INTERNATIONAL INTERCONNECT FORUM FOR SERVICES OVER IP

(i3 Forum)

www.i3forum.org

IP international interconnections for voice and other related services (V 1.0) June 2009

This document updates and replaces the i3 Forum document "IP international interconnections for bilateral voice services (V.1.0) May 2008".

Scope

This document is the second sub-set of deliverables of the i3 forum Services and Requirements work stream. It includes the migration from TDM to IP of the international bilateral voice interconnections and extends the recommendations for the migration of other international voice related services from TDM towards IP interconnections.

Table of Contents

1 Introduction

It is difficult to predict the full variety of ways that IP will transform the voice industry in the coming years. Today's principal revenue streams may become less prominent, but the new environment will spawn new services, creating an exciting array of business opportunities. During this transition, the industry will need to overcome technical and interoperability challenges, refine standards, and codify best practices before operators become as fluent in IP as they currently are in TDM.

To meet these challenges head-on, the i3 Forum came together in 2007 to proactively develop strategies for a smooth, efficient and speedy transition to IP. The group has quickly grown to include 24 member carriers from around the globe. Collectively, Forum members serve over 1.5 billion retail customers in over 100 countries, and carry more than 80% of the world's international voice traffic.

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 international voice related services over IP interconnections. In other words, carriers need to find common and practical operational guidelines on how to implement the various existing voice over IP standards.

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 interconnections, 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 interconnections is broadly based on calling network's party pays, settlement and hubbing regime. These models will remain applicable over IP, their implementation will be decided by the carriers within commercial 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 recommendations for the international voice related services that are to be exchanged over IP between international **carriers**. These services should include (but may not be limited to) the voice and other related services currently exchanged over TDM interconnections. Such services include;

- Voice bilateral interconnections
- Voice hubbing interconnections
- Voice services such as telephony, fax, modem, ISDN, special services (international toll free..)
- SS7 signaling for mobile roaming

The recommendations should be usable not only by members of the forum, but also by any carrier outside of the forum to migrate existing TDM interconnections 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. This document also describes a set of Quality of Service indicators that are relevant to voice over IP and that can be exchanged between Carriers and Customers to report and eventually to commit on a level of QoS with relevant, measurable and commonly described indicators.

This document is to be read jointly with the other related i3 Forum documents which can be found at www.i3forum.org :

- Technical Interconnection Model for International Voice Services
- White Paper Optimal Codec Selection in International IP based Voice Networks
- White Paper Mapping of Signaling Protocols ISUP to/from SIP, SIP-I
- Interoperability Test Plan for International Voice services

3 Scope

The scope of this document is to cover **international voice and other related services currently transported over TDM interconnections**.

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

3pcc	Third Party Call Control
ACL	Access Control List
ACM	Address Complete Message
AF	Assured Forwarding
ALG	Application Level Gateway
ALOC	Average Length Of Call
ASR	Answer Seizure Rate
ATM	Asynchronous Transfer Mode
BA	Behavior Aggregate
BE	Best Effort
BFD	Bidirectional Forwarding Detection
BGCF	Breakout Gateway Control Function
BGP	Border Gateway Protocol
BSS	Business Support System
CBC	Cipher Block Chaining
CC	Country Code
CDR	Call Detail Record
CLI	Calling Line Identity
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
COLP	COnnected Line identification Presentation
COLR	COnnected Line identification Restriction
CSCF	Call Session Control Function
DDoS	Distributed Denial of Service
DES	Data Encryption Standard
Diffserv	Differentiated Services

DoS	Denial of Service
DPO	Dynamic Port Opening
DSCP	Differentiated Services Code Point
DTMF	Dual-Tone Multi-Frequency
DWDM	Dense Wavelength Division Multiplexing
EF	Expedite Forward
EXP	MPLS header EXPerimental use field
FoIP	Fax over IP
GIC	Group Identification Code
GSDN	Global Software Defined Network
GSN	Global Subscriber Number
IAM	Initial Address Message
IBCF	Interconnection Border Control Function
I-BGF	Interconnection Border Gateway Function
IC	Identification Code
IFP	Internet Facsimile Protocol
IFT	Internet Facsimile Transfer
IKE	Internet Key Exchange
IMS	IP Multimedia Subsystem
IP	Internet Protocol
IPSec	IP Security
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
IVR	Interactive Voice Response
KPI	Key Performance Indicator
MF	Multi-Field Classifier
MGCF	Media Gateway Control Function
MGF	Media Gateway Function
MIME	Multipurpose Internet Mail Extensions
MNO	Mobile Network Operator
MoIP	Modem over IP
MOS	Mean Opinion Scale
MOS_{CQE}	Mean Opinion Score, Communication Quality Estimated
MPLS	Multiprotocol Label Switching
MTP	Message Transfer Part (SS7)
NAPT	Network Address and Port Translation
NAT	Network Address Translation
NDC	National Destination Code
NER	Network Efficiency Ratio
NNI	Network Network Interface
NN	National Number
OCN	Original Called Number
OLO	Other Licensed Operator
OSS	Operations Support System
P-router	Provider router
PDH	Plesiochronous Digital Hierarchy
PE-router	Provider Edge router
PGRD	Post Gateway Ringing Delay
PHB	Per-Hop Behavior
POS	Packet Over Sonet
PSTN	Public Switched Telephone Network
QoS	Quality of Service
R-Factor	Rating-Factor
RgN	Redirecting Number
RI	Redirecting Information
RTCP	Real Time Control Protocol
RTD	Round Trip Delay
RTP	Real-Time Protocol
SCTP	Stream Control Transmission Protocol
SDES	Source Description
SDH	Synchronous Digital Hierarchy
SDP	Session Description Protocol
SGF	Signalling Gateway Function
SIP	Session Initiation Protocol
SIGTRAN	Signalling Transport suite of Protocols
SIP URI	SIP protocol Uniform Resource Identifier

SIP-I	SIP with encapsulated ISUP
SIP-T	SIP for Telephones
SLA	Service Level Agreement
SN	Subscriber Number
SPRT	Simple Packet Relay Transport
SR/RR	Sender Report/Receiver Report
TCP	Transmission Control Protocol
TDM	Time Division Multiplexing
TE MPLS	Traffic Engineering MPLS
tel-URI	Telephone Uniform Resource Identifier
TISPAN	Telecommunications and Internet converged Services and Protocols for Advanced Networking
TLS	Transport Layer Security
TOS	Type Of Service
TSG	Trunk Group
TUP	Telephone User Part
UDP	User Datagram Protocol
UDPTL	Xxxx, Appears in 8.2
URI	Uniform Resource Identifier
URL	Uniform Resource Locator
UUI	User-to-User Information
VBD	Voice Band Data
VLAN	Virtual Local Area Network
VoIP	Voice over IP
VPN	Virtual Private Network

5 References

[1] i3 forum “Technical Interconnection Model for International Voice Services” current release
[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 “Technical Interconnection Model for International Voice Services” current release
[15] ITU-T Recommendation E.105 “INTERNATIONAL TELEPHONE SERVICE”, August 1992
[16] ITU-T Recommendation G.114 “One-way transmission time“, May 2003

6 Voice bilateral and hubbing service description

6.1 Bilateral service description

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

International bilateral voice interconnections are used to interconnect retail networks (mobile or fixed). Bilateral interconnections in the TDM world provide the highest quality available for international voice calls, they rely on the best quality of transport available and limit 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 telecom 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.

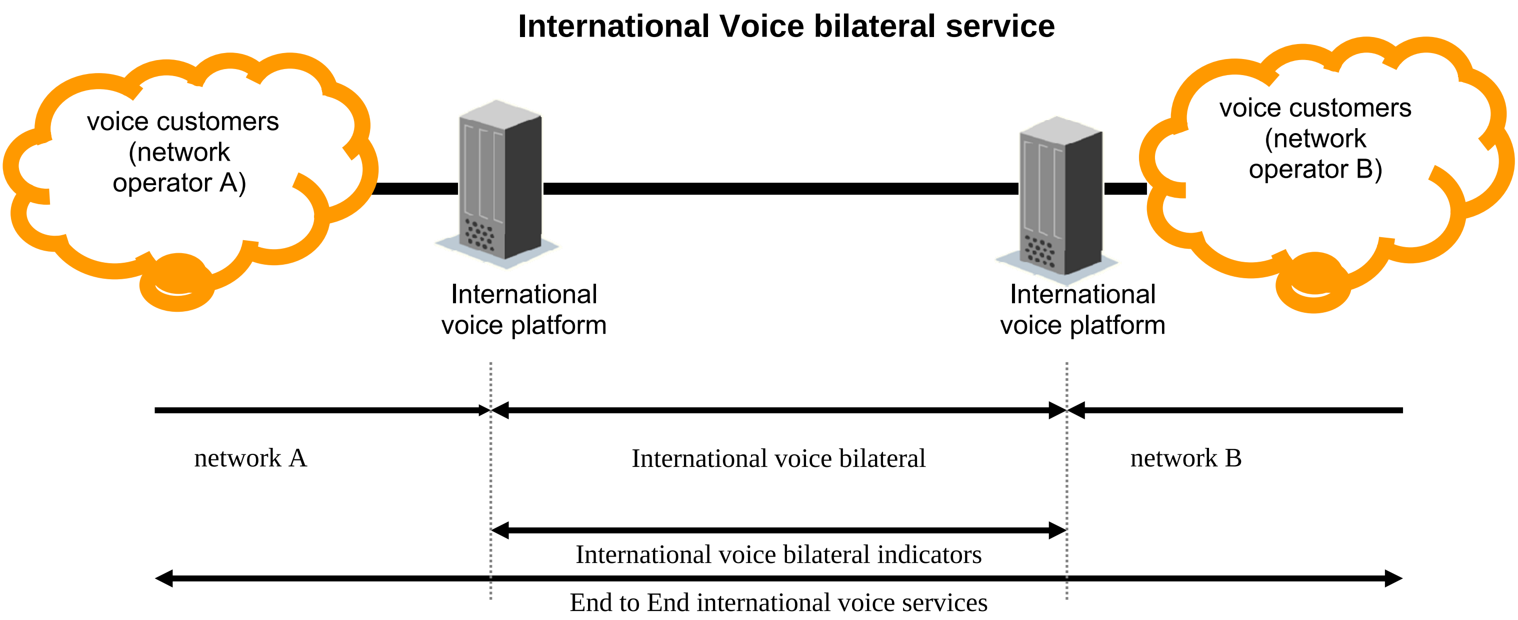


Figure 1 - International Voice bilateral service

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 terminating customers can be voice over IP 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 voice over IP. It is left to the bilateral negotiations between carriers to decide about which combination of voice traffic to carry through the international IP interconnection (figure 2).

