

The Essential Report on IP Telephony

**BY THE GROUP OF EXPERTS
ON IP TELEPHONY / ITU-D**

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ITU E-Strategies Unit

The Essential Report on
IP Telephony

by the Group of Experts on IP Telephony / ITU-D



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PREFACE

The possibility of transmitting voice over IP-based networks, with all the challenges and associated opportunities, such as voice and data integration, constitutes a milestone in the convergence of the ICT sector. The "IP telephony" topic has been taboo for both its supporters and its opponents, two camps that are highly divided. After due consultation with the Directors of the ITU Telecommunication Standardization Bureau and the ITU Radiocommunication Bureau, BDT undertook to put the issue on the table for discussion in accordance with the terms of Opinion D Part 3 (see Annex O) adopted by the third World Telecommunication Policy Forum (Geneva, 7-9 March 2001).

Considering that important technical, socio-economic and policy issues need to be addressed by developing countries in order to introduce IP telephony, WTPF-01 invited ITU-D to prepare a report to the World Telecommunication Development Conference (WTDC-02) in order for this conference to take the necessary action.

I established the relevant Group of Experts to carry out the defined tasks in order to facilitate the introduction of IP telephony, including inter-operability considerations and implications, when implementing IP telephony in association with the existing circuit-switched national and international telecommunication networks in developing countries.

Today, we are pleased that the two camps are getting closer, and many difficult questions have been raised and answered. BDT's roles of catalyst and as a vehicle for information dissemination are essential to address issues of this nature, and we will continue to do so in the future.

In response to the need to provide administrations of developing countries with guidance on strategies to facilitate the introduction of IP telephony, the "Essential Report on IP Telephony" was prepared by the Group of Experts from both developing and developed countries, Member States of ITU and ITU-D Sector Members, chaired by Mr Nabil Kisrawi.

The work of the Group of Experts on IP Telephony has already generated interest within the regular studies and technical assistance of BDT concerning the use and management of IP-based networks in developing countries. I would like to take this opportunity to thank Chairman Nabil Kisrawi for his invaluable support and initiatives taken over the past months that have enabled us to overcome major obstacles. I also wish to thank all the experts and their respective administrations and companies for their fruitful contributions.



Hamadoun I. Touré

Director
Telecommunication Development Bureau

FOREWORD

Internet Protocol (IP)-based networks were recognized by the Plenipotentiary Conference of Minneapolis, 1998, in its Resolution 101, as an issue of crucial importance to the future, as an important engine for growth in the world economy in the twenty-first century, stressing the need to identify the implications of the development of such networks in ITU Member States, and this to include interoperability issues between IP-based networks and other telecommunication networks as well as how to provide the quality of service required by the users.

Data traffic is growing rapidly compared to voice traffic and consequently, the past concept of telephone networks that also carry data might be replaced (when and how?) by the concept of data networks that also carry voice.

By Decision 498, the 2000 session of the ITU Council decided to convene the third World Telecommunication Policy Forum (WTPF-01) in Geneva, from 7 to 9 March 2001, in order to discuss and exchange views on the theme of Internet Protocol (IP) Telephony. Having in mind the "IP telephony" challenges in developing countries, Opinion D was adopted by the Forum. Opinion D was supposed to answer many challenges and issues facing developing countries, in particular those facing many public (or privately dominant) telecommunication operators when "IP telephony" is introduced, such as:

- its impact on their revenue streams, resulting from lower-priced "IP telephony" tariffs compared with their PSTN tariff schemes
- how not to place any additional requirements on PSTN networks when interconnected to IP-based networks
- how to meet the performance metrics, and traffic identifications when IP-based networks interwork with PSTN
- how to generate the necessary funds to invest in IP-based networks
- how to deal with numbering and addressing issues

The conclusions and main issues on "IP telephony" derived from this report represents the answers to many of these challenges as well as the answers to the tasks enumerated in Part 3 of Opinion D.

I would like to take this opportunity to thank all Experts and Rapporteurs for their hard work and to thank Mr Hamadoun I. Touré, BDT Director, and his staff for their support to the Experts Group.



Nabil Kisrawi

Chairman of the ITU-D Group of Experts on
"IP Telephony" related to Opinion D Part 3

ACKNOWLEDGEMENTS

ITU-D would like to express its gratitude to the members of the Group of Experts on IP Telephony for their excellent work and tireless efforts in preparing this report.

The text of the report was prepared by a group of experts, led by the Chairman of the Group, Mr Nabil Kisrawi (Syrian Telecommunication Establishment), and assisted by Vice-Chairman Mr Peter Kenduiywo (Telkom Kenya Ltd). Overall coordination was carried out by Mr Désiré Karyabwite, IP Coordinator, of ITU/BDT's E-Strategies Unit.

The Group divided into various Rapporteur Groups, namely: the Rapporteur Group on Technical Aspects, led by Mr Jamel Zenkri (Tunisia), assisted by Mr Souheil Marine (ALCATEL France); the Rapporteur Group on Economic Aspects, led by Mr Sameer Sharma (Telecom Regulatory Authority of India), assisted by Mr Kumar Jayant (India); the Rapporteur Group on Policy Issues, led by Ms Virginia Sheffield (USA), assisted by Ms Julie Kearney (Federal Communications Commission, USA); the Rapporteur Group on Workshops and Training Aspects led by Ms Rosa Rizvangoul Cissé (SOTELMA, Mali), assisted by Mr Désiré Karyabwite; and the Rapporteur Group for the Checklist of Factors for the introduction of "IP telephony", led by the Chairman of the Group of Experts.

In addition, the report has benefited from the input and comments of many experts to whom we owe our thanks. In particular, we would like to thank:

Ms Fiona ALEXANDER, Department of Commerce (USA)
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Mr Nangithia MBOGORI, Telkom Kenya Ltd
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Mr Hussein Ahmed Mohamed OSMAN, SUDATEL (Sudan)
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Mr Barka Koigoumo TOURÉ, SOTELMA (Mali)
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Furthermore, this report would not have been possible without help from E-Strategies Unit members, in particular: Mr Alexander Ntoko, Ms Christine Ochienghs, Ms Martine Métral and Mr Efrem Yosef, to whom we owe our thanks. We would also like to express our appreciation to Ms Renée Zbinden (ITU's Publication Composition Service) and her team for producing the report, with special thanks to Nicolas Stauble for the cover design.

