GSMA Packet Voice Interworking

PVI White Paper
Packet Voice Interworking for Mobile Service Providers

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1 Overview

1.1 Purpose

With the introduction of all-IP voice cores in mobile and fixed networks, service providers have the choice for their interconnect connection of either:

- handing the traffic to the carrier for onward routing as circuit switched (TDM) traffic, with the necessary codec conversion at the edge of the network
  or
- passing it as IP traffic and leaving the conversion to be carried out at a later stage in its onward transmission to the distant end.

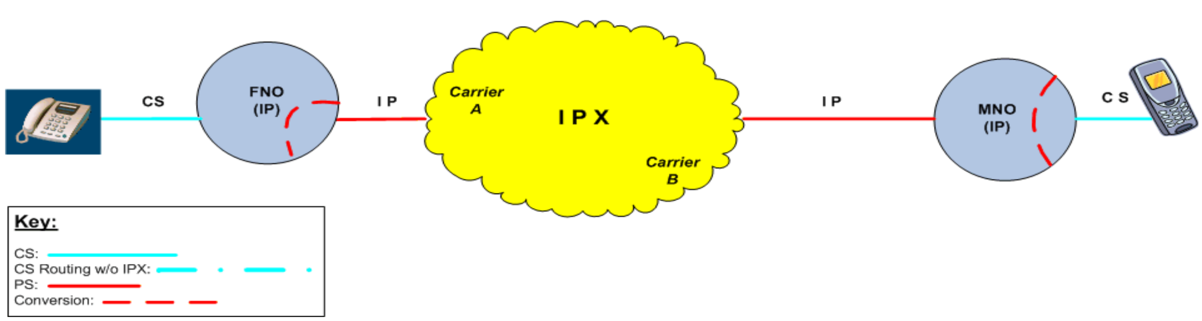
Without clear policies and rules as to where and how this should be done, end to end quality will vary considerably as a result of too many conversions and may be unacceptable to the customer.

The purpose of this paper is to:

- Identify the commercial and technical issues that arise on interconnection with the introduction of all-IP core networks by both fixed and mobile network service providers and the need for a structured and agreed approach to be taken
- Provide suggested solutions to the issues raised
- Recommend next steps and the roadmap to such solutions
- Address the issue of a standard protocol for interworking for IP based voice

1.2 Executive Summary

Given the current uncoordinated way in which service providers and carriers are changing to an IP infrastructure, the risk of varying voice call quality as a result of too many conversions in the voice call is considerable. The current envisaged method to minimize this is to convert traffic to TDM at the edge of the network. A preferable method would be to route IP originated traffic as far as is possible as IP and to only convert to TDM at the far end if it is to be passed to a TDM network, resulting in a maximum of two conversions – the first between the RAN and core network, and a second conversion to TDM at the far end if required.



Note: Diagram does not depict the exact conversion point in the Fixed Network Operator (FNO) network.

This has a number of advantages:

- Minimizes the number of times conversion takes place and thus minimizes end to end latency, while maximizing Quality of Service (QoS)
- Reduces the infrastructure cost for service providers for interface equipment. The cost and need to provide conversion is eliminated. Such equipment would inevitably become redundant as all carriers and service providers move to IP
- Routing between service providers would require a managed QoS based interconnect IP network. This will stimulate the provision of such a network, which will in turn provide the foundation for the interconnection of other types of service providers and services.

Inherent in such a strategy is the determination of the necessary interface for voice IP interworking and the associated protocol. To date some mobile service providers have invested in BICN (Bearer Independent Core Network) and the use of BICC (Bearer Independent Call Control) within their networks. This standard has not been proposed for fixed service providers.

This White Paper proposes that SIP-I, SIP with encapsulated ISUP, be used as the interconnect signalling protocol for the IPX, the GSMA's recommended interconnect network for IP based services. Where mobile service providers have already deployed BICC as part of the BICN, they have the option for future migration to SIP-I, to deploy a BICC to SIP-I IWF or for the IPX to support SIP-I interworking (interconnect using BICC is not excluded).

This document outlines various solutions, with an overall recommendation for the transition from TDM to IP for mobile service providers and the carrier networks that interlink them. The recommendation is for traffic to be routed as far as possible as IP via the IPX and only to be converted to TDM at the far end if it is to be passed to a TDM network. This document also redefines the target call control protocol for BICN (mobile networks) as SIP-I for interconnect to the IPX. It is recognised that for those mobile service providers that have already invested in BICN, BICC may be used nationally for interworking. However, it is unlikely that BICC would be used internationally for interworking with mobile service providers, fixed service providers or carriers.