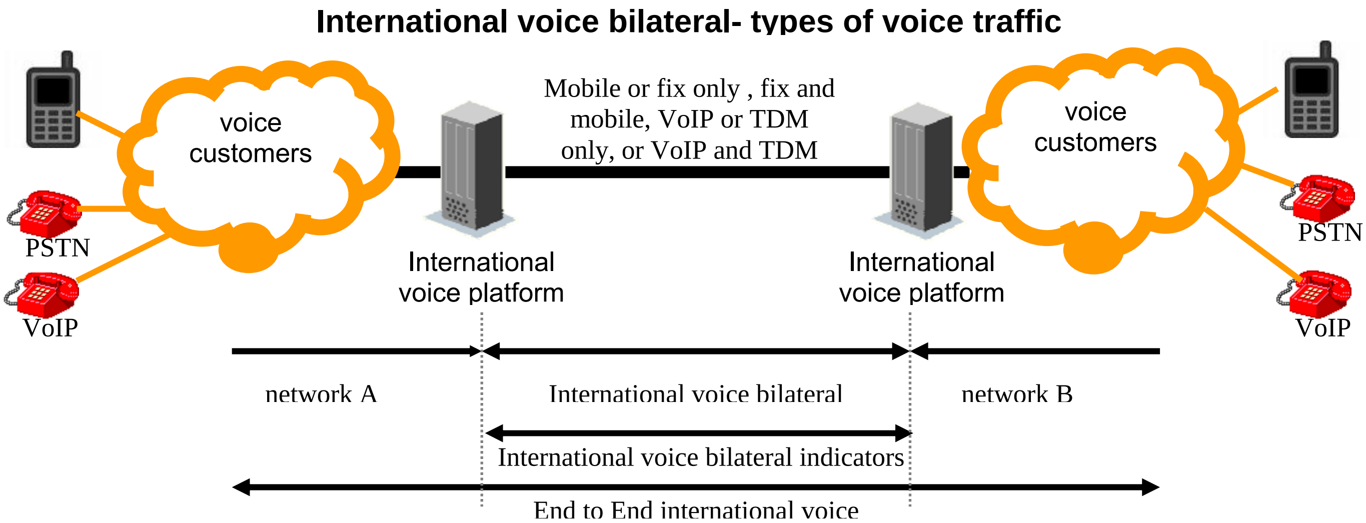


Figure 2 - International voice bilateral- types of voice traffic

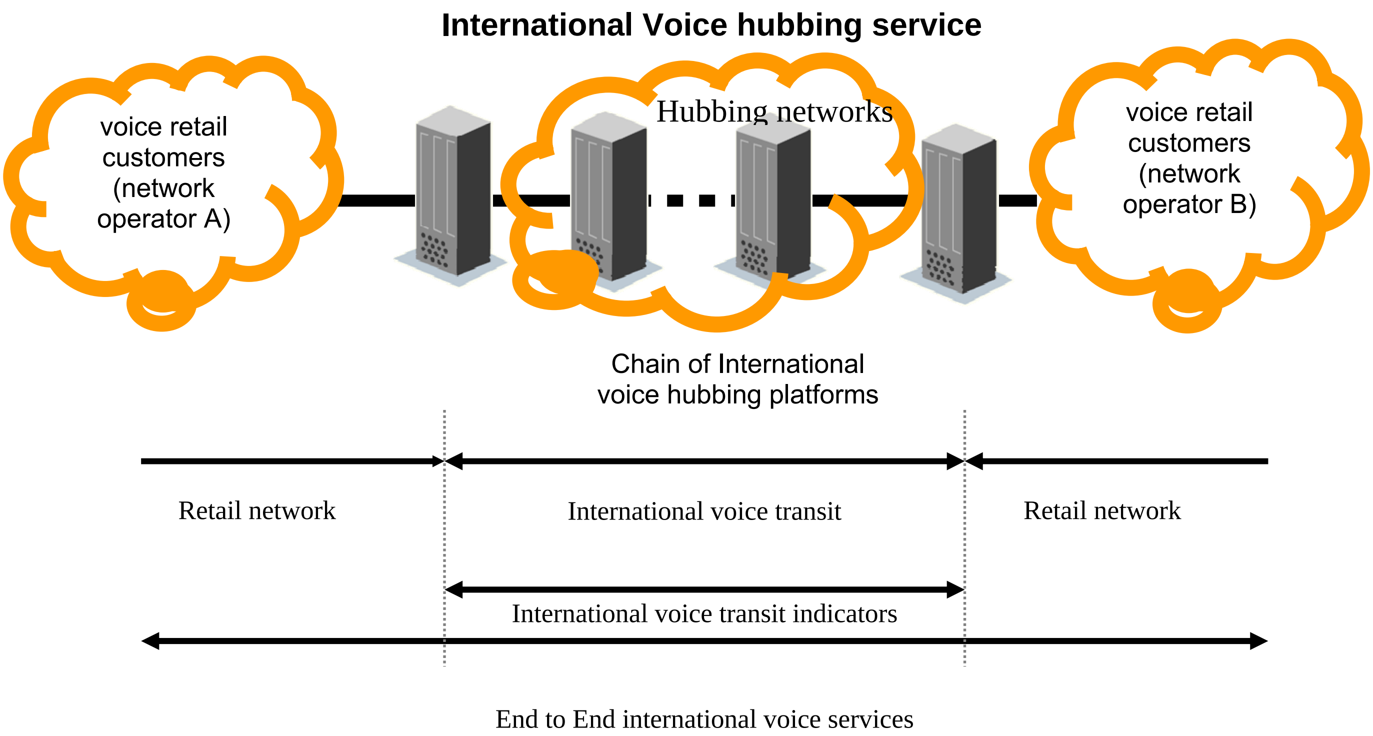
6.2 Hubbing service description

The international voice hubbing service is provided via multiple (more then 2) international carriers in order to deliver voice services between end users.

International voice hubbing services are used to exchange international voice calls with multiple networks via one voice over IP interconnection. This is different than the bilateral interconnection where the exchange only happen between two networks, one on each side of the voice over IP interconnection. The quality of the voice interconnection commercially required (low, medium or high) will dictate the interconnection topology to be selected. For the low quality level, it should be expected that the calls will have multiple transit hops as they transit through several carriers towards the destinations. For the high quality level, it should be expected that the quality of the calls should be close to the one of a bilateral service as the hubbing calls would run over the bilateral interconnections (also called direct routes). The hubbing service provider will have as many bilateral connections as high quality hubbing destinations provided.

The quality of hubbing voice interconnections will vary depending on the selected voice interconnection topology and will, like bilateral interconnections 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.



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

The terminating customers can be voice over IP 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 traffic: mobile, fixe, retail, wholesale TDM or Voice over IP. It is left to the hubbing negotiations between carriers to decide about which combination of voice traffic to carry through the international IP interconnection.

6.3 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 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.

6.4 Interconnection solutions

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

6.4.1 Via private and dedicated IP link

Carriers can establish IP interconnections implementing a dedicated IP connection directly between the two carriers (logically equivalent to their existing direct TDM 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 voice over IP traffic to ensure security and also voice prioritization versus other internet traffic in case of network congestion. The international voice interconnection and underlying dedicated link(s) should be engineered to provide all specifications of section 6.5.

- 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 4).

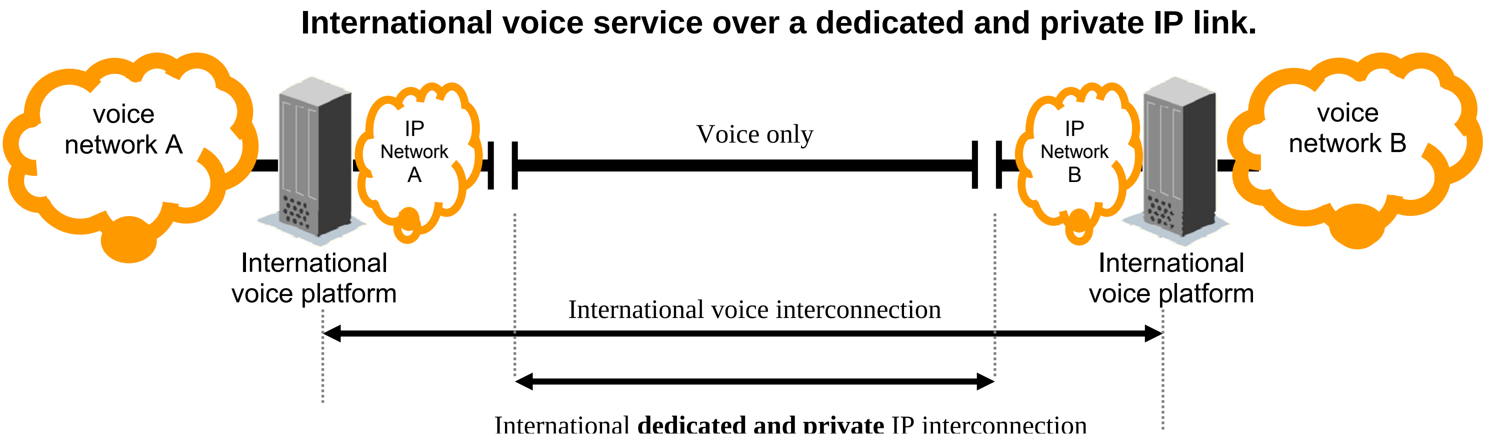


Figure 3 - International voice service over a dedicated and private IP link.

- 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 5).