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PART I

GENERAL CONSIDERATIONS ON THE INTRODUCTION OF IP TELEPHONY

Chapter I.1 – Introduction to IP telephony considerations

I.1.1 Background

In accordance with Opinion D part 3 (Annex O attached) adopted by the third World Telecommunication Policy Forum on IP telephony (Geneva, 7-9 March 2001), Mr Hamadoun I. Touré, Director of BDT, invited experts from developed and developing countries to plan a strategy of migration for their networks into IP-based networks. In response to his invitations, three meetings of experts on IP telephony chaired by Mr Nabil Kisrawi from Syria, assisted by Mr Peter Kenduiywo (Kenyan Expert) as vice-chairman, took place (9-10 July; 8, 9 and 10 October and 13 to 14 December 2001).

These Experts Meetings were organized to carry out the following tasks identified in Opinion D as related to ITU-D:

- a) to prepare, as soon as possible, a checklist of factors which developing countries may use in the process of accelerating the introduction of IP networks, thus facilitating the introduction of IP telephony;
- b) to advise and assist in response to the concerns and needs of the developing countries on the technical, socio-economic and policy impacts on the introduction of IP telephony;
- c) to prepare a report to the forthcoming World Telecommunication Development Conference, in order for this Conference to take the necessary action.

The work was organized as follows: The first meeting agreed to create Rapporteur Groups and each Group was led by a Rapporteur. The Rapporteur on the technical aspects was Mr Jamel Zenkri (Tunisian Expert) assisted by Mr Souheil Marine (Alcatel France). The Rapporteur on economic aspects was Mr Sameer Sharma (Indian Expert) assisted by Mr Kumar Jayant (Indian Expert). The Rapporteur on policy issues was Ms Virginia Sheffield (US Expert) assisted by Ms Julie Kearney (US Expert); the Rapporteur on workshops and training aspects was Mrs Rosa Rizvangoul Cisse (Malian Expert) assisted by Mr Désiré Karyabwite (ITU/BDT). Finally, the Rapporteur on the checklist was Mr Nabil Kisrawi, Chairman of the Group of Experts

I.1.2 Working definition of IP telephony

I.1.2.1 Technical motivations for IP telephony

Although IP telephony does not yet constitute a substantial percentage of the global worldwide telephony traffic volume, it is expanding rapidly due to the following technical motivations:

- The circuit-switched network was designed and optimized to provide a single product – full-duplex 4 kHz switchable voice channels between points (64 kbit/s digital channels).
- Data, in general, is characterized by bursts of information rather than the constant bit rate flows typically associated with speech.
- Data bursts can be most efficiently transported using packets of information that can be interleaved in time within a network with other packets being carried between other sources and destinations.
- For more than 40 years, voice has been digitally encoded into 64 kbit/s streams that can be transported over the 64 kbit/s channels. However, advances in voice coding permit a wide range of options, e.g. from 5-8 kbit/s to higher quality audio at 64 kbit/s. Multiplexing voice at other than 64 kbit/s is difficult in the 64 kbit/s circuit-switched network. However, IP telephony subscribers need to interconnect with the more than 1 billion worldwide subscribers of classical telephony and when doing a transcoding mechanism it is necessary to transform their lower bit-rate to the legacy 64 kbit/s encoding (much like what happened when connecting low-rate encoding of mobile networks to fixed PSTN networks).

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- Significant work has been performed in the Internet Engineering Task Force (IETF) and elsewhere to provide real-time or near real-time capabilities using IP that can permit voice to be transported over IP using the range of voice coding. Carrier-grade products that integrate those protocols are being introduced in the field to meet quality of service to satisfy their customers. IETF is currently working on protocols that ensure that QoS constraints are met in a consistent manner over a set of traversed networks.
- This flexibility to transport a variety of user information streams, i.e. constant and variable bit rate, different speeds, etc., allows packet-switched networks to evolve with the objective of one integrated network for a wide range of applications.
- One integrated network (packet-switching) can reduce the operational and maintenance costs over those associated with multiple overlay networks. However, in the short term there may be additional expenses.
- In addition, the flexibility of packet-switched networks to accommodate new information streams with a wide range of characteristics and based upon the Internet Protocol and the host of open, standardized interfaces and languages available to it allows the introduction of new applications producing new revenue streams. In some cases those capabilities could be *the* real driver for the introduction of IP transport within telecommunication networks rather than the "reproduction" of existing telephony services.
- IP-based networks can use the same underlying lower layer transport facilities, i.e. twisted metallic pairs, cable, wireless, optical fibre, satellite. The evolution to IP-based networks can be accomplished economically by deploying IP-based packet switches/routers that can be connected by existing transport facilities. This was a tremendous vehicle for being able to offer an Internet access to mass markets in developed countries due to the availability and ubiquity of those transport facilities.

I.1.2.2 Introduction to the different types of IP telephony

According to the nature of the IP network used, we may speak of two major categories for voice transmission over IP networks. The first is essentially based on the Internet, which is seen as the interconnection of a host of public or private networks on a global scale. The second is provided by service operators using managed IP networks, within which a number of pre-installed mechanisms (routing algorithms, coding, etc.) serve to ensure a quality of service level that is acceptable for speech.

There are three Voice over Internet Protocol (VoIP) usage scenarios according to terminal equipment and types of network:

Scenario 1: PC to PC

In this scenario, the calling and called parties both have computers¹ that enable them to connect to the Internet, usually via the network of an Internet service provider (ISP)². The two correspondents are able to establish voice communication only by prior arrangement, since both users have to be connected to the Internet at the same time (having fixed in advance the time at which they will

¹ Actually, the term computer or PC indicates a device capable of executing a VoIP application software programme. Today, we see the emergence of advanced user appliances like personal digital assistants (PDA) or advanced mobile handsets that are capable of running VoIP software; therefore the term PC used in the sequel is used for convenience and should be understood in the above general meaning.