The recommendations of this White Paper represent an intermediary step towards the end goal of the roll out of IP Multimedia Subsystems (IMS). It is also recognised that some mobile service providers may skip this intermediary step and move straight to IMS, which enables the provision of enhanced voice services by direct use of the 3GPP SIP profile.

While this strategy minimises delay and helps to maintain QoS, it should be noted that there are operational challenges associated with the roll out of IP core networks. Many service providers are moving from circuit switched networks to IP networks with the intention of reducing OPEX. However, real time services require additional capacity, which implies additional infrastructure capex costs.

2 References

[1] 3GPP TS 23.205: ‘Bearer-independent circuit-switched core network’

[2] ITU-T recommendation Q.1912.5

[3] 3GPP TS 23.153: ‘Out of band transcoder control’

[4] 3G Americas (2007): *“Why SIP-I? A Switching Core Protocol Recommendation for GSM/UMTS Operators”;* http://www.3gamericas.org/

3 Introduction

Voice over IP (VoIP) is a term that is used in many areas of the telecoms industry and will continue to become an ever-more significant focus of service provider (SP) strategy. VoIP can manifest itself in many different ways, but the initial focus for most SPs, both fixed and mobile, is on the replacement of existing monolithic switches with an IP ‘Softswitch’ architecture. For the mobile industry, Softswitch architecture means the Bearer Independent Core Network (BICN) as defined by 3GPP in Release 4 (see 3GPP TS 23.205 [1]).

This White Paper considers the implications of this industry trend towards Softswitch architecture, with particular focus on the synergies and differences between the mobile and fixed line service provider approaches, the technical solutions that are available, the possibilities to improve the environment technically and how this movement will impact the interconnection of service providers as they move from TDM towards IP.

Furthermore, it is envisaged that this approach will lead to an improved level of quality for voice calls over a well managed quality network. This same IP inter service provider network can be used to carry other IP based traffic in the future including many of the IP services (for example, Video Share) that are being developed by the industry in accordance with the GSMA’s IPI strategy.

For further information on IPI and IPX, please refer to GSM World: http://www.gsmworld.com/ipi/index.shtml

IPX Specifications are available at; http://www.gsmworld.com/ipi/documents.shtml

4 Assumptions

This White Paper is based on the following assumptions:

- Core networks will transition from TDM to IP
- Customers require and value QoS

- Conversion to/from IP/TDM deteriorates quality

5 IPI and IPX overview

The GSMA's IP Interworking (IPI) initiative is based on the following key principles:

- Openness – it is entirely open to any potential service player in the delivery of IP services
- Quality – high quality voice and data services will be delivered end to end
- Efficient Connectivity – bilateral and multilateral connectivity will simplify end to end IP service delivery
- Cascading Payments – value and revenue can be cascaded through the delivery chain ensuring participants receive value for their investment and deliver their obligations which are also cascaded through the value chain
- Security – the IPX is a Secure Managed Network with security rules and compliance

The IP Packet Exchange (IPX), proposed by the GSMA, is a managed network, which provides interconnection of IP Services between trusted end parties as part of the realization of the GSMA's IPI initiative. The IPX is the core of a private global trusted environment for the exchange of IP traffic in a scalable and secure way supporting end-to-end QoS and the principle of cascading interconnect payments. In order to provide these features the IPX is service aware unlike the Internet and the GRX.

The IPX environment will consist of a number of IPX providers operating in open competition, selling interconnect services to service providers. Service providers include all providers of IP based services who wish to avail of the advantages, and agree to the principles, of the IPX. This can include mobile service providers, fixed service providers, ISPs, ASPs, cable providers, and so on. By service providers agreeing to the principles of the IPX, this can ensure it remains an open, yet trusted environment. The IPXs will be mutually interconnected where there is demand from service providers. Both IPX providers and service providers will participate in and be subject to IPX governance to ensure the quality, security and the technical and commercial principles of the environment are maintained.

The IPX may be used to interconnect any IP service. To provide interoperability (commercially and technically) amongst the community of service providers the interconnect aspects of the services must be standardised. An IPX is free to support any of the services they consider viable. A service provider may negotiate interworking with any IPX on a per service basis. An IPX may offer other non-standardised value-added services to requesting service providers, provided they do not conflict with the operation of the standardised IPX services.