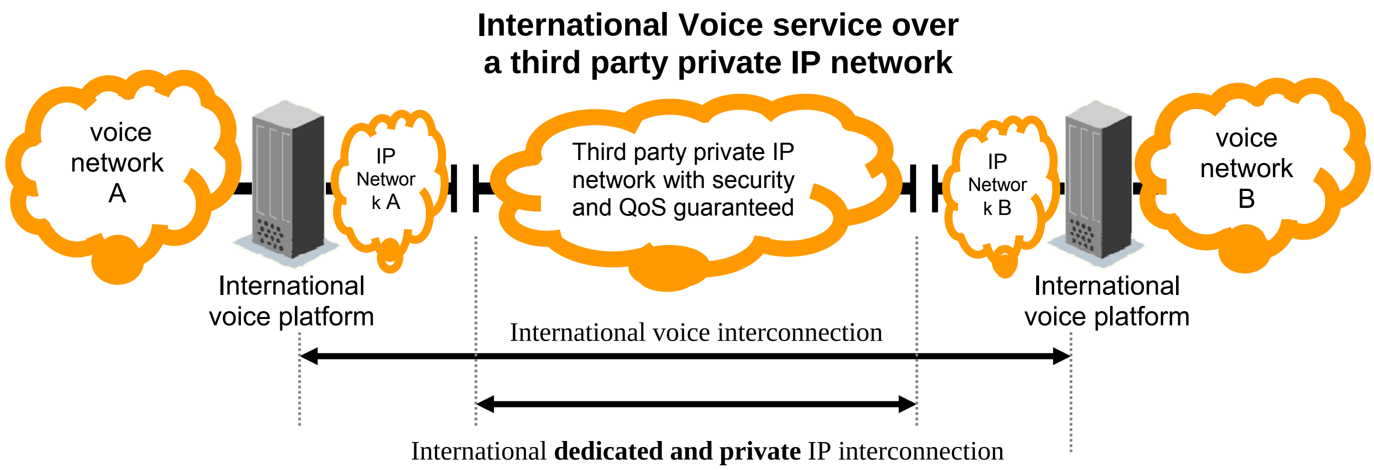


Figure 4 - International Voice service over an international private IP network

6.4.2 Via Public IP networks

Carriers can establish an IP 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 voice interconnection, and a second scheme that uses public networks that are not controlled nor managed by the carriers implementing the voice interconnection (i.e. using third party Internet networks).

6.4.2.1 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 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.

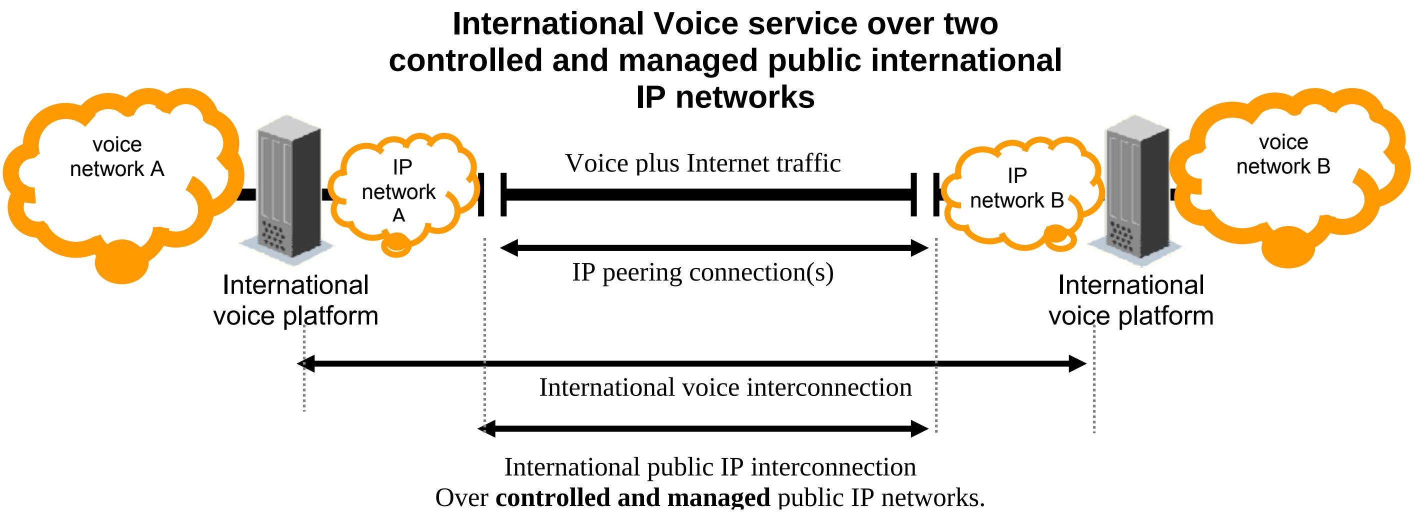


Figure 5 - International Voice service over two controlled and managed public international IP networks

- In this scenario both carriers can have an IP commercial 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 6.5.

6.4.2.2 Over third parties Internet networks.

With this solution voice packets are transported over the public internet without any control from the voice 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.

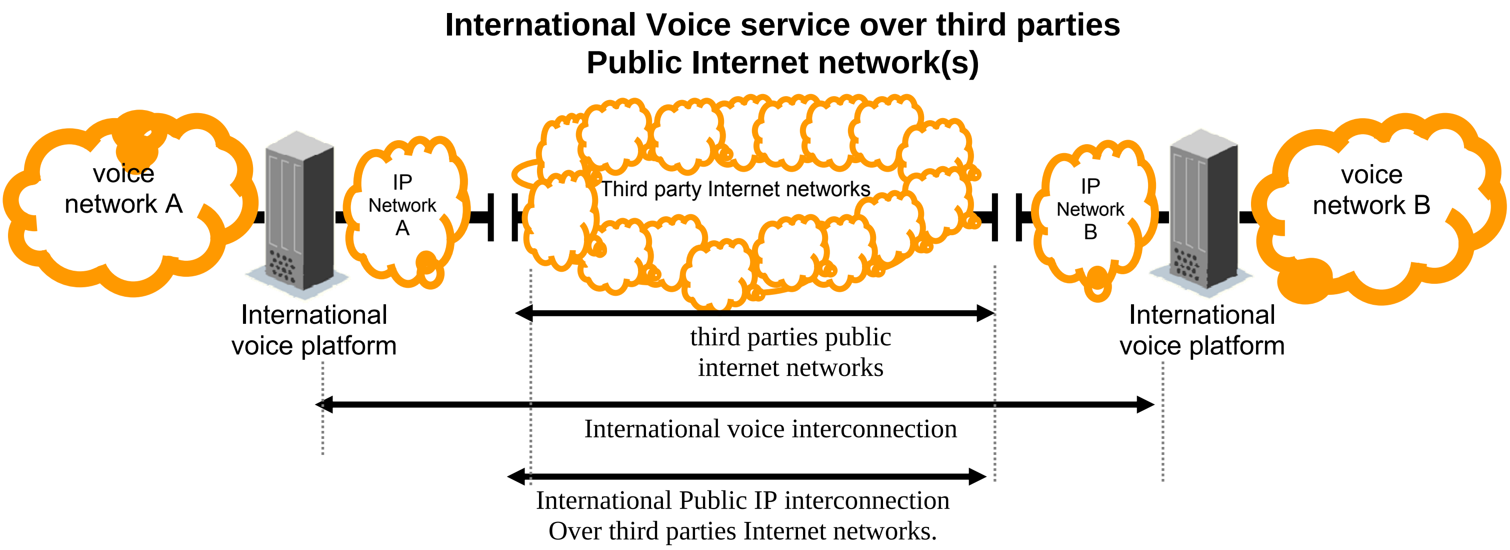


Figure 6 -International Voice service over third parties Public Internet network(s)

6.5 Services quality and specifications for international voice over IP interconnections

IP interconnections should guarantee the same 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 (hubbing) 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 or high quality hubbing service 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 interconnections.

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 4)	highest
Via two controlled and managed public network (figure 6)	high
Via a private third party network with guaranteed QoS (figure 5)	high
Via uncontrolled third public party networks (the internet) (figure 7)	best effort

Figure 7 Quality of voice over IP interconnections

Transport quality (6.5.2)	+	**Codecs quality (6.5.4)**	+	**Signaling quality* (6.5.4)**	+	**Security quality (6.5.3)**	=	**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 voice over IP 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.

6.5.1 Voice Quality

	High quality	Various levels of qualities ⟷ Best effort
ASR (end to end voice indicator)	ASR includes customer behaviour and is route (destination) 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 "Technical Interconnection Model for International 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 "Technical Interconnection Model for International 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)	see "Interconnection Model for bilateral Voice Services" [14] document for definition of PGRD and exact scope of measurement	best effort
ALOC	ALOC includes customer behavior 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.	

6.5.2 Network transport and architecture

	High quality — Various levels of qualities —	Best effort
Number of international voice switching hops	International voice switching (and transcoding) should be guaranteed to be minimized. In most cases carriers should only use one international switch to terminate the international traffic they receive (figure 1). This is applicable for bilateral and high quality hubbing service interconnections.	Lower quality hubbing services could use lower quality interconnection topologies with several (more than two) international carriers in the chain.
IP Network availability	Market benchmark for highest IP quality 99.99% monthly with dual access, 99.95% with single access see “Technical Interconnection Model for International Voice Services” [14] document for definition of network availability and exact scope of measurement	best effort
RTD	The international section of an international voice call has to minimize the network Round Trip Delay and network processing time to provide high voice quality. Indicative RTD values for specific routes are described in GSMA IR34 recommendations. see Technical Interconnection Model for International Voice Services” [14] document for definition of RTD and exact scope of measurement	worse values than private oriented interconnections
Packet Loss	< or equal to 0.1% see “Technical Interconnection Model for International Voice Services” [14] document for definition of packet loss and exact scope of measurement	> or equal to 0.1%

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

6.5.3 Security, privacy

	High quality ← Various levels of qualities →	Best effort
secured transport of call setup (SIP messages).	encryption required except 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 with 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.	

6.5.4 Supplementary services

	High Quality	Various levels of qualities ↔ Best effort
CLI transparency	guaranteed	Possible but not guaranteed.
OCN and RDN transparency	guaranteed	Possible but not guaranteed.
Fax (T38) and G.711 pass through	guaranteed	Possible but not guaranteed.
DTMF Inband RFC2833	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 “Technical Interconnection Model for International 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.

6.6 Routing

Currently the routing is the carrier’s responsibility and depends on its own management choice. Voice over IP 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 voice over IP 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 voice over IP are still being reviewed the time this document is published and recommendations on this topic will be part of a future release.

6.7 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 standardization. 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 neglectible 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 and high quality hubbing over IP in order to achieve a high quality comparable to TDM.

6.8 Accounting requirements

The migration from TDM to Voice over IP 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 Interconnection Model for International Voice Services [14]

7 Migrating SS7/C7 (for mobile roaming) over IP

7.1 Definition

The signaling system known in North America as SS7 and elsewhere as C7 (the SS7/C7 terminology will be used in the following) is a set of protocols by which network elements (signaling points) exchange information to facilitate wireline and wireless call set-up, calls control, routing, billing and network management.

SS7/C7 has been developed for TDM based interconnection hence, migrating from TDM to Voice over IP.there is a need to have a solution to transport the SS7-C7 signaling in IP. This is the purpose of this section.