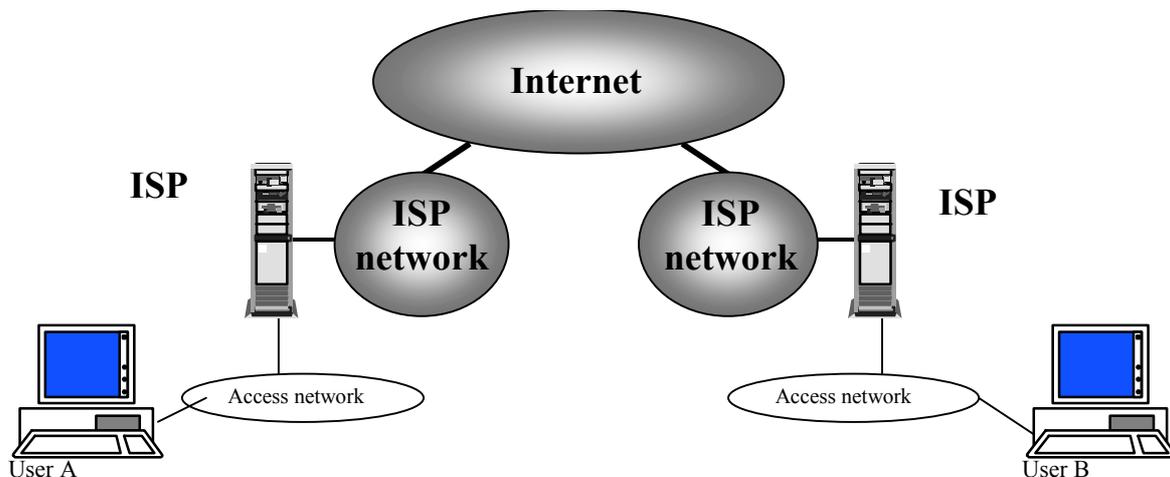
² The role of the ISP is primarily to allow its subscribers to connect to its network and provide them with an IP address allowing them to use Internet applications. The case of accessing to the Internet through an ISP is cited here as the dominant example. Of course, users connected directly to a LAN or WAN (enterprises or academia networks) can have an IP address – albeit a private one behind a network address translation (NAT) scheme – and use the Internet applications without the intervention of an ISP.

communicate via the Internet, unless of course they are permanently online) and use VoIP-compatible software³. Furthermore, the caller must know the IP address of the called party; to overcome this, correspondents must agree to consult an online directory server (updated with each connection) where users register prior to each communication or have other ways of locating or being aware of the availability of their correspondent's connection to the Internet (Instant Messaging technologies).

In this scenario, the ISP is generally accessed via the public telephone network by means of a simple telephone call. This means of access still predominates, even in developed countries. Alternative solutions, known as "broadband" and based on the telephone network (DSL technology), a cable television network or a wireless access network (LDMS technology), are currently at the early stages of deployment, but have not yet come into widespread use, even though certain countries are already well equipped⁴.

The ISP's role in this scenario is limited to the simple provision of access to the network, which in turn enables the user to access the Internet. The voice application used by the customer is transparent for the ISP, which takes no specific measures to guarantee the quality of the voice service. In short, one cannot in such a scenario speak of "telephony" in the conventional sense of the word, i.e. the provision of a service by a third provider, but merely of the use of a voice application via the Internet, such usage having become as commonplace as any other network application. The protocol used between the two communicating parties is often the H.323 (see Annex F.1) protocol defined by ITU-T (e.g. the NetMeeting application); however, IETF's SIP protocol (see Annex F.2) could see more widespread use in the future. This solution is illustrated in Figure 1 below.

Figure 1 – PC-to-PC IP telephony



³ The telephony softwares currently available on the market all have a similar structure, displaying a control panel from which the main telephony functions may be controlled and the configuration and options menus consulted. All such softwares provide access to Internet relay chat (IRC) areas, in which users can exchange text messages in real time, to which end a list of individuals using the same software and currently online is displayed. According to the product, there is also a menu which enables the user to make a call to a specific IP address that is permanent and corresponds to a machine that is already connected to the network. Some products may include encryption of voice communication. A voice-mail option enables voice messages to be recorded by the machine.

⁴ The main European Union, North American and Korean operators already report an availability in the order of 90% ADSL access (see also ITU's series "The ITU New Initiatives Programme" reports on "Promoting Broadband", "Economic and Regulatory Implications of Broadband", and "Birth of Broadband").

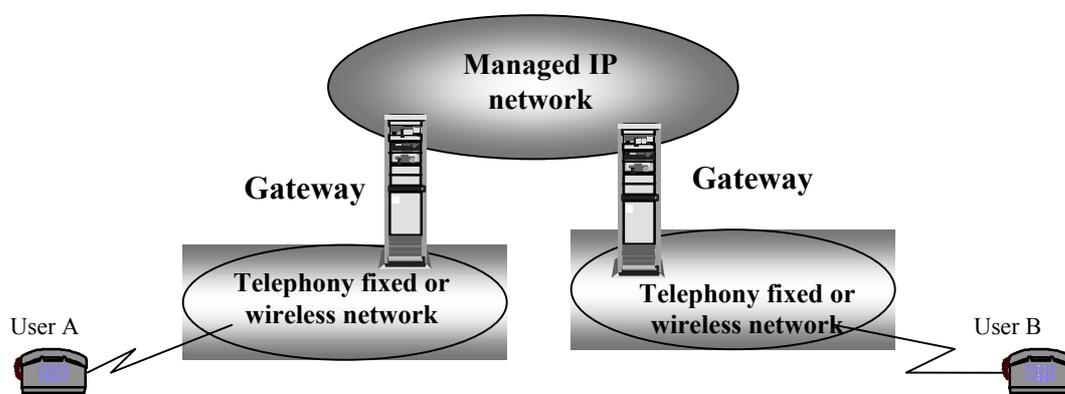
Scenario 2: Phone-to-phone over IP

In this case, the calling and called parties are both subscribers to the public telephony network (fixed or mobile) and use their telephone set for voice communication in the normal way. There are two methods for communicating by means of two ordinary telephone sets via an IP or Internet network.

Use of gateways

This means that one or more telecommunication players have established gateways that enable the transmission of voice over an IP network in a way that is transparent to telephone users. What we have in this case is not the Internet but a "managed" IP network, i.e. a network which has been dimensioned in such a way as to enable voice to be carried with an acceptable quality of service. Figure 2 below illustrates such a scenario.

Figure 2 – Phone-to-phone IP telephony using gateways



In this scenario, the gateways and managed IP network could belong to different players, depending on whether we are looking at:

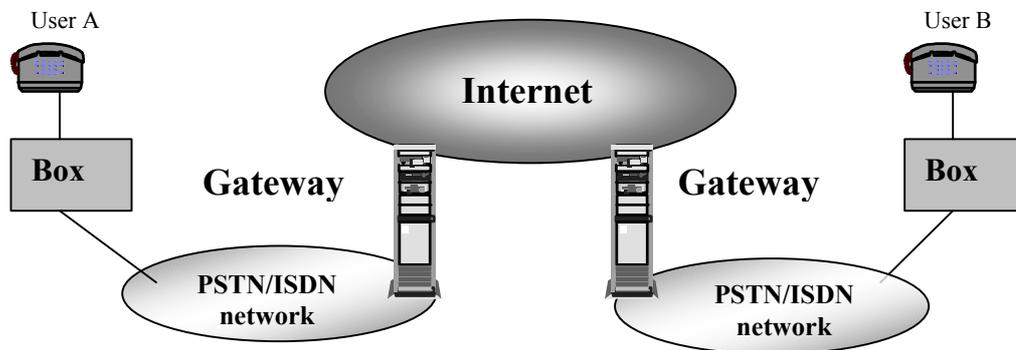
- the purely internal use of VoIP within the network of a single telephone operator, which owns and manages the entire operation, handling both users A and B;
- the provision of a long-distance voice service by a long-distance operator using VoIP technology (users A and B in this case belonging to different networks), in which case the whole operation belongs to and is managed by such a long-distance operator.

