range of needs, specificities, challenges and opportunities of different mobile and fixed carriers across the world.
- Providing termination to destinations with a broad range of network infrastructures, from the best in class networks to the more difficult to reach countries with limited and lower quality network infrastructures.
- Dealing with carriers with a different level of maturity about VoIP technology and readiness to migrate.

Over the past few years, carriers have experienced the opportunities of a migration to IP interconnections, but have also been dealing with the lack of a clear and commonly accepted path in the industry to implement and to migrate to international voice over IP interconnections.

The members of the i3 forum leveraged their experience and expertise and propose to the industry a set of comprehensive recommendations on how to implement and to foster the move to international voice over IP interconnections. These recommendations take into account different market needs, the business models being used, the diversity of the networks available, the existing standards, the requirements for operations and billing functions, the upcoming IP technology and services evolutions and a clear commitment to keep providing solutions that guarantee the highest quality.

1.1 Business model

In order to permit a worldwide and unrestrained migration to IP of the thousands of existing TDM International voice bilaterals, the existing business model in TDM shall remain fully applicable with voice over IP interconnections. Such functions include but are not limited to accounting, billing and customer service. The existing and widely implemented business model for international voice bilateral interconnections is based on calling network's party pays and settlement regime. This model will remain applicable over IP, its implementation will be decided by the carriers within the bilateral negotiations.

International interconnections over IP will enable the implementation of existing but also new voice services, and new products. While the existing business model shall remain possible, other business principles may become pertinent as well. At the time of printing this document discussions on new business principles are not completed and if appropriate will be part of a subsequent document.

2 Document Purpose

The purpose of this document is to provide a detailed list of requirements for the international voice services (encompassing fax and modem connections) that are to be exchanged over IP between international **carriers**. These services should include (but may not be limited to) the voice services currently exchanged over TDM.
This document should serve as a basis for the implementation of the adequate technical solutions. The recommendations should be usable not only by members of the forum, but also by any carrier outside of the forum to migrate existing voice, fax and data **bilaterals** interconnections to voice over IP.

The migration from a current legacy voice interconnection to voice over IP shall target to maintain or exceed the quality levels and service availability currently in place on TDM interconnections.

The recommendations and forum propositions should also contribute to the reduction of the provisioning and service delivery timeframes which should be at least identical and ultimately faster than those on TDM.

This document is to be read jointly with the two associated i3 forum technical documents "Interconnection model for bilateral voice services" [14] and "Interoperability test plan for bilateral voice services" [1].

3 Scope

The scope of this document is to cover **international voice bilaterals** only.

This deliverable is the first version of the document. Future versions will be released encompassing new features / capabilities to address the evolution of services, equipment capabilities and international standards.

4 Acronyms

ALOC	Average Length of Call
ASR	Answer Seizure Rate
CDR	Call Detail Record
CLI	Calling Line Identity
CLIR	Calling Line ID Restriction
CPN	Calling Party Number
DTMF	Dual-Tone Multi-Frequency
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
MOS	Mean Opinion Scale
NER	Network Efficiency Ratio
OCN	Original Called Number
OSS	Operations Support System
PGRD	Post Gateway Ringing Delay
PSTN	Public Switched Telephone Network
RTD	Round Trip Delay
SDH	Synchronous Digital Hierarchy
SIP	Session Initiation Protocol
SIP-I	SIP with encapsulated ISUP
SLA	Service Level Agreement
TDM	Time Division Multiplexing
VoIP	Voice over IP
VPN	Virtual Private Network