7.2 The Role of SS7/C7

This section gives a brief introduction on the major services and list of functions that Signaling transport service on SS7/C7 network, is able to provide :

1. Setting up and tearing down circuit-switched connections, such as telephone calls made over both cellular and fixed-line.
2. Advanced network features such as those offered by supplementary services (calling name/number presentation, Automatic Callback, and so on).
3. Mobility management in cellular networks, which permits subscribers to move geographically while remaining attached to the network, even while an active call is in place. This is the central function of a cellular network.
4. Customized Applications for Mobile network Enhanced Logic *(Camel)*. Camel enables end-user to roam between different networks (maybe in different countries) be reachable at the same number and receive only one bill from the original service provider (home Operator). In particular CAMEL allows: PrePaid roaming, Short Code, VPN Roaming, Location Based services (Virtual Home Environment).
5. *Short Message Service (SMS)* and *Enhanced Messaging Service (EMS)*, where SS7/C7 is used not only for signaling but also for content transport of alphanumeric text.
6. Support for Intelligent Network (IN) services such as toll-free (800) calling.
7. Support for ISDN.
8. *Local Number Portability (LNP)* to allow subscribers to change their service, service provider, and location without needing to change their telephone number.

Session Initiation Protocol (SIP) or the ISUP enabled SIP signaling profile (SIP-I) can be used for signaling of voice applications. For specific mobile roaming and data applications, such as those listed above under #3, #4 and #5, the usage of the Sigtran protocol stack is recommended to allow the transport of the SCCP (Signaling Connection Control Part) included in the SS7/C7 messages in order to enable the management of applications which are still SS7/C7 based. It has to be noted that SIP-I can not be used to transport the signaling (SCCP) required for mobile data uses (i.e roaming ..).

7.3 Protocol to be used to implement SS7/C7 over IP

To migrate the transport of international signaling traffic for roaming messages and SMS/EMS traffic over IP, Sigtran capabilities has to be supported on international signaling gateway .

Sigtran Signaling Gateway (SGW) is a network component that performs packet level translation of signaling from common channel signaling (based upon SS7/C7) to Sigtran signaling (based upon IP). A Signaling Gateway can be implemented as an embedded component of some other network element, or can be provided as a stand-alone network element. The Signaling Gateway function can also be included within the larger operational domain of a Signal Transfer Point (STP).

Signal Transfer Point (STP) is a router that relays SS7/C7 messages between Signaling End-Points (SEPs) and other Signaling Transfer Points (STPs). Typical SEPs include Service Switching Points (SSPs) and Service Control Points (SCPs).

7.4 Sigtran for inter-carrier connectivity

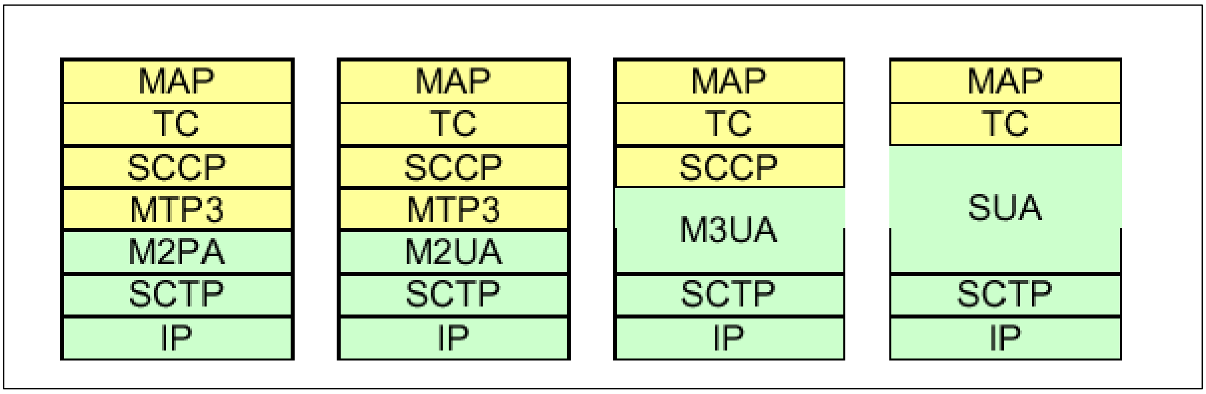
Signaling Transport (Sigtran) is a new set of standards defined by the International Engineering Task Force (IETF). This set of protocols has been defined in order to provide the architectural model of signaling transport over IP networks. IP network should provide an effective way to transport user data, expand networks and build new services.

Signaling Transport over IP network, in comparison to a legacy TDM–based network, should enable the following benefits:

- Less costly equipment: there should be no need for further expensive investments in the legacy signaling transport elements.
- Higher bandwidth—Sigtran information over IP should be not subject to a link capacity constraint as it is in the SS7/C7 network. The IP network is much more flexible than the TDM-based legacy network.
- Enhanced services: implementing a core IP network should facilitate a variety of new solutions and value-added services (VAS).

The architecture defined by IETF Sigtran working group consists of:

- a standard IP protocol;
- a new signaling transport protocol: SCTP;
- adaptation sub-layers: M2PA, M2UA, M3UA, SUA. Only one protocol can be set at a given time



Among the various Sigtran adaptation protocol stacks, for interconnection between Signalling Gateways (i.e for inter-carrier connectivity), the preferred solutions are those ensuring reliable message delivery. In “i3F - Technical Interconnection Model for international voice services” two different Sigtran protocol stacks have been selected:

- Message Transfer Part 2 (Peer-to-Peer Adaptation Layer (M2PA) and
- Message Transfer Part 3 (User Adaptation Layer (M3UA).

Further technical details together with figure of the relevant protocol stacks for the 2 cases mentioned above are given in the latest version of “i3F - Technical Interconnection Model for international voice services”.

While this protocol realizes only an encapsulation of the SS7/C7 signaling, Sigtran does not perform any ANSI/ETSI conversion. Therefore any interoperability between C7 and SS7 needs to be addressed independently of Sigtran. This problem is not related to nor solved by a migration from TDM to IP.

7.5 Security and reliability

7.5.1 Security

This section examines and recommends the security measures to be implemented in order to achieve secured communication.

The convergence of SS7/C7 with IP networks architectures requires the need for security enforcement. At present, traditional SS7/C7 uses some security mechanisms (TCAP, Handshake and TPAC) in order to secure signaling messages exchanged between SS7/C7 network nodes.

In general, security objectives should focus on :

- Communication security
    - Authentication of peers
    - Integrity of user data transport
    - Confidentiality of user data
    - Replay protection
- System security avoidance of :
    - Unauthorized use
    - Inappropriate use
    - Denial of services

When a network using Sigtran protocol involves more than one party, it may not be reasonable to expect that all parties have implemented security in a sufficient manner. Therefore, in order to guarantee end-to-end security and confidentiality, for public interconnections it is recommended to use IPsec.

7.6 Reliability

Signaling information for mobile uses is highly sensitive and requires high availability and QoS. Solutions to provide this QoS can be different for Sigtran than for other Voice over IP implementations. While some voice interconnections can be implemented over lower security and lower quality topologies, mobile signaling should only be implemented over highly secured and with high quality topologies. Sigtran QoS and network architecture will be further discussed in a later version of this document.

Monitoring signaling traffic is the simplest method to detect the accidental or intentional abuse of the SS7/C7 network. If the SS7/C7 network goes down, signaling performance is affected, including incorrect billing, lack of cellular roaming functionality, failure of Short Messaging Service (SMS) transfer, unexpected cut-off during calls, poor line quality, etc.

7.7 Sigtran Signaling Gateway (SGW) to SGW interconnection

The interconnection between two SGWs can be realized according to one of the following two options:

- IP dedicated connection: direct IP connection between SGWs. When using a public direct IP connection, use of IPSec is recommended.
- Public IP network connection plus IPSec: IPSec tunnel needs to be set-up between the SGW.

General Architecture Configuration

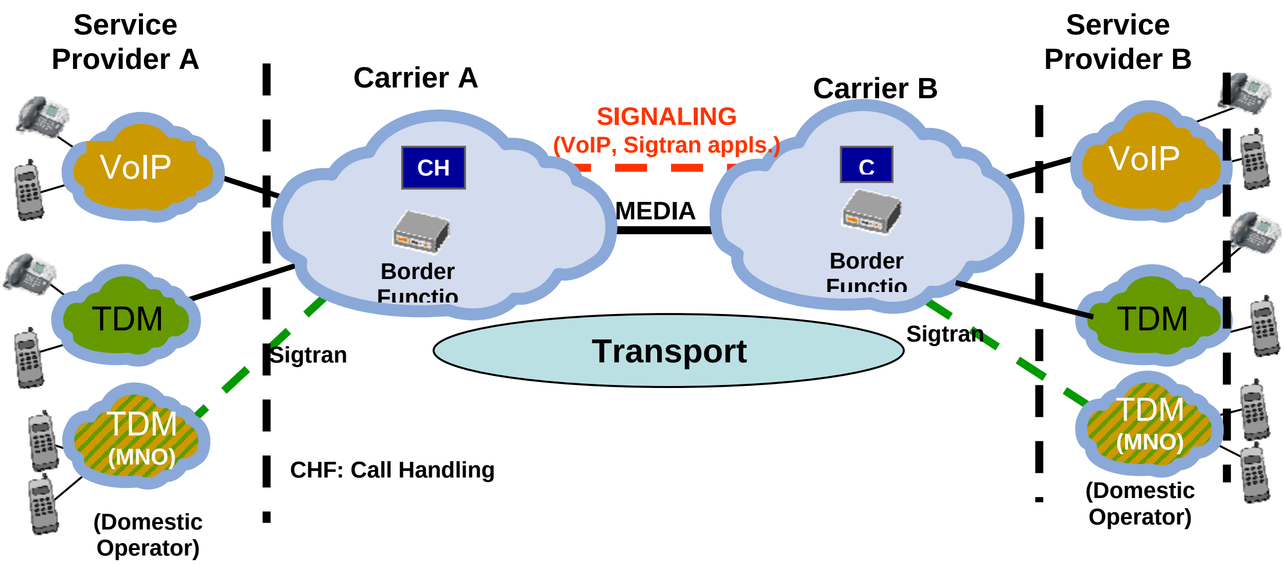


Figure 9 - General Architecture Configuration

For operators that need to use SS7/C7 signaling to enable non-voice services (e.g mobile roaming, messaging..), and that wish to cut the TDM links, it is recommended to use Sigtran as described in this document and in the technical document "**Technical Interconnection Model for International Voice Services**".

For operators that are not cutting their TDM links but that wish to save transport costs and use IP to transport SS7/C7 signaling, the recommended solution is Sigtran as described in this document and in the technical document "**Technical Interconnection Model for International Voice Services".**

8 Ring-back-tone

The context how a ring-back-tone (or other early media) is delivered towards the calling party is fundamentally different in a voice over IP environment, compared to a TDM environment. Many different standards are defined, sometimes formulating totally opposite principles. For that, the i3forum has defined a recommendation on how ring-back-tone should be managed over a voice over IP interconnection between carriers and/or operators.