Use of adapter boxes

A number of companies market boxes which resemble modems and are installed between the user's telephone set and his connection to the PSTN.

In order for this arrangement to work properly, each of the two users needs to have a subscription with an ISP whose access parameters have been preprogrammed in the box.

The calling party initiates his call in the same way as in a conventional telecommunication network, and the first phase of the call is in fact set-up on that network; however, immediately after this the boxes exchange the information required for the second phase. The conventional call is then broken off and the boxes, on the basis of the data they have exchanged and the pre-established parameters, establish a connection between each of the two correspondents and their respective ISP. Once the call has been established, the boxes locally convert the voice signals into IP packets to be transported over the Internet as illustrated in Figure 3. This scenario is in principle very similar to Scenario 1, except that the two users do not require a PC and the need for an Internet "rendez-vous" is facilitated by the procedure being initiated in the form of a telephone call. However, this type of arrangement has been only marginally successful since it requires – as in the PC-to-PC case – that the two correspondents each be equipped with the same type of box.

Figure 3 – Phone-to-phone IP telephony using adapter boxes


The two methods in this scenario call on two types of network to establish the telephone call, i.e. the Internet or a managed IP network, and the PSTN.

Scenario 3: PC-to-Phone or Phone-to-PC

In this scenario, one of the users has a computer by which he connects to the Internet via an access network and an ISP (in a similar way to scenario 1)⁵, while the other user is a "normal" subscriber to a fixed or mobile telephone network.

PC-to-Phone

When the computerized user wishes to call a correspondent on the latter's telephone set, he must begin by connecting to the Internet in the traditional manner via the network of his ISP. Once connected, he uses the services of an Internet telephony service provider (ITSP) operating a gateway which ensures access to the point that is closest to the telephone exchange of the called subscriber. It is this gateway that will handle the calling party's call and all of the signalling relating to the telephone call at the called party end.

It should be noted that the ITSP provides a one-way PC-to-phone service and does not manage subscribers as such; *in fact, the PC subscriber uses the ITSP's services solely for outgoing calls*. It should also be noted that the ITSP has a managed IP network, thereby ensuring a certain quality of service for voice as far as the gateway closest to the called subscriber, and that the ITSP also manages the interconnection with the latter's telephone operator. Despite their use of VoIP technology, ITSPs consider themselves to be telephone service providers and generally provide their services to individuals in the conventional manner, i.e. with a charge per minute.

Phone-to-PC

In this case, the calling party is the telephony user and the called party is the PC user. Since a telephony user can essentially dial an E.164 number to reach the called party, then somehow the PC user should have an E.164 number:

- either indirectly: in the case of its interconnection to the network behind an IP-technology Private Branch Exchange (PABX) switch (actually, in this case we can more properly speak about an "IP Phone" rather than a PC device that is connected to the LAN managed by the IP PABX);

⁵ The same remark noted for Scenario 1 applies here; the ISP case is only the dominant example. The user can be connected to the Internet behind a LAN or WAN without the need of an ISP mediation.

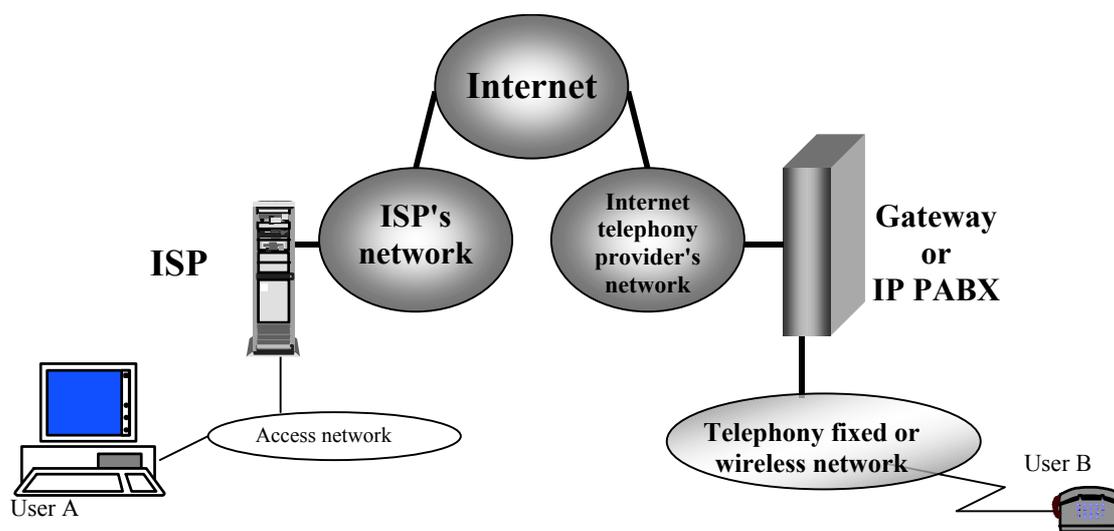
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- or directly: in this case, it the IP side subscriber who has an E.164 address allocated by an IP telephony operator.

Technically speaking, only the first of the above cases works today through the availability of IP PABX devices. The second case will work pending the availability of an intermediary translation mechanism implemented by the IP side that translates the public E.164 number to the IP address of the called party. This will only be available pending the implementation of a technology like ENUM that will be discussed in Annex H.

Figure 4 below illustrates this scenario.

Figure 4 – PC-to-Phone or Phone-to-PC IP telephony



1.1.2.3 Working definition of IP telephony

ITU-T Study-Group 2 (SG2) issued the following explanations of the term "IP telephony":

"IP is an abbreviation for Internet Protocol. It is a communications protocol developed to support a packet-switched network. The protocol has been developed by the Internet Engineering Task Force (IETF). **IP telephony** is the exchange of information primarily in the form of speech that utilizes a mechanism known as Internet Protocol."

The position of SG2 regarding the term "Internet Telephony" should also be noted:

"The combination of the term "Internet" with the term "telephony" is seen as inappropriate. The Internet offers many capabilities to users including the ability to carry bi-directional speech in real time or near real time. We consider this to be an intrinsic capability of the Internet and not a telecommunication service."