5 References

[1] I3 forum "Interoperability Test Plan for Bilateral Voice services" Version 1.0, May 2008
[2] I3 forum "Bilateral Voice Service Description" Version 1.0, May 2008
[3] ITU-T T.38 Procedures for real-time Group 3 facsimile communication over IP networks, 1998
[4] ITU-T Recommendation G.729 "Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP)", 1996
[5] ITU-T Recommendation G.711 "Pulse Code Modulation of Voice Frequencies", 1988
[6] ITU-T Recommendation T.38 "Procedures for real-time Group 3 facsimile communication over IP networks" (04/2007)
[7] ITU-T Recommendation G.711 "Pulse Code Modulation (PCM) of voice frequencies"
[8] ITU-T Recommendation G.729 "Coding of speech at 8 kbit/s using conjugate-structure algebraic code excited linear-prediction (CS-ALEP (03/96)
[9] ITU-T Recommendation G.729 Annex A "Reduced complexity 8kbit/s CS-ALEP codec" (11/96)
[10] ITU-T Recommendation G.729 Annex B Silence compression scheme for G.729 optimized for terminals conforming to Recommendation V.70" (11/96)
[11] ITU-T Recommendation G.729 Annex A and B
[12] ITU-T Recommendation G.107 "The E model, a computational model for use in transmission planning", March 2005
[13] ETSI EG 202 057-2 "Speech processing transmission and quality aspects (STQ); user related QoS parameter definitions and measurements; Part 2: Voice Telephony, Group 3 Fax, modem data services and SMS"; October 2005
[14] i3 forum "Interconnection model for bilateral Voice services" Version 1.0, May 2008
[15] ITU-T Recommendation E.105 "INTERNATIONAL TELEPHONE SERVICE", August 1992
[16] ITU-T Recommendation G.114 "One-way transmission time", May 2003

6 Bi-lateral service description

The bilateral voice international IP interconnection is provided by two carriers to transport and interconnect voice calls and services between domestic voice networks.

International bilateral voice interconnections are used to interconnect retail domestic networks (mobile or fixed). Bilateral interconnections in the TDM world provide the highest quality available for international voice calls, it relies on the best quality of transport available and it limits the number of international voice switching hops to the minimum (maximum of two international voice platforms/switches in most cases).
A well dimensioned and engineered bilateral will always provide the highest quality available in the market for a given destination. The quality of bilateral voice interconnections can vary between destinations reflecting uneven and limited technical infrastructures in some parts of the world.

The quality recommendations put forth in this document reflect a specific network and industry situation at the time that this document is published. These recommendations may be reviewed in future releases and may evolve to reflect the evolutions in technologies, networks and markets.

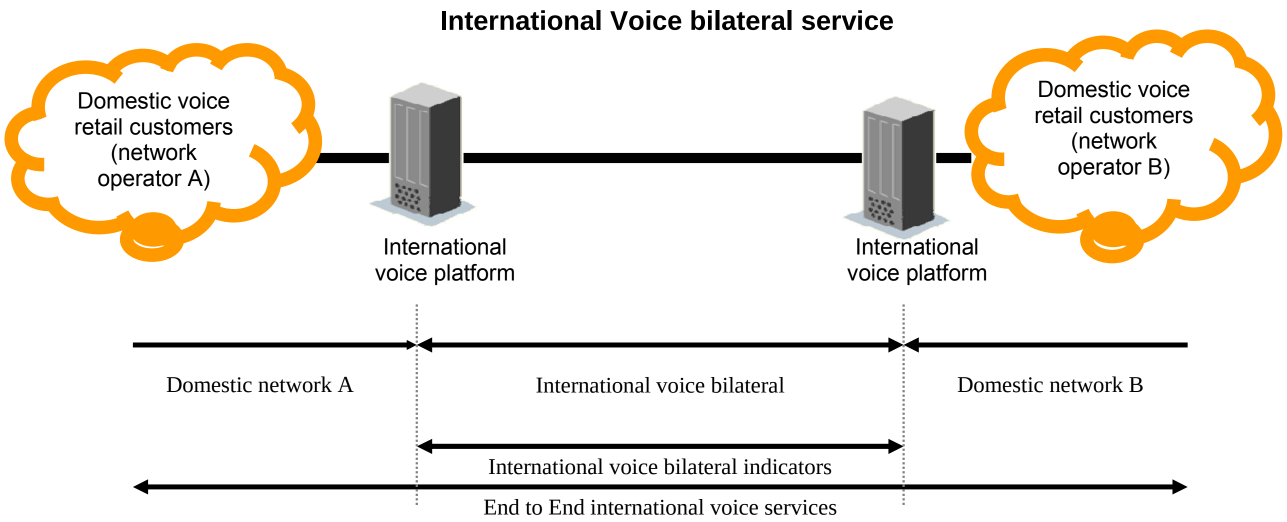

Figure 1 - International Voice bilateral service

International voice bilateral services are defined as the voice services between the international voice platforms of the two international carriers taking part in the bilateral interconnection, as illustrated above (figure 1).