8.1 Description of the ring-back-tone topic

8.1.1 Definition

During the process of call set-up, the calling party must hear a ring-back-tone (or other kind of early media) confirming the device of the called party is effectively ringing and waiting for the called party to answer the call. This is a key feature of any voice service, for any end-customer.

8.1.2 Ring-back-tone context in TDM networks

In TDM networks, this ring-back-tone is always produced by the called party network. It is sent over currently through the TDM voice channel (a 64k circuit) which is set-up up to the device of the calling party.

As a result of this approach the calling parties hear a different ring-back-tone, depending on the B-end network they are calling, which provides the ring-back-tone.

It is current industry practice that in the context of a TDM service, billing only starts the moment a call is answered. Costs related to transporting the ring-back-tone end-to-end often via multiple networks are not billed separately.

It is important to mention that in TDM networks, 10% to 20% of the invested network infrastructure is (on a permanent basis) occupied for the purpose of transporting ring-back tone, and the transport of ring-back-tone as such, represents a relatively high network cost for the industry.

For example:

- assuming a successful call of an average 3 minutes (180 seconds), and on average 20 seconds of ringing before the call is answered.
- assuming for unsuccessful calls, ringing is also on average 20 seconds, and of course the call duration for unanswered calls is zero.
- assuming that the ratio of successful/total calls (ASR) is 50%

This example gives an approximate average: [(20 + 20) / (180 + 20 +20) of 18 % network occupation by ring-back-tones, versus approximately 82 % of effective traffic.

Of course, each carrier must analyze its own statistics on call duration (ALOC), ringing-time and ASR, for a more accurate carrier-specific estimation.

8.1.3 Ring-back-tone context in Voice over IP networks

In Voice over IP networks the TDM principle of providing ring-back-tone can be improved, providing a positive change if the ring-back-tone is produced by the calling party's voice over IP end-device, or by the local network (Voice over IP software, Voice over IP phone, Voice over IP class 5 gateways, etc.). This action is triggered by SIP signaling (SIP message 180), instead of transporting the ring-back-tone over the network as a media-flow (RTP) coming from the (terminating) far end. This approach is generally followed in most of today's operational Voice over IP networks.

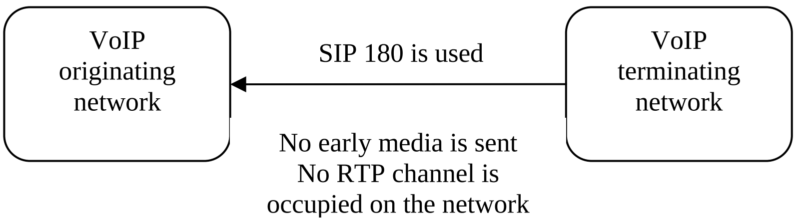
Some consequences of the Voice over IP SIP 180 approach for delivering ring-back-tone are:

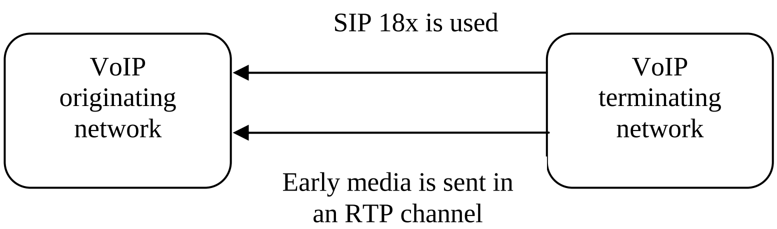
- Because the ring-back-tone is produced at the calling party's end, and is not related anymore to the ring-back-tone sent by the B-end network that is called, the calling party will in most cases hear the same ring-back-tone for all the countries called. The calling party network has of course the option to replace the classic ring-back-tone by any other kind of early media, which could enable new services (e.g. advertising). It is also possible to produce early media to support destination based ring-back-tone. However this is rarely implemented in practice because of the high cost to produce and manage multiple ring-back-tones.
- The main advantage of not passing the ring-back-tone from the B-end network to the calling party is that there is no RTP network capacity used anymore to transport a ring-back-tone media-flow over the network(s). This improves the network(s) utilization efficiency, as the capacity once needed for RTP is released for voice traffic. As explained above under the TDM paragraph, the improvement is substantial at approximately 10 to 20%.

In a Voice over IP network, it is also possible to send early media such as "ring-back-tone" from the called network to the calling party. The called network indicates this by sending a SIP message 18x (x can be 1-9), instead of SIP 180, and then sets-up an RTP channel in which the early media is sent.

As such, a Voice over IP terminating network behaves in a similar way to a TDM network, but of course the advantages related to use of SIP 180 to produce the ring-back-tone at the calling end (as described above) are lost.

The two scenarios for early media (ring-back-tone) over a Voice over IP interconnection





8.1.4 Ring-back-tone context in hybrid networks (Voice over IP and TDM)

As voice over IP and TDM will still have to interwork for the coming years, an i3 Forum recommendation had to be defined as to how to manage this issue.

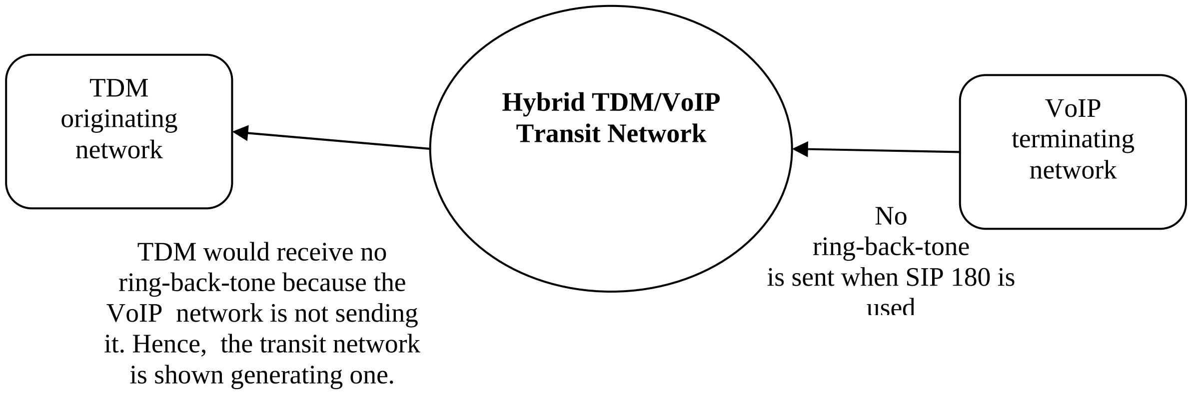
For TDM customers calling a Voice over IP network:

A potential issue is when a legacy TDM network makes a call towards a Voice over IP network via a hybrid transit network. Expectations and technical standards in this scenario could be contradictory, and as a result the TDM calling party could potentially never receive a ring-back-tone from the called party network (because it uses a SIP 180 setup). The caller will as a result hear nothing until the call is effectively answered; this is not acceptable per existing industry standard practice.

For a Voice over IP customer calling a TDM network:

In the opposite direction, Voice over IP calling TDM, and where the calling network is not expecting to receive a ring-back-tone but only the SIP 180 message, networks in the chain must be configured to avoid a conflict with the Voice over IP SIP 180 principle of working.

Example of a potential problem with ring-back-tone in a hybrid context



8.1.5 Foundation for the recommendation

The i3forum service group recommendation formulated below takes into account the following business parameters:

- For both Voice over IP & TDM carrier customers, the needed operational management related to ring-back-tone must at all time be clear at the interconnection level by using the correct SIP signaling.
- For TDM customers they must continue to receive in all cases some kind of ring-back-tone from their upstream provider, unless otherwise commercially agreed.
- Costs must at all times be optimized as much as possible and savings opportunities offered by the use of new technology (in this case Voice over IP) must be stimulated. Saving from 10-20% on transport network resources by using SIP 180 compared to SIP 18x plus early media is a very good example of such an opportunity. However, other commercial needs might be negotiated and implemented on a case by case basis.
- Implementation and management of a voice interconnection has to be as much as possible standardized between all carriers and must enable simple commercial solutions. It has to

take into account the complexity of having multiple carriers in a chain. The recommendation must be simple and straight forward and cover all possible originating, transit and termination scenarios in a reliable and at all time predictable manner.

It has to be noted that in specific situations certain business parameters could overrule the ones selected above by the i3 Forum and this could drive other interconnection principles related to ring-back-tone and other early media. These specific solutions are considered as being “non-default” and therefore might only be possible under an end-to-end commercial & technical agreement between the originating party, termination party and/or transit supplier(s) in the chain.

8.2 Recommendation for ring-back-tone management

As a solution on all possible scenarios and taking into account the above defined foundations, the following recommendations have been defined by the I3 Forum:

The recommendations are:

- A Voice over IP service supplier terminating a call and using the SIP or SIP-I signaling protocol shall at all times, as a minimum send a SIP 180 (ringing) signaling message towards the originating network, indicating that the receiving party’s device is ringing. Following SIP or SIP-I standards, no ring-back-tone or other early media will be made available in a RTP channel towards the originating network when SIP 180 signaling is used.