Apart from the possible use of the telephone network as a network providing access to the Internet, it is possible to categorize the scenarios presented above into two types:

Type 1: Those requiring the intervention of an operator and enabling, by means of a gateway, the partial (in one direction as in Scenario 3) or full (in both directions as in Scenario 2 with gateways) provision of communication to the global public switched network.

Type 2: Those requiring no intervention by a third provider (as in Scenario 1 or Scenario 2 with boxes) and without the need for a gateway; in this case, the application of VoIP is seen as one of the multiple applications of the Internet world.

Type 2 is close to what SG2 considers as "Internet Telephony" in the sense that it uses "the intrinsic capabilities of the Internet and [does] not [involve] a telecommunication service". Type 1 scenarios, on the other hand, use the Internet Protocol as a bearer for speech but involve an intervention of an operator if only for the provision of an interconnection service towards a telephony network subscriber. They are closer to the above definition of IP telephony though that definition focuses only on the transport technology used for speech transmission (namely, the Internet Protocol) and does not seem to address the other known attributes of Telephony as a **service** provided by an operator.

It goes without saying that the first type of usage is the more advantageous, at least in the short and medium terms. It is alone in providing access to over one billion telecommunication network users throughout the world, thereby contributing to universal access to telecommunication services.

The second type of usage is of interest, in the short term, only to the community of Internet users, and will become valid as a long-term universal communication model once all user equipment (particularly terminals) throughout the world has migrated to "native" IP technology for accessing the Internet, and once the technologies needed to implement the quality of service for applications involving interaction between individuals (whether by voice and/or other media) have been widely introduced in IP networks. Later in this document, we shall be focusing on the discussion of problems relating to implementation of the IP telephony service and to the ways in which the PSTN and networks using IP technology interact. We shall also be looking at the technological factors favouring migration by the telephony service to IP network technology and at the prospects that are opened up by that migration in terms of new services.

I.2 Checklist of Factors for the introduction of IP telephony

Following WTPF-01, the Group of Experts on "IP Telephony" carried out its assigned tasks in the meetings. At the outset, the following checklist of factors was developed ("Checklist of Factors") to be of assistance to national policy-makers and regulators when considering, within their national sovereignty, the introduction of IP telephony. Recognizing that each country has unique circumstances to consider, this checklist provides Member States with a checklist of factors which they may use in the process of accelerating the introduction of IP networks, which may lead to the introduction of IP telephony. It is important to note that the suggestions presented in the checklist are not prerequisites for the introduction of IP telephony.

- 1) Does the growth of telecommunication traffic and the relative proportions of voice and data support the introduction of IP telephony?
- 2) Consider the need for know-how and training to deploy rapidly skilled personnel who are able to meet the technical, operational, management and policy challenges stemming from the new IP telephony environment.
- 3) Universal service: Role of an IP telephony system in the provision of universal/access service.
- 4) Affordable prices: What parameters should be taken into account for:
 - prices of proposed IP telephony to be considered as being at a level to make the applications as widely utilized as possible,
 - costs that affect the affordability of the service such as:
 - Interconnection charges
 - Reasonable transit charges, if applicable
 - Government charges (if any).
- 5) Interconnection matters: Are there technical and/or operational limitations that would prevent interconnection, and what steps must be taken to resolve them?
- 6) Numbering: What steps may be need to be taken to implement the internationally numbering scheme for IP telephony systems (E-164 and/or ENUM)
- 7) Consider the extent to which a system may prevent and/or identify unauthorized uses, means for identifying them, their impact on PSTN and other telecommunication service providers.
- 8) Consider the impact of the operation of IP telephony on existing telecommunication networks and services and on their revenues, while weighing the overall benefit that may be derived from the introduction of IP telephony.
- 9) Consider those competition matters between IP-based systems and existing telecommunication networks and services to ensure a competitive environment.
- 10) Consider the extent to which IP-based systems are able to provide for privacy and security of communications.
- 11) Consider the extent to which IP telephony is responsive to telecommunication requirements in cases of emergency.
- 12) Consider broad-based investment sources, including private-public partnerships and domestic and foreign sources, for the introduction of IP-based networks and services.
- 13) Consider non-discriminatory, transparent and efficient rulemaking processes which are consistent with the sustainability of new technologies.

PART II

IP TELEPHONY – TECHNICAL ASPECTS

Chapter II.1 – Network architecture

A telecommunication network may be described as the set of infrastructures which enable the transfer of information from one point of connection to the network to another. The notion of network arises from the need to share infrastructures in order to optimize their costs, it being possible for a single transmission line to be utilized by different users for different periods. To these internal lines we must add distribution or access networks in order to reach all potential users of the network.

II.1.1 Existing legacy telephony network architectures

Telephony networks have in the course of time undergone major evolutionary changes, driven essentially by technological progress in various fields (switching, transmission, access and maintenance). The latest such change is digitization of their transport technology, which has had a considerable integrating influence.

Still, the end purpose of a telephone network was always associated with the provision of a universal communication service with a certain quality. This has several implications for the technologies used and the mode of interconnection between subnetworks.

Every telephone operator throughout the world operates a subnetwork of the global telephone network. In order to provide a universal communication service to all of their respective subscribers, they have an obligation to interconnect their networks and to agree on a single and coherent system for designating their subscribers.

The quality-of-service requirement implies that adequate resources (circuit capacities, transmission speeds, management arrangements) must be mobilized throughout the duration of a call in each of the subnetworks implicated in the call between the two communicating parties. This has a bearing not only on the technology used to carry voice, but also, and more fundamentally, on the very design of the logic incorporated in the network's active components (switches) and of the mutual language (signalling) they use to ensure the proper routing of a call between two or more subscribers.

The current technology used for the purpose of transporting voice within telephony networks is known as "circuit switching". It is based on the principle that a resource (circuit) must be reserved for a call from the time of its setting-up to its conclusion. The size of this resource – expressed as a bit rate since the digitization of telephone networks – is 64 kbit/s. This limit was chosen at the time because it made for the efficient digitization of human voice samples, the spectrum of which lies between 300 and 3 400 Hz⁶. More recent voice coding techniques allow for a considerable reduction in the 64 kbit/s defined for a circuit; however, since this is the rate that is used in most of the active and transmission components throughout the global telephony network, it would be difficult to change it without incurring excessive costs and without jeopardizing one of the main qualities of that network: the universal service. As a case in point, one can mention that although modern wireless GSM network use an encoding that consumes as low as 8 kbit/s on the radio part, this is transformed to a 64 kbit/s encoding when voice reaches the mobile switches.

Likewise, the architecture of a telephone network is based on the assumption that it will essentially be used for person-to-person voice communication, this being an advantage in terms of optimization of the network for this application, but also a source of weakness if the same network is used for other applications (data transfer) where the requirements differ from those pertaining to voice.

