INTERNATIONAL INTERCONNECT FORUM FOR SERVICES OVER IP

i3 FORUM

Executive summary
June 2, 2008

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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 must 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 Scope and Objective

As far as the technical perspective is concerned, the objective of this i3 forum activity has been the definition, on the basis of existing standards/recommendations specified by international bodies (e.g. ITU-T, IETF), of a unique network architecture capable to support one (or a limited number of) interconnection model(s) for the implementation of trusted, secure and QoS compliant bilateral VoIP interconnection between International Wholesale Carriers. This objective has been complemented with the definition of a migration template and a testing process for an easier implementation of such VoIP international interconnection.

In order to achieve this goal, the scope of the activity has covered all the relevant technical issues considering:

- transport protocols/capabilities;
- signalling protocols;
- media codec schemes;
- QoS levels with measurements and performance needs;
- E.164-based addressing schemes
- Security issues
- Accounting and Charging issues

together with the examination of any regulatory international framework.

The specification of the VoIP and TDM interconnections of the international switching facilities with the domestic networks is outside the scope of this initiative.

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

3 Interconnection Configuration at Transport layer

Assuming a general reference configuration encompassing:

- switching platforms fed with TDM traffic as well as VoIP traffic from the domestic fixed and mobile networks and capable to manage signaling and media information onto an IP transport layer;
- border functions in order to separate IP domains enhancing service and network level of security;
- routing functions according to IP networking;
- transmission functions according to SDH/Ethernet –based systems and protocols;

and considering Public Internet as a global infrastructure, interconnecting managed IP networks, carrying mixed types of traffic with public announced IP addresses, two main sets of configurations are recommended:

- *Private-oriented interconnection*: when no unidentified third party is able to affect the bilateral VoIP service;
- *Public-oriented interconnection:* when the VoIP traffic is mixed with other IP traffic coming from the Public Internet, thus allowing the gateways' interfaces to be reached from unidentified third parties which can affect the service performance and quality.

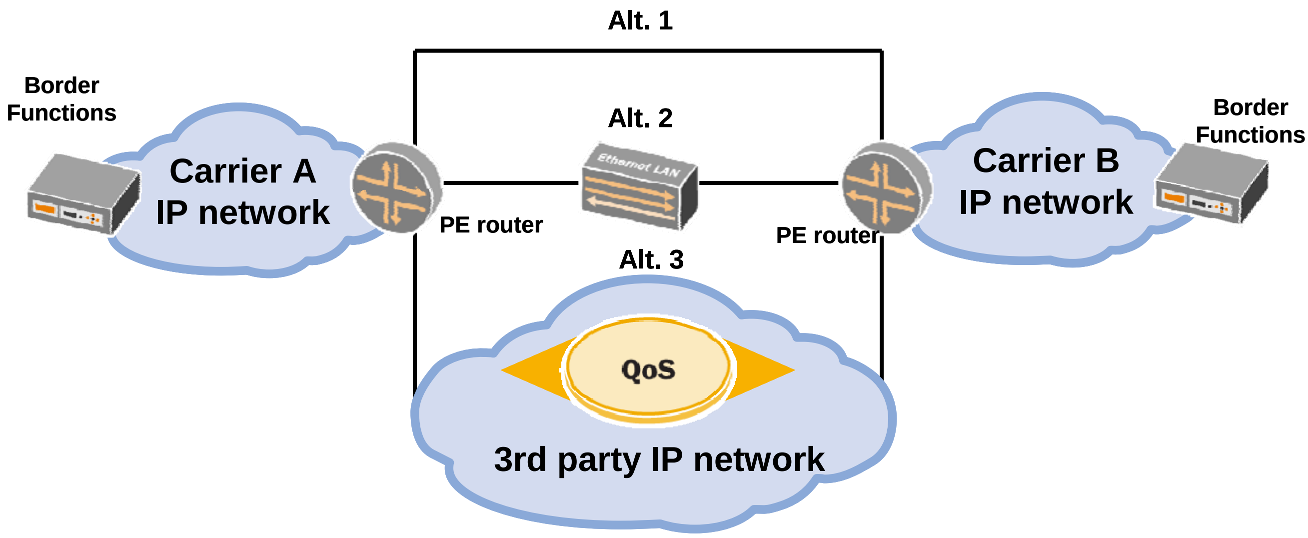
3.1 Transport Functions for Private-oriented Interconnections

In order to retain the private interconnection feature the following conditions have to be satisfied:

1) only VoIP traffic is exchanged across a specific interconnection;
2) all the involved IP addresses can not be reached from unidentified entities via Public Internet;
3) the VoIP traffic, from the PE router to the border functions in a carrier's domain, shall be secured, either physically or logically, from the Internet Transit traffic.