The recommendations of this document are limited to the international part of an international voice call. The quality and characteristics of the domestic networks are out of scope.

The retail bases can be VoIP or TDM or both. Carriers can implement dedicated international IP connections for a specific type of voice traffic (mobile traffic only for instance), or use the same interconnection to transport any type of domestic retail traffic: mobile, fixe, TDM or VoIP. It is left to the bilateral negotiations between carriers to decide on which combination of voice traffic to carry through the international IP interconnection (figure 2).

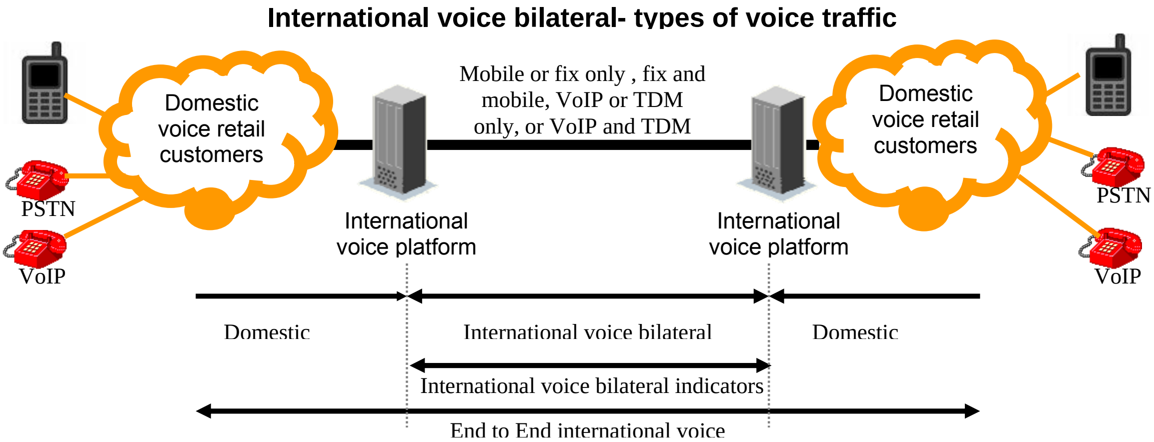

Figure 2 - International voice bilateral- types of voice traffic

6.1 Types of voice call

International voice interconnections are used to transport INTERNATIONAL TELEPHONE SERVICE as described per the ITU E.105 recommendation [15]

It is understood that international voice bilaterals over IP should transport all the international voice services (encompassing fax and data connections) currently available in TDM, including traditional and ISDN voice telephony, collect calling, international toll free traffic and home country direct. ISDN data services are not included in this first phase of the i3 forum recommendations and will be discussed and implemented over IP in a later phase.

6.2 Interconnection solutions

This section defines the main interconnections options, between two international voice networks, as a model for voice bilaterals over IP interconnections.

6.2.1 Via private and dedicated IP link

Carriers can establish IP bilateral interconnections implementing a dedicated IP connection directly between the two carriers (logically equivalent to their existing direct TDM bilateral interconnection). Dedicated IP connections or links provide transport with controlled and monitored levels of quality and security, allowing carriers to control the final voice quality characteristics and specifications matching those of TDM environment. This link is a private connection completely unknown from the Internet. Only voice packets are transmitted onto this link. IP networks A and B handle the VoIP traffic to ensure security and also voice prioritization versus other internet traffic in case of network congestion. The international voice bilateral interconnection and underlying dedicated link(s) should be engineered to provide all specifications of section 6.3 .