Where did the idea come from of using the telephone network for other applications and the concomitant notion that voice could be carried other than over 64 kbit/s circuits? This will be discussed after the following brief historical overview of data networks.

⁶ At the rate of 8 000 samples per second, each with 8-bit coding.

II.1.2 Data network architectures

Data networks were initially designed – and are still widely used – to interconnect computers, their servers and the management platforms between them. The applications which use data networks are generally computer applications for the exchange of data between machines. The deployment of data networks experienced a boom following the widespread introduction of computers in companies and universities in the wake of the progress made in the field of information technology (mini-computers, then micro-computers, including portable).

Data networks display a certain "neutrality" vis-à-vis the applications they support, being generally characterized by the transfer of data in "packets", each containing all or part of the data to be transferred between two computers and the address of the destination computer. The network is essentially responsible for transferring the packets to the destination computer with a quality of service defined by the user, understood here as the application on the transmitting computer, without any *a priori* knowledge of the nature of that application. The advantage here is that of a multiservice network whose architecture is not subordinated to a specific application, it being even possible to use the network at a later stage for applications which had not even come into being at the time of its original deployment. This flexibility had led, for instance, to the use of data networks (especially IP ones) for new types of "human-related" communication applications, like voice and video transmission, and through – still negligible but likely to take off in the coming years – appliance devices capable of executing a given subset of communication applications without the need of having a "general-purpose" computer device.

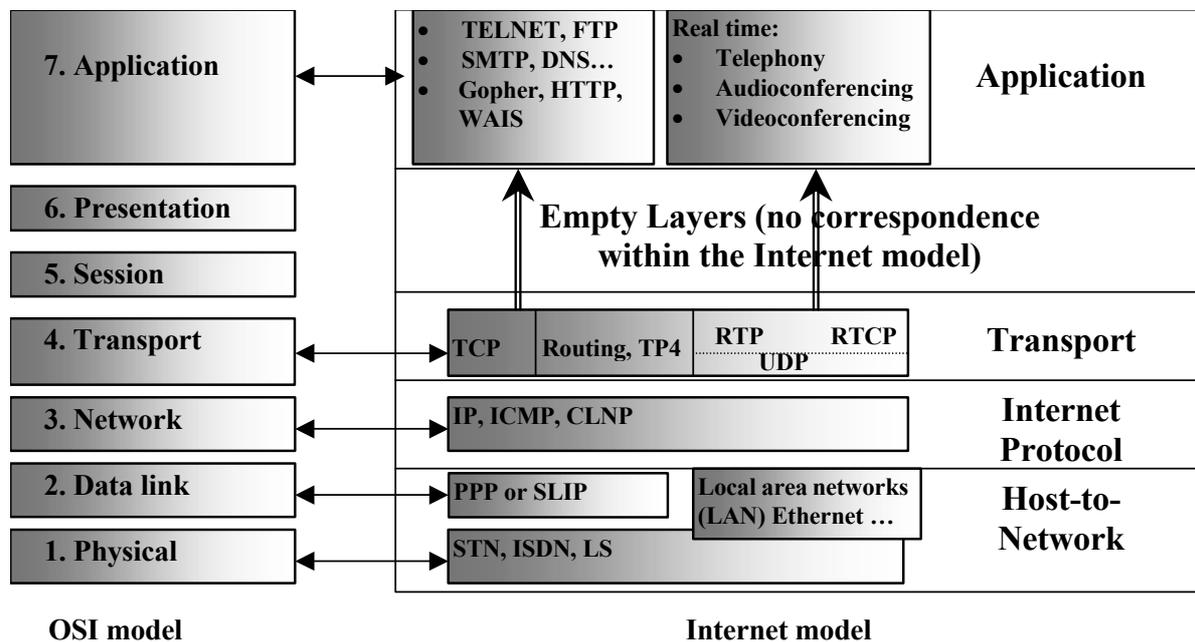
This pursuit of openness and generality (i.e. neutrality and transparency vis-à-vis applications) has led to the definition by ISO of the Open System Interconnection (OSI) model. The cornerstone of this model is a seven-layer architecture in which each layer provides services to the layer immediately below, the "application" layer being the one situated at the highest level of the model. A whole host of specific protocols has been defined in order to "populate" the different levels of the OSI model. It is clear that, in any given network deployment, only a subset or "profile" of these protocols is used. The network interconnections are therefore not pre-established, and must be implemented on a case-by-case basis by means of gateways or bridges of varying complexity.

An interconnection protocol known as the Internetworking or Internet Protocol was developed in the latter part of the 1960s in the United States for the initial purpose of enabling communication between university computers used in United States defence work. This protocol belongs to layer 3 (network) of the OSI model. Its correspondence with the protocols and applications developed in connection with it are illustrated in Figure 5 hereunder.

The IP protocol is independent of the underlying layers, which means that it is able to adapt to a network whose supports are both varied and numerous. It is, however, a simple protocol which does not implement error control.

Section A.1 in Annex A describes the IP protocol and Section A.2 in Annex A describes the UDP protocol used to transport voice over IP. It should be noted that the applications associated with the IP protocol have been gradually developed over time. Up until the early 1990s, the applications most used were Telnet (initiation of a session on a remote computer), FTP (for obtaining files from a remote computer) and e-mail (for the exchange of electronic mail). These applications were essentially used by individuals within the scientific and technological community accessing computers at their place of work. It was the arrival on the scene of the application known as the World Wide Web (HTTP protocol) and the availability of relatively powerful personal computers which, in the mid-1990s, sparked the development of the Internet for use by the public at large.

Figure 5 – Correspondence between the Internet and the OSI model



II.1.3 Data invasion of the telecommunication network

Once the need to interconnect personal computers to data networks became apparent, the most natural choice of access network was the global telecommunication network!

The reasons for this choice were twofold: first, the ubiquity of the telephone network, particularly in developed countries, where one or even more telephone lines are installed in each household; and second, the availability of modems for converting digital information from the personal computer into analogue signals for transmission over the telephone network.

The situation we have today is one in which the interconnection of personal computers to the Internet is for the most part via the global telephone network, including in the developed countries. The upshot of this is that, in some developed countries, the volume of data traffic over telephone networks exceeds that of the voice traffic for which those networks were originally designed⁷. This growth of the data traffic is the result of good Internet penetration (especially in developed countries), innovation in IP-based technology and the dynamic growth of user applications and Internet infrastructure. This growth of data traffic had also led to the demand and development of so-called "broadband" access to the Internet through new technologies like DSL, cable TV, or Wireless Local Loop to cite the most important. However, those technologies have not yet replaced the still dominant dial-up access mode to the Internet even in developed countries. It should also be noted that the DSL technology at least reckons on the presence of the physical copper wire that was laid out by a telephony operator.