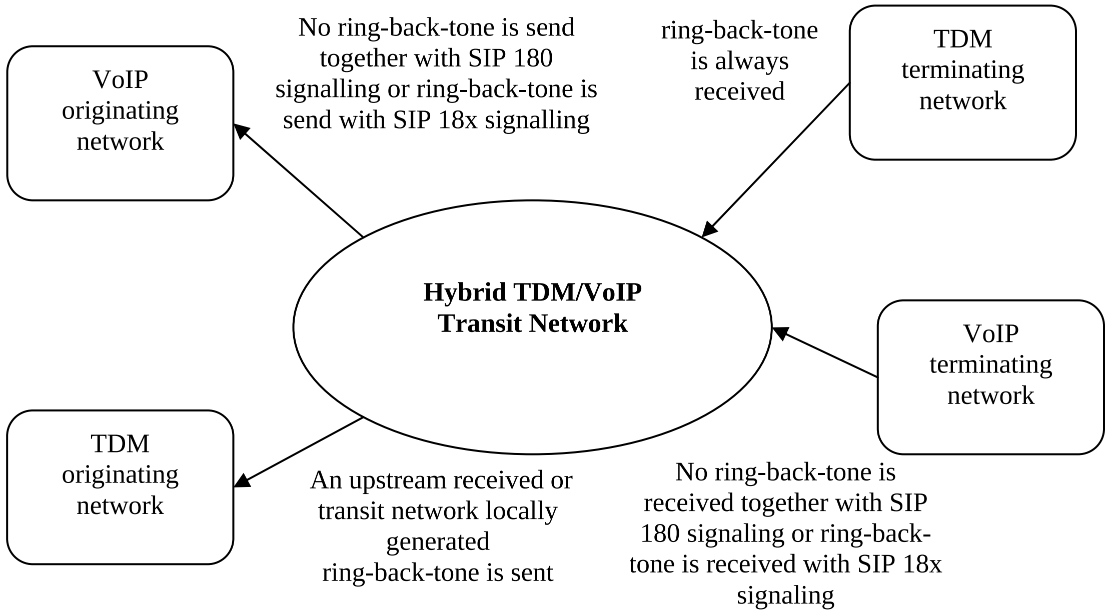
- On a “best effort basis” a Voice over IP service supplier/carrier will transparently pass a ring-back-tone, or other early media (if received downstream or made available locally) in a RTP channel towards the originating network. In that case a SIP 18x (x is number from 1 to 9) signaling message rather than a SIP 180 will be used to indicate towards the originating network that the receiving party’s device is ringing, and that a related early media is also available in an RTP channel.

- In a hybrid Voice over IP-TDM transit network where there is a possibility for a TDM interconnection originated call to be terminated towards a supplier over a Voice over IP interconnection, a ring-back-tone must at all times be delivered towards the TDM-originating network, together with the relevant ISUP signaling (unless otherwise commercially agreed). This means that the transit provider is transparent for a received ring-back-tone (if any) from the upstream supplier, or that it will make available a locally generated ring-back-tone (of the transit provider’s choice).

- In all cases, and by default, there is no guarantee that the ring-back-tone sent will be a country-specific ring-back-tone.

- Encapsulated ISUP information available when using SIP-I is always transparently passed, but is not used by the transit provider related to the management of ring-back-tone, or other early media. This is done on SIP signaling level only as described above.

Schematic overview of the above I3 forum recommendations



Remark: The given consequences in the picture are views <u>from</u> the transit network in the middle.

9 Summary of recommendations for all services migrated from TDM to IP interconnections.

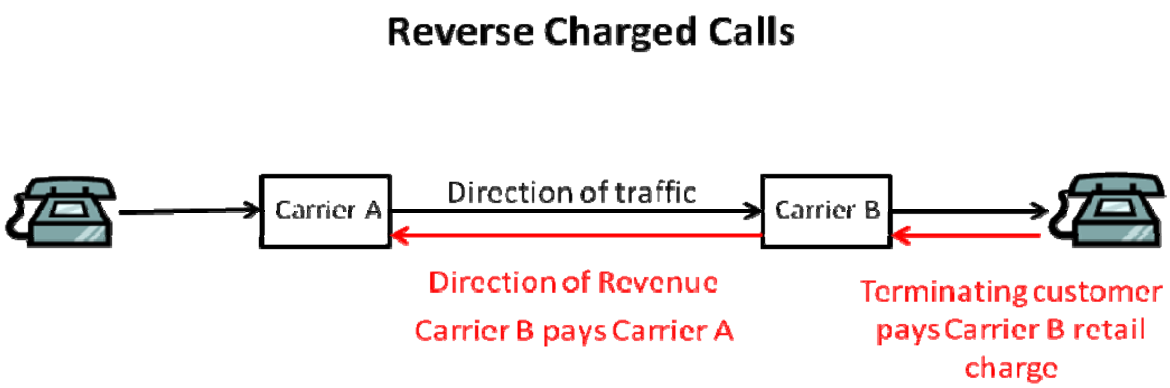
There are currently a large number of voice and related services delivered using legacy SS7/C7 TDM circuit switched architecture that require SIP-I to be delivered via IP. Although these services can still be delivered using a mixed TDM/ IP architecture, there may be a need to ensure that they can be delivered via routes that utilize IP.

Although in the long term the requirement for many of these services (e.g. ISDN, fax and data) will disappear, in the short to medium term, a commercial need to deliver all of these services when migrating to an IP environment may exist for some carriers.

9.1 Specific case of reverse charged calls and special services

This applies to a number of services where the direction of the revenue flow is the opposite of the traffic flow. These services can easily be delivered using IP but require a different billing mechanism.

Special voice services and reverse charged call are not dependant on TDM/ISUP or Voice over IP SIP/SIP-I but the on the IN (Intelligent Network) platform and carrier specific billing system that enable these services. Carriers that want to migrate their special services from TDM to Voice over IP need to ensure that the IN platform and billing system are also available and functioning for Voice over IP calls.



9.2 Current SS7/C7 TDM Services

The table below lists services which are currently delivered using the legacy switched SS7/C7 TDM network and shows the signaling protocols and codecs required to deliver the same services in an IP environment.

TDM Service	Signalling Protocol	Codec
TDM Voice	SIP or SIP-I	Can use many different codecs but to avoid transcoding loss G.711 is preferred
TDM Data (modem)	SIP or SIP-I	G.711
TDM Fax	SIP or SIP-I	G.711 or T.38
ISDN Voice (including ISDN supplementary	SIP-I	G.711

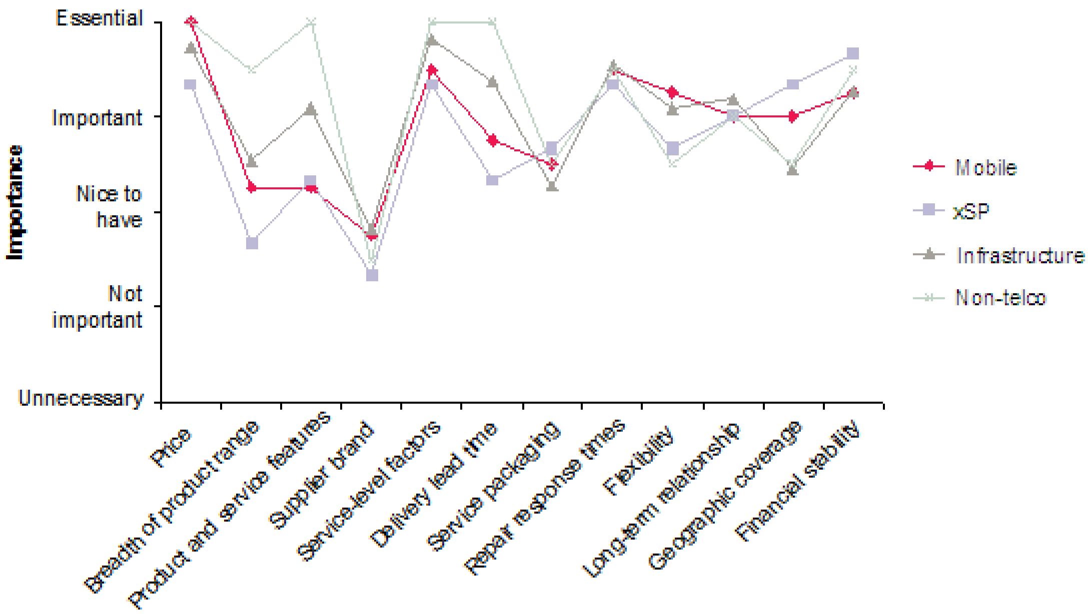
TDM Service	Signalling Protocol	Codec
services)		
ISDN Data	SIP-I	G.711 64 kbit/s unrestricted RFC 4040
ISDN Video	SIP-I	G.711 64 kbit/s unrestricted RFC 4040
DTMF Tones	SIP or SIP-I	G.711 RFC 2833
Text phone e.g. Text telephony for deaf people (T.140 and V.18)	SIP or SIP-I	G.711 – as TDM Data
SMS/Roaming Signalling/Camel	SIGTRAN	
International Freephone and Universal International Freephone (E.152)	As TDM Voice/Data/Fax but reverse charged. Identified through B-Numbers.	
Country Direct (E.153)	As TDM Voice/Data/Fax but reverse charged. Identified through B-Numbers	
Voice VPNs - a voice network which offers the features and characteristics of a private network but in fact is configured this way from a part of the public network.	As TDM Voice/Data/Fax. Identified through B-Numbers	
Operator connected products e.g. collect, station and personal calls (E.140)	As TDM Voice/Data/Fax. Charge in same way as voice to cover network costs and have separate charging arrangements for collect calls.	
International Shared Revenue Service (E.154)	As TDM Voice/Data/Fax. Charged in a similar way to standard voice but have separate charging arrangements at the retail level. Identified through B-Numbers.	
International Premium Rate Service (E.155)	As TDM Voice/Data/Fax. But different charges apply. Identified through B-Numbers.	
Short code calls	where appropriate in an international context but would be handled in a similar way to Voice/Data/Fax and identified through routing digits.	
Various GSMA specified key services e.g. call forwarding, CLI delivery	Presumably these can all be delivered by using SIP-I	

10 Quality of Service for voice

10.1 Needs to enable guaranteed QoS for voice services over IP

International voice customers have always required wholesale voice carriers to provide them with a high level of quality for their premium routes. The industry is seeing increased demand for quality. Some customers are requiring QoS guarantees with corresponding Service Level Agreements. Research suggests that good Service Level Factors will help the customer in the decision-making process. There are a number of factors that influence a purchaser of international wholesale services when choosing a supplier. As one would expect, price remains most important, but it is not the only factor. Increasingly, service-level factors, lead times to delivery and repair response times are mentioned as key elements in the relationships they have with their suppliers.

The findings from an Ovum recent survey of wholesale customers named “Wholesale Customer Survey 2007" published March 2008 demonstrated how important service-level factors, delivery times and repair times are to buyers of wholesale services, as shown in the Figure below.



Source: Ovum 2008

Given such studies and the need expressed by some international wholesale customers, there is a requirement to define QoS parameters that will be commonly used among international wholesale carriers and customers. To facilitate uptake of such indicators, and create meaningful information it is compulsory to have clear and understood definitions and operational principles.

The following sections will formulate the service requirements and the possibilities to measures QoS for wholesale voice in a voice over IP environment and to cascade these QoS values in order to enable the implementation of SLAs or SLOs with back-to-back mechanisms across multiple networks. The purpose is not to recommend any specific QoS levels, but to make possible the implementation of these QoS measurement and reporting for carriers, customers and providers, that might want to use them in their service offering.