End-to-end quality is a key requirement for the IPX environment. Service providers and IPX providers alike will be subject to Service Level Agreements (SLAs) on a per service basis. The SLAs will be enforced by the principle of cascading responsibility, governance and by contracts.

The IPX architecture consists of a transport network capable of providing end-to-end QoS, for example, the conversational traffic class in a private network.

Security is provided by establishing a separate routing domain from the Internet and by the separation of traffic within the IPX. Peer-to-peer traffic is separated from server-to-server traffic as the former is considered a higher security risk. Security also relies on the establishment of a trusted environment whereby all connecting parties implement robust security in their own networks.

The recommendation to minimise the number of signalling and codec conversions is to route traffic via the IPX and provide clear requirements to the IPX to ensure QoS can be maintained. In the next 2-3 years, as networks migrate to next generation IP networks, it will become increasingly important to enable service providers to connect to each other over IP interconnects.

The IPX is the foundation of the model we envisage for the interconnection of IP core networks deployed by Mobile Network Operators (MNOs), Fixed Network Operators (FNOs) and other IP service providers. The terms of usage of the IPX encourages a maximum of 2 packet switched (PS) to circuit switched (CS) conversions in any call scenario.

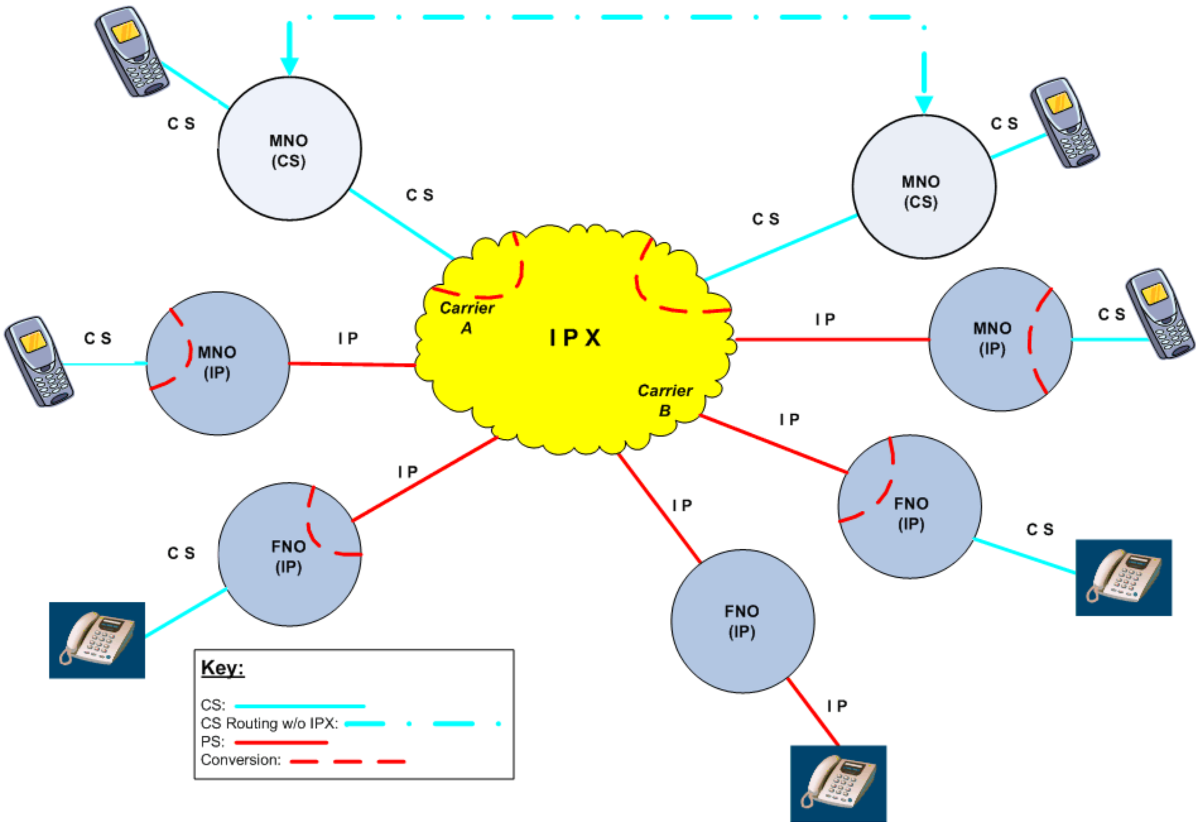


Figure 1: IPX Overview

*Diagram is for illustration purposes only; there will be cases where traffic is routed on from one operator to another and there may be cases of non IPX providers involved in delivering FNO to FNO traffic. Note: diagram does not depict the exact conversion point in the FNO network.