Three different alternatives are possible depending on the type of the adopted link (see figure below):

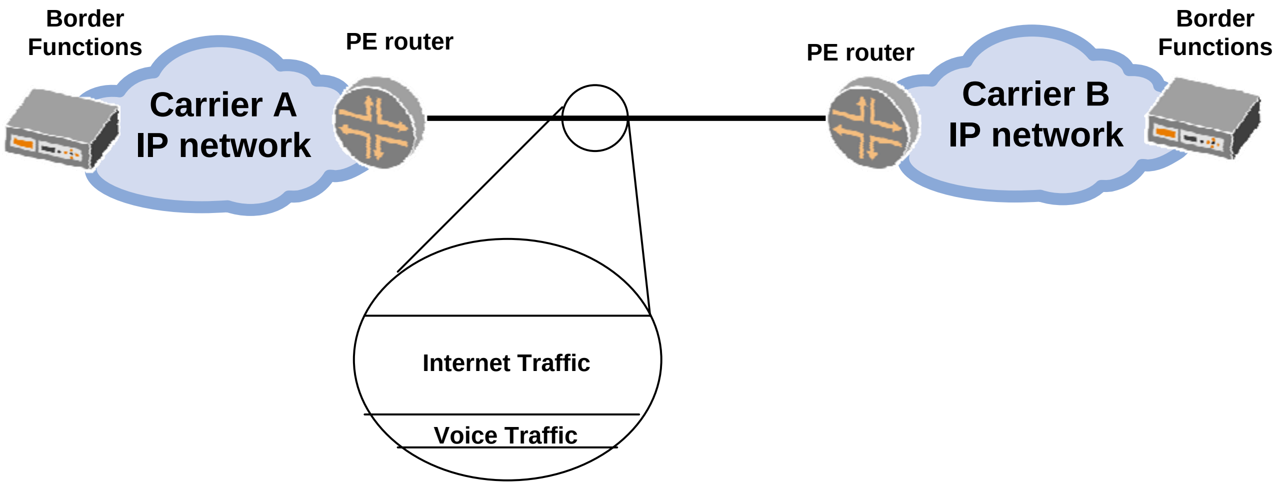
Alt. 1) via a dedicated physical link (layer1)
Alt. 2) via an Ethernet switch (layer 2)
Alt. 3) via an IP network (layer 3)



3.2 Transport Functions for Public-oriented Interconnection

Two alternatives have been envisaged for public-oriented scenario which differentiate each other at the interconnection layer:

Alt. 1) Layer 1 / Layer 2 direct interconnection sharing data and VoIP traffic (see figure below): the IP addresses of the involved PE routers interfaces have to be public and they may be announced over the Public Internet. Border function IP addresses have to be exchanged only by the two carriers.



Alt. 2) Non direct interconnection via Public Internet (see figure below): the VoIP traffic passes through Public Internet i.e. through a third (or multiple) Internet Transit providers. The IP addresses of the PE routers as well as of the Border functions shall be public and they shall be announced over the Public Internet.

3.3 IP Routing and IP Addressing

The usage of the standard IP networking protocols (e.g. BGP) is recommended as well as IPv4 addressing scheme. The support of IPv6 addressing scheme is optional and can be agreed on a bilateral basis.

3.4 IP Packet Marking

Taking into account that different traffic types can cross the interconnection (i.e. Voice Media, Voice Signalling, Internet traffic, other data traffic, control/management traffic) a specific marking of the TOS field of the IP packet is recommended assuming the DSCP and IP Precedence coding scheme.