- Physical IP link implementation: The IP link can be transported via a PDH /Sonet/SDH circuit, it can be a local loop, or an international leased line according to the geographical distance between network A and network B. The IP link can also be carried over an Ethernet connection especially if both networks A and B are collocated in a carrier hotel (figure 3).

International voice bilateral service over a dedicated and private IP link.

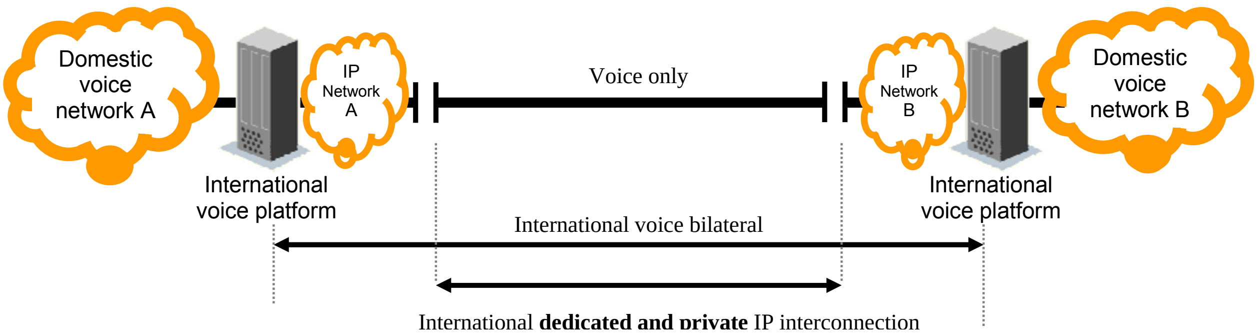

Figure 3 - International voice bilateral service over a dedicated and private IP link.

- Logical IP link implementation: The IP link can be implemented as a logical/virtual IP link using a third party international private IP network. Private IP networks provide transport, control, security (i.e isolated from the Internet) and service layers with controlled and monitored levels of quality and security (figure 4).

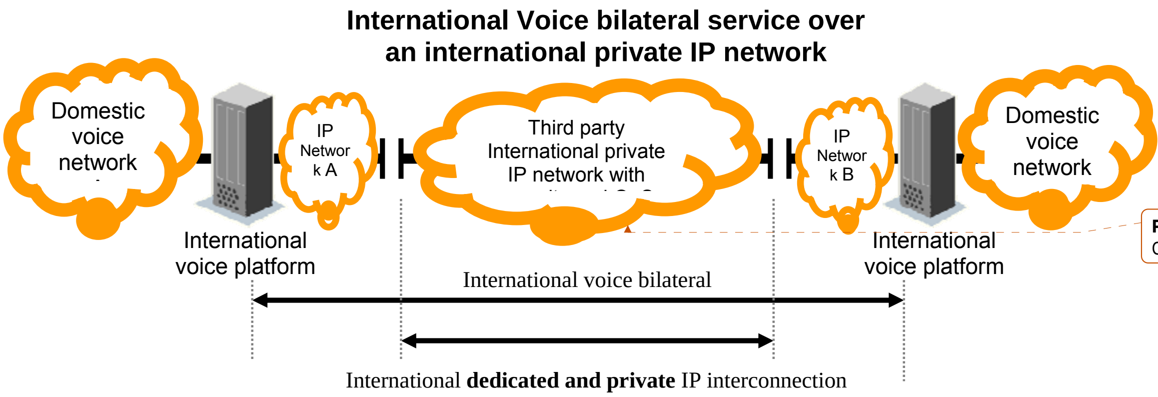

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Figure 4 - International Voice bilateral service over an international private IP network

6.2.2 Via Public IP networks

Carriers can establish an IP bilateral interconnection via public IP networks. Based on this interconnection scheme, a range of network and voice performance, reliability and reporting can be offered, depending on the type of public IP interconnection that is implemented. Two types of public IP interconnections are described, one that uses public network(s) controlled and managed by the carriers implementing the bilateral, and a second scheme that uses public networks that are not controlled nor managed by the carriers implementing the bilateral interconnection (i.e. using third party Internet networks).