For more information on the GSMA's IPI initiative and the IPX please see:

http://www.gsmworld.com/ipi

6 Advantages of voice over IPX

6.1 To Mobile and Fixed Service Providers:

- This approach guarantees a maximum of two conversions end to end regardless of the number of parties involved in routing the call, resulting in improved service quality
- The service provider can be assured that the best possible quality will be obtained since the call will be carried as IP to the distant end across the IPX ensuring SLA commitments for end to end quality are met
- Current cascade billing and settlement rules can be maintained to ensure continuity of existing interconnection agreements

- By having a complete end to end strategy it is possible to obtain the advantages above but at the same time ensure that we continue to promote general interoperability of services between service providers, utilizing commercially sustainable interconnection arrangements
- The IPX may also support new commercial interconnect charging models for new services (for example, revenue share, called party pays, usage based, and so on), as may be appropriate for that service)
- Multilateral connections via the IPX ease the technical, commercial and administrative burden placed on mobile service providers to establish and manage multiple bilateral connections
- Multiple services can be inter-connected over a single connection
- Without this strategy service providers who have deployed an IP core network will have two conversion points on their networks; at the point between their TDM access network to their IP core and at the exit of their network to the carrier. Use of the IPX will remove the conversion point at the exit of their network with a resultant saving in cost and latency; additionally, this interface equipment would become redundant as the handover point will inevitably become IP in the future
- It will be a proving ground for the fixed service provider to understand and evaluate the benefits of IPI and the IPX for other products and services

6.2 To IPX Providers:

- While many IPX providers can see the advantages of the IPX, the business case is difficult to justify on the basis of new products and services that the industry is planning to launch, which are both risky and speculative. Voice traffic is far more definable and is, over time going to become IP based.
- Gives the IPX providers considerable economy of scale and justification to evolve their networks from TDM to IP
- Opens up new opportunities for IPX providers to provide conversion for service providers between TDM and IP
- Provides opportunities for value added services such as trans-coding, providing inter-operability, routing and number portability
- IP skills become more pervasive which eases recruitment of staff
- Multiple services can be inter-connected over a single connection

6.3 To Customers:

- Minimizes end to end latency, helping to maintain QoS
- The IPX provides us with the platform to promote the efficient interoperability of services between a greater number of service providers, thus enabling

service providers to continue to expand the interoperable communities their customers can connect and communicate with

- Existing retail charging models may be maintained (for example Calling Party Pays or Receiving Party Pays) meaning that the customer understands the charges and only pays for what is valuable to him/her
- Improved security and spam Protection: the IPX provides isolation from Internet security issues, and provides accountability and traceability for spam and security issues

7 Signalling Protocols for Interworking

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7.1 Control Plane Differences

Mobile service providers are moving their core networks from the current circuit switched network to one of an all-IP Softswitch design, an IP core network. Similarly fixed service providers are also moving to an all-IP core networks.

However, in the control plane, fixed networks and mobile networks show a clear difference. Fixed networks use SIP-I (see ITU-T recommendation Q.1912.5 [2]) as the basis of their control plane protocol whereas mobile networks might use BICC with mobile specific extensions (see 3GPP TS 23.205[1]). The mobile specific extensions to BICC are to support Out of Band Transcoder Control (OoBTC), which is an essential enabling technology function for Transcoder Free Operation (TrFO) as described in 3GPP TS 23.153[3].

This issue is significant on two levels. Firstly, even without the 3GPP-defined extensions to BICC, there is currently no defined interworking between BICC and 3GPP SIP-I, which means that in order to perform a BICC to 3GPP SIP-I conversion, fall back to ISUP is generally required. Work on interworking between BICC and 3GPP SIP-I will be documented in 3GPP TS 29.235, but this work is not yet completed. Secondly, in many cases, mobile-to-mobile calls traverse at least one transit network. While these transit networks are also moving in the direction of Softswitch architecture, they are using SIP-I call control which means that even though both mobile networks may be TrFO enabled, there is no possibility to gain benefit from this because the transit network is not.