For the private-oriented interconnection, since voice traffic is highly sensitive to delay and jitter the recommended coding (e.g. DSCP 46 /Expedite Forwarding or IP Precedence = 5 for the transmitted traffic) aims to exploiting the low latency queues of the IP platform minimising delay and guaranteeing better quality.

For the public-oriented interconnection, in addition to the requirement of minimising the packet delay and jitter, security issues have to be considered since traffic is received from the Public Internet. As a result, for the transmitted traffic, the voice signalling and media can be coded either as priority traffic or as best effort traffic.

4 Signalling Functions

Tough several signalling protocols are available on the market, two protocols have been selected as appropriate in this scenario:

- SIP protocol and defined in IETF RCF 3261 and complementing documents.
- ISUP enabled SIP profile as recommended in ITU-T Q.1912.5

Other protocols have been not considered suitable either due to their obsolescence (H.323) or since not fully compliant with the Forum requirements in terms of level of standardisation and more complex interworking with the TDM environment (e.g. SIP-T).

As far as SIP (IETF 3261) signalling protocol is concerned, a specific profile has been defined together with the relevant SIP Header support.

As far as ISUP enabled profile (ITU-T Recc. Q.1912.5) signalling protocol is concerned, the profile specified in Annex C profile C has been adopted.

5 Media Functions

Media functions should assure transport for all the services and transcoding between different codecs. In the scope of this initiative the following services, to be accessible for TDM and VoIP subscribers, have been agreed to be supported:

- Voice phone calls using different codecs;
- Voice conference calls;
- DTMF support;
- Fax connections based on T.38 protocol;
- Modem connections according to ITU-T Recc. V.150.1.

5.1 Voice phone calls and conference calls

For phone calls between two terminals as well as for conference calls with more than 2 participants the following protocol stack is recommended:

- RTP protocol for real time media with its control protocol (RTCP);
- UDP protocol at the transport layer.

The real time media protocol (RTP) that shall be used for international voice services is defined in IETF RFC 3550 “RTP: A Transport Protocol for Real-Time Applications” and complementing RFCs.

Many different coding schemes have been defined, implemented and used for international voice service. In the scope of this activity these codecs have been divided into 2 categories:

Mandatory codecs: provided at least one of the mandatory codecs is present in session description protocol (SDP) offer, and provided at least one of the mandatory codecs is supported by the end side, then codec negotiation is guaranteed to be successful.
Optional codecs: other codecs which are considered with a market relevance.

Group 1. Mandatory	Group 2. Optional
G.711 A-law, µ-law 64 kbit/s	G.722
G.729, G.729a, G.729b, G.729ab 8kbit/s	G.723.1
	G.729.1
	AMR
	WB-AMR

In next releases of this document, other codecs can be considered as mandatory as well as other codecs can be added to the list of optional codecs.

6 Security Issues

Security, both from the network and service perspective, has been considered as a primary requirement for international VoIP interconnection. As a result, it is recommended that all voice traffic coming into / leaving the network operator passes through Border Functions, i.e. all IP packets (for signalling and media), crossing this bilateral voice interconnection, are originated and received by such Border Functions.

Border Functions perform many different features, among them it is worth outlining the role of topology hiding and NAT/NAPT translation. The first feature allows hiding Network Element addresses/names from third parties and it can be implemented by the NAT/NAPT mechanism which is applied at the IP level and is defined in IETF RFC 2663. This IP topology hiding function is carried out for signalling traffic in the IBCF part, and for media traffic in the I-BGF part of Border Functions.

Since voice traffic will be exchanged between Border Functions, the addresses of the Border Functions will be the only visible IP endpoints. The application of NAT/NAPT shall have no impact on the interconnection functionality and shall be transparent to the interconnecting carriers.

Further security can be achieved by encryption technique. Though no encryption is required in case of private-oriented interconnection, it is recommended to encrypt by mmenas of the IPSec mechanism the signaling flow in case of public-oriented interconnection in order to enhance the service security level. No encryption is needed for the media flow.

7 Quality of Service parameters

As far as the quality of service measurement and monitoring is concerned, the following QoS parameters are considered the most important in this first deliverable and they are divided in 3 different sets relevant to the transmission/IP layer, the voice/media quality and the network, respectively. Other parameters can be measured and/or monitored by carriers.