10.2 Preferred QoS KPIs for service quality reporting.

Given the large number of Key Performance Indicators (KPIs) that could be used by carriers in determining Quality of Service (QoS) for voice, this section will list and define most commonly used indicators and will select those that are relevant and that add value to the customers. These KPIs must accurately report the quality provided for voice services and enable standard measurements and fault analysis. It should be noted that a carrier could use other KPIs, as they deem necessary in satisfying the commercial demands of their customers.

10.2.1 List and Definitions of QoS Parameters

Service (Voice) Related KPIs	Definition
Answer Seizure Ratio (ASR)	The Answer Seizure Ratio (ASR) expresses the ratio between the number of call session requests and the number of calls effectively answered in a given period of time. Number of billable (answered) calls divided by the number of calls dialed. See “Technical Interconnection Model for International Voice Services ”[14] document for full definition and reference(s) to specifications
Network Efficiency Ratio (NER)	The Network Efficiency Ratio (NER) expresses the ability of a network to deliver a call (measure of network performance) in a given period of time, without taking into account user interferences. Number of successful calls submitted to the next network (ringing information is one sign of successful submission) divided by the number calls dialed. See “Technical Interconnection Model for International Voice Services ” [14] document for full definition and reference(s) to specifications
One Way Speech/Audio	A measurement of the cumulative number of calls where the speech/audio was only detected in one direction in a given period of time. See “Technical Interconnection Model for International Voice Services ” [14] document for full definition and reference(s) to specifications
Average Length of Call (ALOC)	The Average Length of Call (ALOC) expresses the average time in seconds of conversation for all the calls successfully established in a given period of time. See “Technical Interconnection Model for International Voice Services ” [14] document for full definition and reference(s) to specifications
Mean opinion score (MOS) & R-Factor	The MOS (Mean Opinion Score) is a representation of the perceived quality of the call by the end-user rated from 1 to 5 where 5 is the best quality. ITU-T has defined a mathematical model to assess this MOS (Mean Opinion Score) value including lost packets, delay impairments, and codec. This figure is derived from the R-factor whose values in turn are computed according to the same ITU-T specifications. See “Technical Interconnection Model for International Voice Services ” [14] document for full definition and reference(s) to specifications
Fax Quality Ratio	The ratio of successful fax transactions (e.g. fax submitted and received with confirmation) to the total number of fax transactions.
Calling Line Identification (CLI) Delivery Ratio	The CLI is the calling number which in some cases is displayed by the receiving (called) party’s device. The delivery ratio is a ratio of call setups where a Calling Line

	Identification (CLI) was received by the carrier, transported across its network and delivered unmodified to the next network in the chain; divided by the number of total CLIs received from the previous network. See "Technical Interconnection Model for International Voice Services " [14] document for full definition and reference(s) to specifications
Post Gateway Ringing Delay (PGRD)	The Post Gateway Ringing Delay (PGRD) expresses the time elapsed between a request for a call setup and the alerting signal received for that call by the called party. See "Technical Interconnection Model for International Voice Services " [14] document for full definition and reference(s) to specifications

IP transport Related KPIs	Definition
Jitter	The Jitter is the absolute value of differences between the delay of consecutive packets. See "Technical Interconnection Model for International Voice Services " [14] document for full definition and reference(s) to specifications
Round Trip Delay	The Round Trip Delay is defined as the time it takes for a packet to go from one point to another and come back. See "Technical Interconnection Model for International Voice Services " [14] document for full definition and reference(s) to specifications
Packet Loss	The Packet Loss is the ratio between the total lost packets and total sent packets. See "Technical Interconnection Model for International Voice Services " document for full definition and reference(s) to specifications

10.3 Recommended List of QoS KPIs

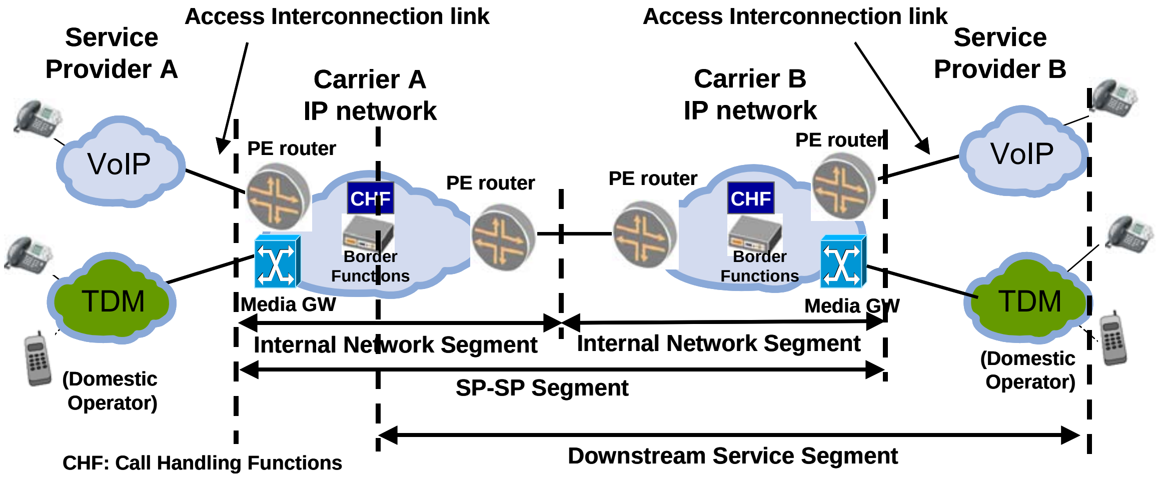
In this section the QoS KPIs identified in the above table will be analyzed based upon:

- The relevance of the KPIs to provide information that impacts and reflects the networks role in the resulting quality of the voice service for the customers.
- The feasibility of these KPIs to be measured, to be agreed and operationally implemented between parties. This may lead to the implementation of SLA/SLO, or as a network specific measurement for information purposes only. The importance in identifying the difference is to ensure that the KPIs inserted into such agreements are fully understood in terms of definitions, as well as to ensure that they are commercially viable KPIs and fully reflect the Quality of Service provided.

All of these parameters are optional and the KPIs to be included in any form of agreement will be part of a commercial negotiation between the involved parties.

10.3.1 KPIs that could be included in a SLA/SLO

The main criteria for consideration are that KPIs are measurable and/or guaranteed across one or several carrier network(s), including the domestic terminating network for the voice indicators. The KPI shall be universally understood and technically measurable and shall exclude any customer/end-user behavior.



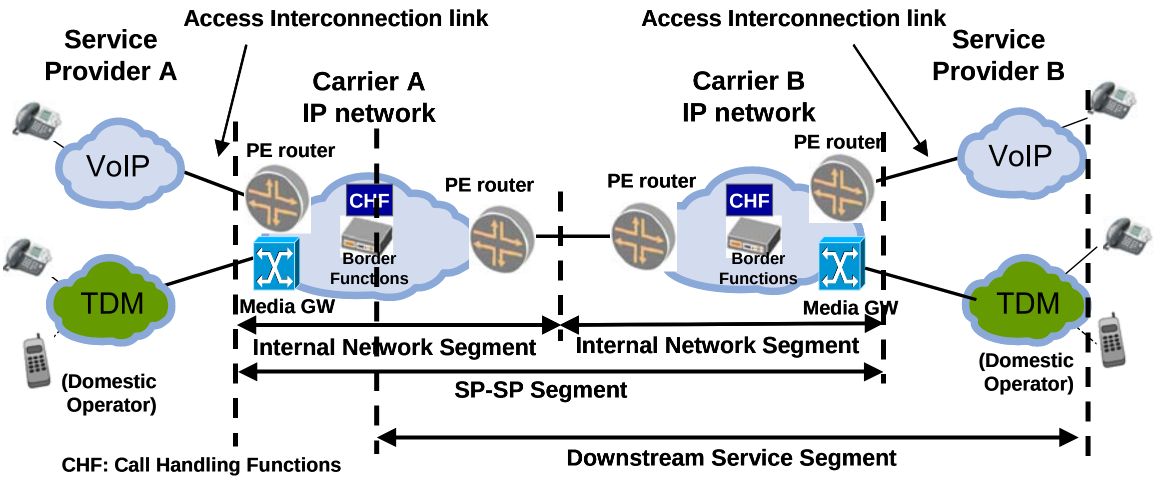
KPIs	Analysis
Network Efficiency Ratio (NER)	The NER measures the effectiveness of the networks to establish calls successfully which have a direct and measurable impact of the quality provided to the customers. Therefore, having a high NER reflects the performance of the carriers' effectiveness in capacity management and engineering of the network. In this case NER can also be perceived as service availability. This KPI can be measured as illustrated in the diagram above as the downstream service segment. See "Technical Interconnection Model for International Voice Services " [14] document for full details on how to measure such KPI
One way speech/audio	Given that it is possible to identify if the RTP packages are transmitted in each direction between originating and terminating networks carriers, it is therefore possible to guarantee that the

	speech/audio is transmitted successfully, which is a necessary requirement for a quality voice call. This KPI can be measured as illustrated in the diagram above as the internal network segment, with the assumption that it also reflects the downstream segment. See "Technical Interconnection Model for International Voice Services " [14] document for full details on how to measure such KPI
Calling Line Identification (CLI) Unaltered Delivery Ratio	Unaltered CLI delivery (CLI Transparency) is a key quality function for customers and it happens to be a reliable and measurable indicator that can be provided by carriers, assuming that the technical systems required are implemented as per the technical recommendations. It should be noted that a Carrier can only transport CLI across its network if the upstream network has presented a CLI. Therefore, a carrier can not guarantee that all calls will have a CLI and can only guarantee to deliver the CLI when a CLI was presented. Delivery means the % of calls with CLIs received from upstream network and delivered unaltered (transparently) to the downstream network This solution can be cascaded up to the last carrier delivering the CLI to the terminating service provider. This excludes the domestic terminating network provider which is illustrated in the diagram above as the SP-SP Segment. CLI probing is another solution that measures CLI delivery from the measuring point (the testing party that does the CLI probing can be anywhere in the chain) up to and including the terminating service provider. However, the result of CLI probing can only be an indicative value of the CLI delivery ratio, because it does not take into consideration the holistic reality of the terminating service provider's network e.g. roaming probes are stationary and covers a limited part of the network only. See "Technical Interconnection Model for International Voice Services " [14] document for full details on how to measure such KPI
Packet Loss Jitter Round Trip Delay	These three (3) IP parameters reflect the quality of the IP transport layer. The transport layer has a direct impact on the resulting voice quality. The measurements of these IP parameters are well defined and are already widely used in IP transport SLAs. The Round Trip Delay, Jitter and Packet Loss can be measured as illustrated in the above diagram as the SP-SP Segment. See "Technical Interconnection Model for International Voice Services " [14] document for full details on how to measure such KPI
Mean opinion score (MOS) & R-Factor	The MOS directly represents the voice quality perceived by the end-users. The reliability of the MOS depends on the portion of the network that it measures. If the MOS is only computed with information from, e.g., 50% of the end-to-end path, it is in this case not reliable and therefore cannot be used for an end-to-end SLA. The portion of the end-to-end path that is measured depends on if and where the RTCP flow is blocked, which is impossible to guarantee between carriers and service providers. As a consequence, the extrapolated MOS is only a reliable measurement under a network-controlled topology, which is why the MOS is recommended to be used only on direct routes, or potentially with a maximum of two carrier networks in the chain. The MOS can also be measured by using external probes which instead of using IP related indicators (R-factor) use the analysis of the voice audio media during test calls establish between calling robots and probes placed in different countries and

	networks. The quality of the measured voice media is compared with the known quality of the pre-recorded voice media played by the probes. While this method can provide relevant MOS measures, it is to be noted that these measures are statistical (one every N calls are tested only), and these measures can only provide relevant information about the MOS quality of the network area where the probe that is called is located. Therefore, the extrapolation of the MOS quality for a destination (network or country) based on a probe can potentially be inaccurate. *Further studies on this solution could be done in a further release of this document.* When measured, the MOS is measured as illustrated in the diagram above as the downstream service segment. See "Technical Interconnection Model for International Voice Services " [14] document for full details on how to measure such KPI