6.2.2.1 Over controlled and managed public IP networks.

Within this solution, voice packets are transmitted onto public IP networks which are controlled and managed by the two carriers establishing the voice bilateral interconnection. Based on this access method, adequate levels of network performance, reliability and reporting can be controlled and offered to provide high voice quality and services matching the TDM specifications.

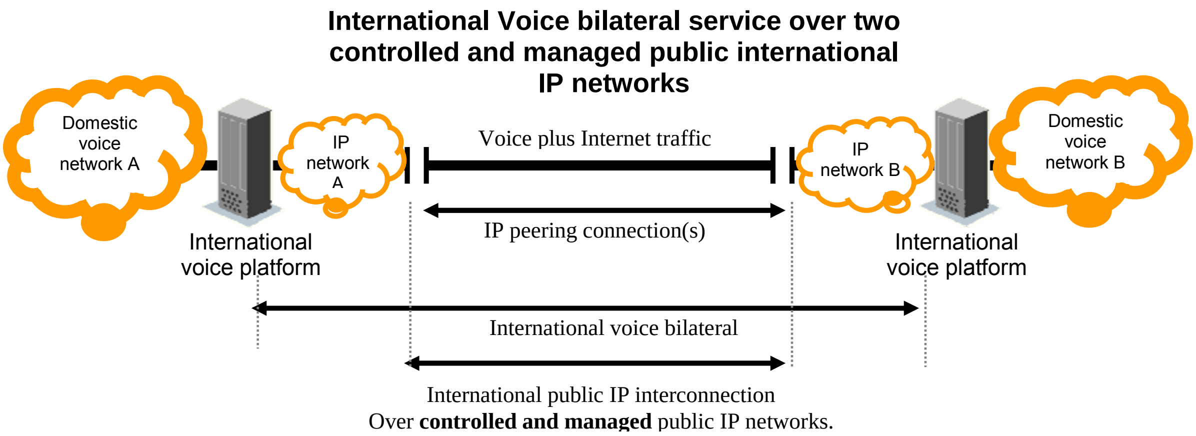

Figure 5 - International Voice bilateral service over two controlled and managed public international IP networks

- In this scenario both carriers can have an IP peering relationship or one of the carriers can be selling IP transit to the other carrier via an IP access link. The interconnection between the carrier A and B can be over PDH, SDH, Sonet, Ethernet …
- The IP connection(s) can be dedicated to voice packets or mutualised for voice and other types of internet traffic.
- The voice packets will always be routed to go directly between carrier A’s IP network and carrier IP B’s network, without using any other uncontrolled Internet network.
- In case of network congestion, voice packets might be prioritized over other Internet traffic.
- Both IP networks and the IP connection(s) between them should target the high quality described in the specifications of section 6.3.

6.2.2.2 **Over third parties Internet networks.**

With this solution voice packets are transported over the public internet without any control from the bilateral carriers neither on the routing nor on the quality of the transport which depends on one or more uncontrolled third party Internet networks.
Based on this access method, levels of network performance, reliability, security and reporting can be offered with a best effort quality, which at times can fluctuate and cannot guarantee to be sufficient to provide the high voice quality comparable to the one available in TDM.