Transmission/IP parameters:
- Bit Error Rate
- RTP round-trip delay
- RTP jitter
- RTP packet loss

Voice/media parameters
- MOS / R-factor for voice quality
- Fax quality

Network parameters
- ALOC
- ASR
- NER
- PGRD

For each parameter the definition is given together with the reference points where the specific parameter should be measured.

8 Numbering and Addressing Scheme (E.164-based)

For this deliverable, the numbering and addressing scheme of voice international call is based on ITU-T Recc. E.164.). These numbers can be used in either in the Tel-URI or SIP URI format.

9 Accounting and Charging capabilities

The information recorded in the Call Data Record is intended to support settlement and performance. As a result, it includes only the data required for these activities. The recommended list of fields addresses neither specific the CDR format nor the collecting method. Each carrier may have additional proprietary fields for internal uses, which is not in the scope of this section.

Since calls may be originated or terminated in TDM or VoIP network, the CDR shall support data attributes for these two types of calls and services.

The recommended list of field encompasses both traditional TDM parameters as well as VoIP-based parameters for performance monitoring (e. g. RTP lost packet, RTP jitter, MOS).

10 Interoperability Test Plan

A Pre-Service Inter-Operability Test Plan has been worked out to test the new transmission path between international VoIP bilateral carriers, ensures signaling compatibly and provides quality and performance levels that meets customer expectations.

The document describes the test approach, specific functionality, assumptions, and test cases that have to be performed for the pre-service inter-operability testing. Test cases in this document will cover calls in both directions between carriers.

Both carriers have to capture and record call traces for each of the test call scenarios and verify that the SIP Messages are in compliance with recommended signaling profile 1. Both carriers will collect and exchange CDRs for the test calls for billing verification.

The following items are not in the scope of this document:

- How the call traces and Call Detail Records are captured
- What equipment is used to capture call traces and CDR

11 Examples of Network Interconnections

On the basis of the content of the deliverables, various interconnection models can be implemented for a bilateral international Voice service depending on the transport configurations, adopted signalling protocol and media codec and additional, interconnection models that imply different quality levels.

In order to better address carriers' needs, it has been recognised useful to complete the specification describing two network examples covering different market scenarios:

1) *direct private-oriented interconnection (dedicated to voice service)*
2) *indirect public-oriented interconnection (via Public Internet)*

in terms of transport configuration, IP protocol parameters, suggested signalling protocol, suggested codec, suggested network security features.

The two given examples of bilateral international interconnection are intended to meet different market requirements.

The first example (private-oriented) describes a possible interconnection configuration to be implemented between two carriers with co-located IP backbones nodes, or that are willing to build a transmission circuit. This interconnection configuration, providing the highest level of quality both in terms of voice call quality, service quality, network availability and network security, can replace existing TDM-based ones and, the more the number of channels is high, the more the suitability of this configuration is high.

The second example (via Public Internet) is more suitable for cases where the two carriers are not co-located and accept the lower quality levels generated by the Public Internet. This interconnection implies a lower cost (resources shared with other services) and, in general, lower provisioning time (no need to set-up an ad-hoc link).

The two examples can both be used to transport International voice traffic, however due to the lower quality levels achievable onto the public internet, carriers that want to provide a high and stable quality of voice services should favour a private and dedicated interconnection solution.

Two tables provide *target values* for the two discussed network scenarios.

12 Contacts

- **at&t**
    - Web: www.att.com/GoWholesale
    - Email: wholesale.product.info@att.com
- **Deutsche Telekom ICSS**
    - i3-forum@telekom.de
- **Orange**
    - i3forum.icss@orange-ftgroup.com
- **Telecom Italia Sparkle**
    - i3forum@telecomitaliasparkle.com
- **Telefónica International Wholesale Services**
    - i3@wholesale.telefonica.com
- **Telekomunikacja Polska**
    - www.tp.pl/carriers
- **TeliaSonera International Carrier**
    - www.teliasoneraic.com
- **SingTel**
    - www.singtel.com