10.3.2 KPIs that should be used for information only

These KPIs are indicators that reflect the overall user-to-user service quality. They are widely used by the industry and are technically measurable. These KPIs are however influenced by the end-user behavior which is out of the carrier's control. The variations of these indicators could reflect the variation of customer behaviors and not of the quality of the networks used. As a consequence these indicators can be used for indicative quality only or for quality management based on historical data comparison, but they are not recommended to be used as part of any SLA/SLO.

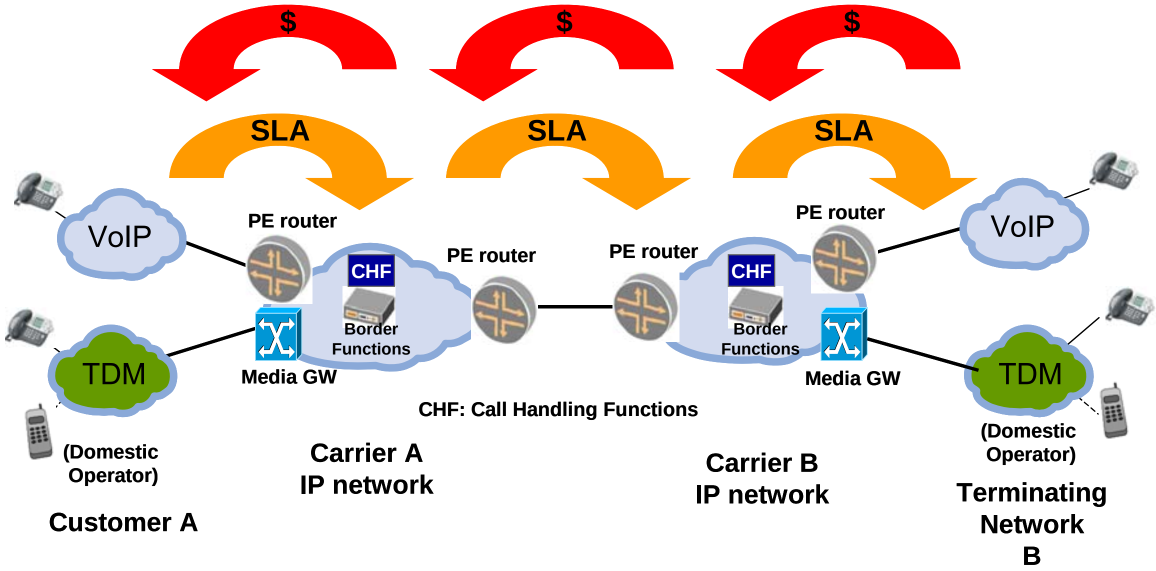


KPIs	Analysis
Answer Seizure Ratio (ASR)	The ASR provides an indication of the networks ability to establish and transport a voice call successfully. The ASR includes user behavior therefore the measurement can only be considered as information only. Example - During certain periods of the year, e.g., holiday's, people are less prone to answer their phones thereby providing a variance of values unrelated to the carrier's network quality. This KPI can be measured as illustrated in the diagram above as the downstream service segment. See "Technical Interconnection Model for International Voice Services" [14] document for full details on how to measure such KPI.
Average Length of Calls (ALOC)	The ALOC provides an indication of the networks ability to transport voice with high quality and customer satisfaction (customers talk longer when the quality is good and hang-up sooner when the quality is not satisfying). But since ALOC includes user behavior, the measurement can only be considered as information only. Example – Business calls e.g. audio conferencing are typically longer in duration than other types of calls, therefore ALOC will be considerable higher for such call destinations and unrelated to the quality of the carriers network. This KPI can be measured as illustrated in the diagram above as the downstream service segment. See "Technical Interconnection Model for International Voice Services" [14] document for full details on how to measure such KPI.
Fax quality	Fax transmission quality is impacted by the quality of the carriers transport network. However, Fax quality also depends on the end-user terminal. E.g. that fax could fail if terminal e.g. runs out of paper and/or buffer it full. Moreover carriers can not identify faxes that use the G.711 codec and as such cannot measure their quality. Therefore a carrier has no way of measuring quality for all faxes transported across its network. Consequently, this indicator cannot be part of an SLA/SLO.
Post Gateway Ringing Delay (PGRD)	The Post Gateway Ringing Delay could be used to measure the downstream carrier network quality related to the call setup between the carrier and up to the service provider, as illustrated in the diagram above as the downstream service segment. However, one can not guarantee that the downstream ringing information has not been manipulated which would falsify the quality reported. Therefore, it is not recommend using the PGRD in a multinetwork SLA as this KPI can be subject to fraud.

10.4 Framework to implement Back-to-Back Quality of Service

To ensure that customers get the quality of service that they require in a SLA/SLO, it is possible to select the recommended KPIs as previously defined. In order to enable this SLA/SLOs across multiple carriers, there is a need to define standard technical and operational procedures that can be implemented by all carriers and customers to measure and implement back-to-back SLA principles.

The diagram below describes the cascading principle where a party in the chain is able to determine and prove that quality degradation is not due to its network and therefore cascades the responsibility downstream to the next party in the chain. In this case the carrier is however still financially responsible to its upstream customer but it can request that same financial responsibility from its downstream provider. This is illustrated in the diagram below as the back-to-back cascading of KPI responsibility.



Practical illustration:

1. Customer A buys from Carrier A the Terminating Network B voice destination. Note that the destination can be a country code or a specific network.
2. Carrier A commits with an SLA to Customer A for a monthly NER of 90% towards Terminating Network B. The penalty if NER goes below 90% is 1000 units.
3. Carrier B commits with an SLA to Carrier A with a monthly NER of 94% towards Terminating Network B. The penalty if NER goes below 94% is 1000 units or higher.
4. Terminating Network B commits with an SLA to Carrier B with a monthly NER of 98% towards Terminating Network B. The penalty if NER goes below 98% is 1000 units or higher.
5. Customer A measures NER of 70% towards Terminating Network B and requests its provider Carrier A to pay the 1000 unit penalty.
   - Option 1: Carrier A measures 94% or more towards terminating Network B, therefore it determines that the degradation seen from customer A is its responsibility and therefore pays the penalty to Customer A and does not request Carrier B to pay any penalties.
   - Option 2: Carrier A measures 68% towards terminating Network B therefore determines that Carrier A is not responsible for the degradation as that the responsibility lies with Carrier B (which fails to meet the 94% requirement) or

further down stream. In this case Carrier A has to pay penalties to Customer A, and Carrier B will have to pay penalties to Carrier A.

- in the case where Carrier A measures 94% or more towards Terminating Network B it is important to assure that the measurements that Customer A performed are valid. The same problem concerning measurement troubles from Carrier A could also occur.

- In case of Option 2 where it is determined by Carrier A that Carrier B or further down the chain is at fault for the degradation of the quality perceived by Carrier B. The same cascading principle will apply between carrier B and the next network in the chain (terminating network B), until the party in the chain that caused the degradation of quality is identified.

In case of disputes about the values measured, **it would be technically difficult as well as very time consuming to solve the reasons that would explain the differences in measurement that led to the disputes.** This situation implies that a good faith approach and technical confidence between parties is necessary. Consequently, this back-to-back mechanism must be well commercially negotiated and only implemented between parties that are willing to do so. It is not possible under other circumstances to envisage SLAs across several networks, as the potential for disputes and even fraud could be worse than the added value generated by the SLAs.

Among all the indicators previously mentioned in Section 10.3 titled “Recommended List of QoS KPI”, some are easier to measure and less prone to disputes than others. For example, IP layer indicators are well known and used in SLAs already. Most of these indicators would be easier to implement for direct routes as they involve less parties in the chain. While theoretically these indicators could be used across multiple networks, the more carriers involved in the chain, the more operationally, extremely challenging and less meaningful of the quality these indicators are.

The further studies of this document will aim at describing operational guidelines on how to implement these indicators and make possible the use of reliable indicators and SLAs across multiple networks.

10.5 Best practices on ensuring Quality of Service management

This section will highlight some best practices, that if followed, would provide a better guarantee of quality between all parties in the value chain.

10.6 Transcoding

Transcoding from TDM to Voice over IP, as well as transcoding between different codecs or packetisation rates within IP, are a few practices that has the most negative impact on the voice quality perceived by the end-user. It is therefore recommended as a best practice to limit transcoding and preferably avoid it wherever possible. *Due to the importance of the topic of transcoding, the next phase of this document will further detail the impacts on voice quality and will describe the technical and/or operational methods that will minimize the need to transcode, as well as provide information and KPIs about transcoding across multiple networks.*

10.7 Quality Management

While some indicators cannot be used to directly reflect the quality of a voice network and its contribution in the end-user perceived quality, these indicators can be used to monitor the variation of quality and target operational action to restore or improve the quality.

For instance, an absolute value of Average Length of Call for a destination is not meaningful. However the comparison of this value with other historical values for the same or identical destinations can reflect an abnormal change in quality. An ALOC that month after month ranges from 200s to 250s and that suddenly drops to 100s could indicate a quality problem in the path.

Most carriers use some sort of internal operational guidelines to use these indicators for quality management.

The next phase of this document will provide some guidelines and options on how to use these informative indicators to detect abnormal quality variances and provide the quality required by the customers.