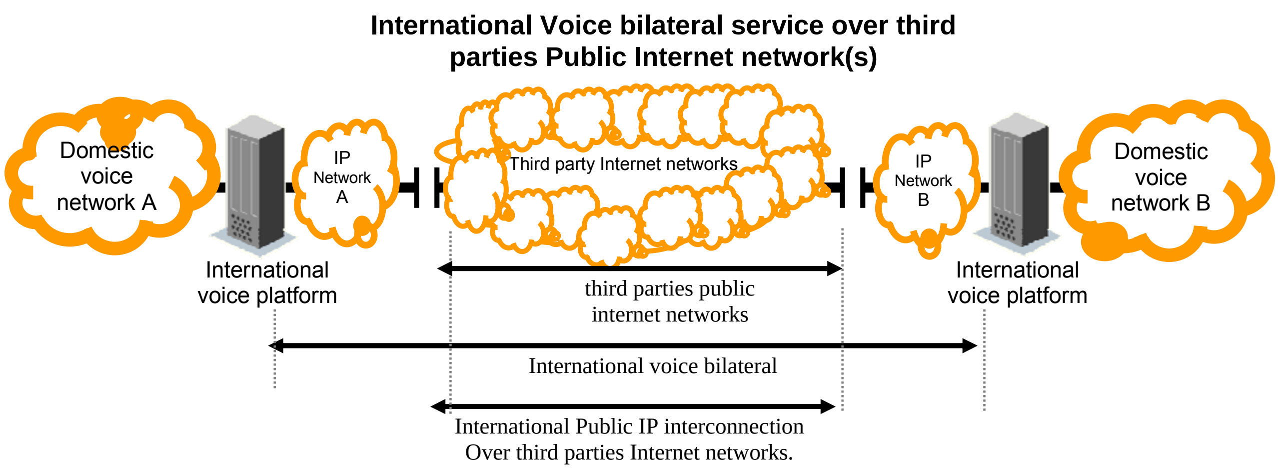

Figure 6 -International Voice bilateral service over third parties Public Internet network(s)

6.3 Services quality and specifications for international voice over IP

IP interconnections should guarantee the same high quality (ASR, PGRD… see details below…) as the quality of the best TDM interconnections existing today, while differences in provisioning and resource optimizations should be significant. However, TDM quality varies depending on the international destinations due to uneven telecom infrastructure existing worldwide. Different markets and different international network infrastructures capabilities reveal that different and lower levels of quality exist and should be addressed differently. But in all the cases a bilateral interconnection should always provide the best quality possible between two destinations, when compared to an indirect voice transport solution via one or more third party voice transit providers.

Voice quality and services specifications are described here after based on two categories of quality.

1. **High quality.** This quality matches the characteristics of the best TDM bilateral interconnections in terms of voice quality, service specifications, availability, privacy, security, and stability over time. International carriers historically buying high level quality services over TDM and servicing retail customers with high expectations should rely on this level of quality for their international voice bilaterals over IP.
2. **Best effort quality.** This quality relies on Internet transport services provided by uncontrolled third parties networks without any guarantee on the services provided. While in some cases the resulting voice quality could be adequate to implement voice over IP interconnections, this solution is not recommended as the first choice to be used to replace the high and consistent voice services quality found on high quality TDM bilaterals.

The quality provided over an international IP interconnection depends on the quality of several elements that constitute the overall interconnection. Some elements impact the voice quality while some others impact the services provided (such as CLI transparency). All these elements can have different levels of quality and the choices made for the transport, codecs, signaling and security will ultimately define the level of the resulting quality for the international voice interconnection. There are several combinations possible and as many resulting levels of quality. For carriers that wish to implement voice over IP interconnections with the high quality equal or better to what is possible in TDM, the high quality solutions should be selected for all of the elements when ever technically possible.

Interconnection solution	Voice services quality
Via a private and dedicated interconnection (figure 3)	highest
Via two controlled and managed public network (figure 5)	high
Via a private third party network with guaranteed QoS (figure 4)	high
Via uncontrolled third public party networks (the internet) (figure 6)	best effort

Figure 7 Quality of voice over IP interconnections

Transport quality (6.3.2)	+	**Codecs quality (6.3.4)**	+	**Signaling quality* (6.3.4)**	+	**Security quality (6.3.3)**	=	**Interconnection quality**

Figure 8 - quality components of an IP voice interconnection

*In the case of an IP international interconnection that connects two PSTN retail bases (PSTN – to IP international interconnection - to PSTN), it can be important that all the signaling information (ISUP) of one PSTN network has to be transported transparently to the other PSTN network. Some VoIP signaling protocols have different level of transparency of the PSTN information. It is clear that for the time being most of the traffic transported over IP international bilaterals will be traffic between two PSTN domestic networks. It is therefore important to understand and to include the impact of signaling into the overall quality of the international IP voice interconnection.

The sections below detail the characteristics of the constituting elements of an international IP interconnection and the values to be used as indicative targets for the high and best effort categories.