Because fixed networks and transit networks both use SIP-I as the control plane interface protocol, mobile networks using BICC look increasingly like islands. This impression is reinforced when considering that IMS uses a SIP profile and enterprise systems have been SIP based for some time. While the SIP profiles used in IMS and enterprise differ considerably from SIP-I, there is still some basic functionality that can be easily replicated. Hence, BICC is a problem.

Three possible solutions exist to resolve this misalignment;

1. Redefine the call control protocol for BICN as SIP-I with suitable extensions to SIP-I to support TrFO
2. Define the interworking of BICC to SIP-I and define TrFO extensions for SIP-I to traverse a transit network
3. Maintain the current TDM interconnects between service providers

It should be noted that many transit networks already transport native TDM traffic over IP, and so in option 3, while maintaining TDM interconnects might technically seem like an option, there are in fact conversion points between TDM and IP and back again already in place. Hence moving towards an IP end-to-end interconnect will result in considerably less delay, less transcoding and a reduction in other bearer manipulation.

Option 2 does not require existing BICC networks to alter their implementation, while also accommodating TrFO through the extensions to the SIP-I profile. However, there is still a requirement for a complex Interworking Function (IWF) at the edge of the mobile network and a considerable amount of standards effort to define that IWF to maintain the TrFO signalling across the conversion point, and also to define the TrFO transport in SIP-I itself. This functionality is currently being defined in 3GPP Release 8 (TS 29.235).

Since TrFO would need to be defined for SIP-I for Option 2 to be successful, this would make Option 1 possible as well, where Option 1 would remove the need for an IWF completely. The decision of whether to go with Option 1 or Option 2 is really one for an individual service provider to make based on what they have deployed. However, the net result is that a TrFO enabled SIP-I interconnect would be an industry wide initiative.

Note: this paper does not cover ISUP variants of SIP-I or BICC.

7.2 Benefits of SIP-I Based Interconnect over BICC

The benefits of moving to a SIP-I based interconnect with TrFO enabled are considerable. There are benefits to be found for mobile service providers, fixed service providers, transit providers and vendors. These are summarized below:

For mobile service providers

- Possibility to have end-to-end TrFO/OoBTC call control even when traversing transit networks
- If fixed network supports TrFO/OoBTC, removal of many transcoding points providing further improvement in QoS through less transcoding and reduced delay
- Commonality between SIP-I and the 3GPP SIP profile will ease the transition from SIP-I to use of the 3GPP SIP profile in the IMS

For fixed service providers

- Removal of many transcoding points providing further improvement in QoS through less transcoding and reduced delay
- Provides a mechanism for negotiating codec out of band at call setup

For transit providers

- Removal of requirement for large amounts of transcoding resource in network nodes
- Provides additional TrFO functionality that can be used as a selling point to mobile and fixed service providers in transit networks

For vendors

- Greater commonality between fixed and mobile product lines, thus reducing overall R&D costs
- Commonality between SIP-I and the 3GPP SIP profile

8 SIP-I and the IPX

The IPX is the foundation of the model we envisage for the interconnection of IP core networks by mobile service providers. Voice is also the key driver for the IPX. The interconnect technology proposal outlined in the sections above is to use SIP-I with TrFO as the control plane protocol for all BICN calls and hence for all interconnection between different network service providers (be they mobile or fixed).

The IPX is intended to support IP traffic in a way that allows any service to utilize it, a packet voice interconnect for voice trunks with TrFO enabled SIP-I as the control plane would be a suitable service for the IPX to support.

This makes SIP-I based interconnect a leading candidate for a service that should be supported by the IPX.

It should also be noted that while SIP-I is the recommended protocol for voice signalling in the IPX, other packet based signalling protocols such as BICC, SIP-T, the 3GPP SIP profile in the IMS may also be supported. Inter-working between these other protocols may also be supported in the IPX as well.

9 Recommended solution

- Mobile service providers should agree that traffic should be routed as far as possible as IP via the IPX and should only be converted to TDM at the far end if it is to be passed to a TDM network

- the 3GPP SIP profile in the IMS represents the end goal call control protocol for interconnect to the IPX
- As an intermediary step, mobile service providers should agree that SIP-I should be utilised as the call control protocol for interconnect to the IPX. If available, the IPX should also support BICC interconnect as well as BICC to SIP-I interworking
- To enable interconnection between (at least) mobile service provider BICN networks, it is recommended that the IPX should be designed to support SIP-I based call control with TrFO

The GSMA welcomes and acknowledges the contribution from the 3G Americas organization, which also identified SIP-I as the protocol of choice for voice inter-working in its White Paper [4].