6.3.1 Voice Quality

	High quality ← Various levels of qualities →	Best effort
ASR (end to end voice indicator)	ASR includes customer behaviour and is route dependent. It is therefore not possible to provide a unique ASR value. However, historical data under comparable destinations and periods of time should show that the ASR for the High Quality solution will provide a higher value.	
NER (applicable to the international bilateral §6)	The NER depends on destinations (geographical and fixed versus mobile) the highest values should be comparable to the ones found in TDM. See "Interconnection Model for bilateral Voice Services"[14] document for definition of NER and exact scope of measurement	best effort
MOS (model E)	The MOS is dependent on many network elements choices (codecs, jitter…), however for the highest quality the MOS should be 4 and higher (ITU G.107). See "Interconnection Model for bilateral Voice Services" [14] document for definition of MOS and exact scope of measurement	best effort Due to the lack of guarantees on the transport solution relying on best effort.
Post Gateway Ringing Delay (PGRD)	under evaluation see "Interconnection Model for bilateral Voice Services" [14] document for definition of PGRD and exact scope of measurement	best effort
ALOC	ALOC includes customer behaviour and is destination dependent making it impossible to provide a unique ALOC value. However, historical data under comparable destinations and periods of time should show that the ALOC for the High Quality solution will provide a higher value.	

6.3.2 Network transport and architecture

<table>
<tr><th></th><th colspan="2">High quality — Various levels of qualities — Best effort</th></tr>
<tr><td>Number of international voice switching hops (applicable to the international bilateral §6)</td><td colspan="2">International voice switching (and transcoding) should be guaranteed to be minimized.
In most cases carriers should only use 1 international switch to terminate the international traffic they receive (figure 1)</td></tr>
<tr><td>Network availability (applicable to the international bilateral §6)</td><td>99.99% monthly with dual access, 99.95% with single access
see “Interconnection Model for bilateral Voice Services” [14] document for definition of network availability and exact scope of measurement</td><td>99.99% monthly with dual access, 99.95% with single access. But these values cannot be guaranteed due to involvement of third party networks</td></tr>
<tr><td>RTD (applicable to the international bilateral §6)</td><td>The international section of an international voice call has to minimise the network RTD and network processing time to provide high voice quality.
Indicative RTD values for specific routes are described in GSMA IR34 recommendations.
see “Interconnection Model for bilateral Voice Services” [14] document for definition of RTD and exact scope of measurement</td><td>higher values than private oriented interconnections</td></tr>
<tr><td>Packet Loss (applicable to the international bilateral §6)</td><td>< or equal to 0.1%
see “Interconnection Model for bilateral Voice Services” [14] document for definition of packet loss and exact scope of measurement</td><td>> or equal to 0.1%</td></tr>
</table>

Transport over dedicated IP link (applicable to the international bilateral §6)	most preferred (it provides the highest quality and guarantees)	not applicable
Transport over controlled and managed public/internet IP backbones (applicable to the international bilateral §6)	next preferred (quasi comparable to the transport over dedicated IP link)	not applicable
Transport over third party uncontrolled Internet networks (applicable to the international bilateral §6)	not applicable	applicable

6.3.3 Security, privacy

	High quality ← Various levels of qualities →	Best effort
secured transport of call setup (SIP messages).	encryption not required in case of private interconnection	encryption recommended
secured transport of voice media flow	not required	not required
need of security border equipment	required	strongly recommended

protection against confidentiality and anti-abuse of voice calls information (fraud, spit…)	required	the target should be the same as for the highest quality level but best effort availability will also be acceptable
CLIR	International voice carriers must comply to local regulations applicable to international voice calls. International carriers must by default be transparent to receive and forward CLI and CLIR information without any alteration. International carriers may agree bilaterally to not transmit the CLI for calls with CLI Restricted information.	