10 Timeline and Roadmap

SIP-I Standards

Work on defining a SIP-I based definition for the Nc interface with support for TrFO was initiated in 3GPP in May 2006. The technical specifications resulting from this (TS 23.231 and 29.231) were completed in the first half of 2008. Work on defining the inter-working of SIP-I to external networks (ISUP, BICC, IMS and other external SIP-I networks) is ongoing and is expected to be completed in 2008.

SIP-I Service in Marketplace

While the 3GPP specification work is being done, it is expected that products and services to support this will be developed and deployed in core networks and also in transit providers in parallel. This can be done as SIP-I is a well understood protocol and products based on ITU-T's Q.1912.5 standard will support basic voice inter-working while the mobile specific SIP-I specifications in 3GPP are finalised. Trials based on this assumption have been undertaken by the GSMA already (see below). Furthermore, the GSMA has already developed the inter-working and service level agreements to support voice inter-working over the IPX.

Voice inter-connection using SIP-I is still at the pre-commercial implementation stage, however, as service providers and IPX providers gather more knowledge and experience in this area and product offerings to support this become more mature; it is expected that we will see circuit-switched voice inter-connect migrate to packet-switched voice inter-connect starting in 2009.

11 SIP-I Inter-Working Trials

In 2007, the GSMA initiated trials to verify SIP-I inter-working between GSM service providers across the IPX. The trials have continued into 2008 and have successfully demonstrated basic inter-working between 9 [1]fixed service providers and mobile service providers using 8 [2]different IPX providers with core network platforms from 4 [3]different vendors. These trials were designed to allow service providers and IPX providers to begin familiarizing themselves with SIP-I inter-working over the IPX and focused on basic call scenarios, supplementary services, CDR generation and QoS. As specification work for SIP-I in GSM environments is still underway the core network platforms in these trials supported SIP-I inter-working based on the ITU-T specification Q.1912.5. Nevertheless, the trials demonstrated a high-level of inter-operability and the feasibility of using SIP-I as the voice inter-working call control protocol of choice.

12 Conclusion

The introduction of all-IP core networks by both fixed and mobile service providers will result in varying call quality unless carefully managed. Traffic should be routed as far as possible as IP and should only be converted to TDM at the far end if it is to be passed to a TDM network. Moving towards an IP end-to-end interconnect (avoiding TDM interconnect except where required at the far end) will result in considerably less delay, less transcoding and less other bearer manipulation. The overall quality of the call will improve as a result.

Voice is the key driver for the IPX and as a result the IPX should support SIP-I based call control. This does not preclude the IPX supporting other SIP based call control protocols.

We envisage that service providers will migrate their existing CS core network towards BICN and recommend that the call control protocol that they use for interconnect to IPX is SIP-I. Where service providers have already deployed BICC as part of the BICN, they have the option for future migration to SIP-I, to deploy a BICC to SIP-I IWF or for the IPX to support SIP-I interworking (interconnect using BICC is not excluded).

The main issue associated with the transition from TDM to IP core networks is the potential increase in the overall end-end voice transmission delay. The strategy detailed above will result in less delay and minimal conversions, helping to maintain QoS.

[1] Vodafone, Telefonica, TeliaSonera, Telecom Italia/TIM, Proximus, Elisa, DTAC, British Telecom

[2] TeliaSonera International Carrier, Belgacom International Carrier Services, Telefonica International Wholesale Services, Telecom Italia Sparkle, Telenor Global Services, iBasis, Reach, Cable and Wireless

[3] Nokia Siemens Networks, Ericsson, Alcatel-Lucent, Italtel

Annex A: Glossary

BICC	Bearer Independent Call Control
BICN	Bearer Independent Core Network
FNO	Fixed Network Operator
IMS	IP Multimedia Subsystem
IP	Internet Protocol
IPX	IP Packet Exchange
ISUP	Integrated Services Digital Network User Part
IWF	Inter-working Function
MNO	Mobile Network Operator
OPEX	Operational Expenditure
OoBTC	Out of Band Transcoder Control
PVI	Packet Voice Interworking
QoS	Quality of Service
SLA	Service Level Agreement
TDM	Time Division Multiplexing
TrFO	Transcoder Free Operation
VoIP	Voice over Internet Protocol