6.3.4 Supplementary services

	High Quality ← Various levels of qualities →	Best effort
CLI transparency (on the international bilateral portion §6)	guaranteed	Possible but not guaranteed.
OCN and RDN transparency (on the international bilateral portion §6)	guaranteed	Possible but not guaranteed.
Fax (T38) and G.711 pass through (on the international bilateral portion §6)	guaranteed	Possible but not guaranteed.
DTMF Inband RFC2833 (on the international bilateral portion §6)	guaranteed	Possible but not guaranteed.
Voice Codecs 300 -3400 Hz	All market segments and type of requirements should be addressed (fixed, uncompressed, fixed compressed, mobile…), however codecs should be limited to one codec per usage (uncompressed, compressed, wideband…). Transcoding services can be provided but are not recommended in order to minimize impact on voice quality. see “Interconnection Model for bilateral Voice Services” [14] document for a detailed definition of codecs and exact scope of implementation	
	G.711 A-law, µ-law 64 kbit/s	G.711 or G.729 a, b, ab 8kbit/s

Voice wideband codec	All market segments and type of requirements must be addressed (fixed, uncompressed, fixed compressed, mobile…) however codecs should be limited to one codec per usage (uncompressed, compressed, wideband…). Transcoding services can be provided but are not recommended.	
	G.729.1, G.722, wideband AMR,	not recommended
media codec transcoding	Acceptable but not desired	Acceptable but not desired
Full ISUP information transport transparency	Preferred guaranteed with SIP-I partially guaranteed with SIP	guaranteed with SIP-I partially guaranteed with SIP
Modem data transmission	guaranteed	possible but not guaranteed.

6.4 Routing

Currently the routing is the carrier's responsibility and depends on its own management choice. VoIP protocols offer the option to route every call at the last digit, enabling a new set of network optimization and new services. Such routing at the last number digit is already possible today in TDM with number portability for instance, but VoIP technically has the potential to deploy number base routing at the last digit more economically and at a faster pace. Evolutions of call routing through to VoIP are still being reviewed the time this document is published and recommendations on this topic will be part of a future release.

6.5 Comparison of the Interconnection solutions

Historically TDM voice bilaterals have been built mainly over international SDH, Sonet and PDH transmission circuits. E1s or T1s are commonly used in the business of international voice. These international transport circuits obey to a strict and well established standardisation. They provide high QoS transport and are available in most parts of the world.

Consequently, the transport technology chosen to build an international voice bilateral between two international destinations is most of the cases identical and provides similar quality which is for most international voice bilaterals very high transport quality (with the exception of satellite transmission which provides lower than terrestrial transport quality).

In IP, the transport of voice can be done through several solutions, which have unequal characteristics for security, privacy and network performance that impact the resulting voice quality. The IP transport quality can range from as good as the existing TDM voice bilaterals, to a much lower and unstable quality. Security and privacy in IP can also have different levels of quality, while in TDM this is not a problem.

To illustrate these differences one can look at the delay due to the distance of an international interconnection which is a critical element of the voice quality (ITU G.114 [16]). In TDM and in IP this delay is partially determined by the geographical distance. In TDM this distance and delay are usually the shortest possible and stay the same over time. In IP this delay depends on the Round Trip Delay which in case of a private and direct interconnection should be the shortest and consistent over time. However, for interconnections over the public internet, this delay can be close to the shortest value but cannot be guaranteed to be consistent and can in fact fluctuate substantially overtime with a noticeable impact in the resulting voice quality.

Packet loss is another important element that impacts the voice quality. Packet loss in TDM is not a problem as this technology by construction guarantees no packet loss. However, in IP, packets loss is a problem that can range from significant on public internet during times of congestions and failures, to neglectable with direct and private connections.

With voice over IP, the mean of transport of the voice becomes a significant differentiator of the final voice service quality. The i3 forum recommends the implementation of direct and private IP interconnections to migrate existing TDM voice bilaterals over IP in order to achieve a high quality comparable to TDM.

6.6 Accounting requirements

The migration from TDM to VoIP needs to keep guaranteeing the applicability of the existing billing principles used between international carriers. The systems today in place rely on common Call Data Record (CDRs) information. The same information found in TDM CDRs must be provided in Voice over IP interconnections. A detailed description of a CDR content is described in the technical recommendations document [14]