⁷ Even though access servers to the Internet have been largely deployed in developed countries and traffic towards ISPs switched off from the telephony network as early as possible (congestion avoided at transit level), two problems remain: 1) congestion at access level is still not removed, and 2) a telephony operator dealing with heavy Internet traffic should deploy data networks to handle the Internet traffic.

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The first problem to which this invasion of the telephone network by data gives rise concerns the appropriateness for data applications of a mode of transport involving circuits of a specific size which must be reserved throughout the period of the call. This represents a problem because data applications often include a very high proportion of silent periods, the term used to describe this being "sporadic application". It is easy to see that the switched-circuit mode of transport is not suited to this type of application and that the packet mode of transport referred to earlier would be more appropriate.

The second problem is as follows: bearing in mind that the telecommunication network carries more data than voice, would it not be a good idea to have voice migrate to a packet transport technology of the IP type? This gives rise to devising what migration paths can lead to such a converged data/voice networks and how they would offer their services to end-users. These problems will be discussed later, but before this it would be useful to conclude this chapter by digressing onto the subject of how tomorrow's telecommunication networks might evolve.

II.1.4 How will tomorrow's telecommunication networks look?

II.1.4.1 Telecordia NGN architecture

Earlier in this report we discussed the parallel evolutions which led to the establishment of two types of network, each optimized for transporting the predominant type of information for which it was designed (speech for the telephony network and all types of data for data networks). The use of the telephony network for accessing the Internet, and the possibility of carrying voice over an IP-type packet network, lead us to consider the possibility of convergence between these two types of network.

Indeed, should the ultimate objective for the future not be to create a unique type of interconnected, service-independent networks that are capable of transporting data, voice or any other type of application? If the answer to this is yes, then what are the main properties that such a next-generation networks must "inherit" from data networks on the one hand, and from telephony networks on the other?

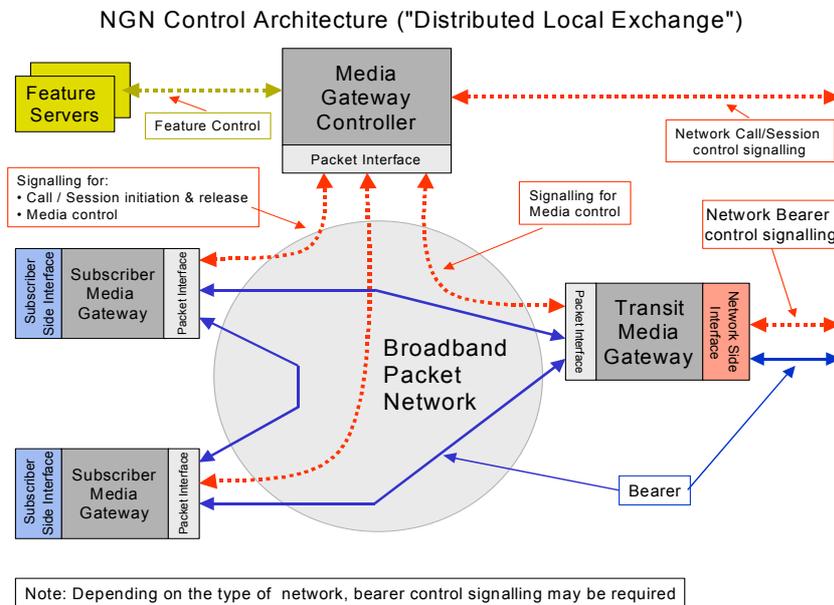
The concept of next-generation network was initially introduced by Bellcore. A next-generation network is essentially characterized by the separation of transport and control function and the fact that the former is based on packet technology. The aim of this architecture is to build a converged network where voice and data share the same transport infrastructure. One other aim of this architecture is that it opens the way for a new breed of services. A next-generation network can have the following technical characteristics:

- All application data are carried in packets/cells:
- Broadband technology in the access
- QoS capable multiservice networks in the edge network
- Optical networking in the core network
- Open distributed control architecture replaces the classical "monolithic" switch
- Distributed intelligent layer that separates control logic from transport
- Open platforms and open APIs for creation, provisioning and delivery of intelligent/enhanced services
- "Web technology" for management of networks and services, including customer "self-management" (*Web-based management*)

The **control architecture** can be characterized by the decomposition of the classical monolithic local exchange into units that handle the media stream, a unit that contains the logic for call/session control and a broadband packet network that provides connectivity. These (functional) units may be contained in physically and geographically separated boxes.

A simple, but adequate, model for the "Distributed Local Exchange" is shown in the figure below.

Figure 6 – Simplified view of Distributed Local Exchange



The above view of the next-generation network cannot be complete without, first, discussing migration scenarios from the current TDM telephony networks to this target (this will be discussed within the next chapter) and, second, discussing various technical problems related to the provisioning of communication services (that replicate or extend the voice communication service offered by telephony networks), namely:

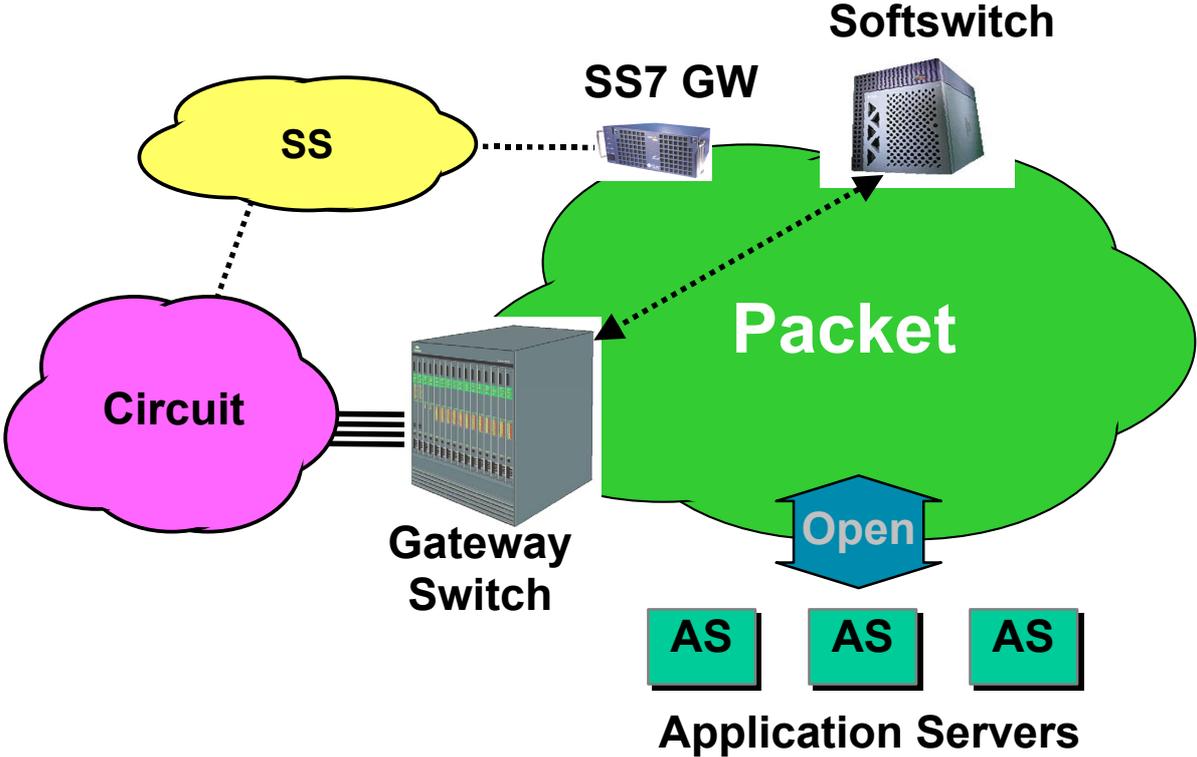
- 1) quality of service;
- 2) call security;
- 3) network interconnection and interworking;
- 4) the provision of service and applications to network users;
- 5) charging for the use of services and applications;
- 6) the designation of users and addressing plans.

Each of these considerations will be discussed within this report, where we shall seek, to the extent possible, to maintain a constant approach by presenting the status of the technologies used in each of the two types of telephony and data networks and the possible or foreseeable synthesis, or syntheses, between them in the case of a convergent network carrying both voice and data services.

II.1.4.2 Software Switch Consortia architecture

An alternative architecture is provided by the Softswitch Consortia and is provided below:

Figure 7 – Software Switch Consortia Architecture



Chapter II.2 – Strategies for migrating telephony networks towards next-generation networks (NGN); when, how, and for what purposes?

II.2.1 A general framework for migration to NGN

In markets with a high growth in traditional voice services (which is the case for most developing countries), substantial extensions will be required to the existing telephony network in order to cover the huge need for new lines. Established service providers will have to decide on how to extend their networks: using more traditional circuit-switched solutions or implementing a distributed network architecture, with a common, packet-based transport layer for voice and data.

For this to occur, many aspects like network consolidation, expansion and migration⁸ need to be taken into account in a way that is specific for each operator. However, one can devise the following generic step-wise approach as follows:

- *Step 1*: use of today's TDM-based network for voice telephony and Internet access;
- *Step 2*: consolidation of switching and access equipment;
- *Step 3*: introduction of Voice-over-Packet technology for trunking;
- *Step 4*: introduction of Voice-over-Packet technology in access and CPE;
- *Step 5*: multimedia services and new applications;
- *Step 6*: end-of-life replacement of legacy infrastructure and migration to all-IP signalling.

The above steps are generic in the sense that they are not mandatory for each specific operator case. Still, they offer interest by highlighting major evolutionary steps of networks that might occur in the following years. Each step will be briefly discussed below.

II.2.2 PSTN for voice and Internet

The starting point for the migration to NGN is today's Public Switched Telephone Network or PSTN as illustrated by Figure 8 hereunder.

II.2.2.1 TDM and SS7 [A]

In this network, all voice traffic is transported over TDM, and controlled by a hierarchy of local (LEX or Class 5) and transit (TEX/Class 4) circuit switches. All the voice-related signalling network (ISUP and INAP) is handled by the SS7 signalling network.

II.2.2.2 Intelligent network services [B]

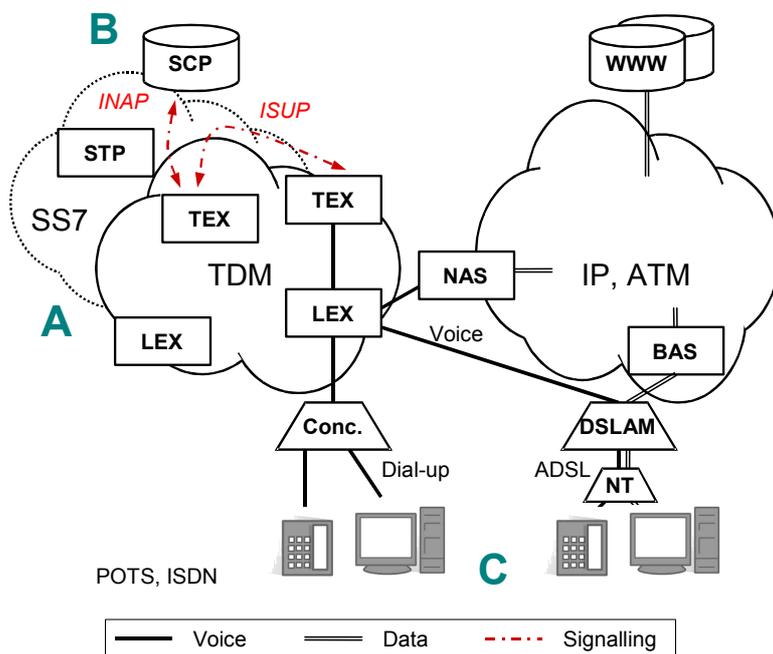
Value-Added Services are provided inside the switches, or through the Intelligent Network (IN). Widely spread IN services include Calling Card services, Number Translation and routing services (such as Freephone, Premium Rate and Universal Access Number), and Enterprise Network services (such as Virtual Private Networks and Wide Area Centrex).

⁸ The above terms need proper definitions to remove any ambiguities in the sequel. Network consolidation refers to the optimization of the installed PSTN base to reduce capital (CAPEX) and/or operational (OPEX) expenditures. Consolidation can be combined with selection of future-safe products to prepare migration to NGN. Expansion refers to the introduction of an overlay NGN (based on broadband access) for addressing new customers and introducing new services (like multimedia communication). Finally, migration refers to the replacement of PSTN components (at their end-of-life) with equivalent NGN ones.

II.2.2.3 Internet access [C]

With the growing number of Internet users, carriers are providing connectivity to Internet Service Providers (ISP) either through narrow-band (PSTN or ISDN) dial-up services, or through introduction of broadband ADSL (with voice split off as a separate service).

Figure 8 – Step 1: PSTN for voice and Internet

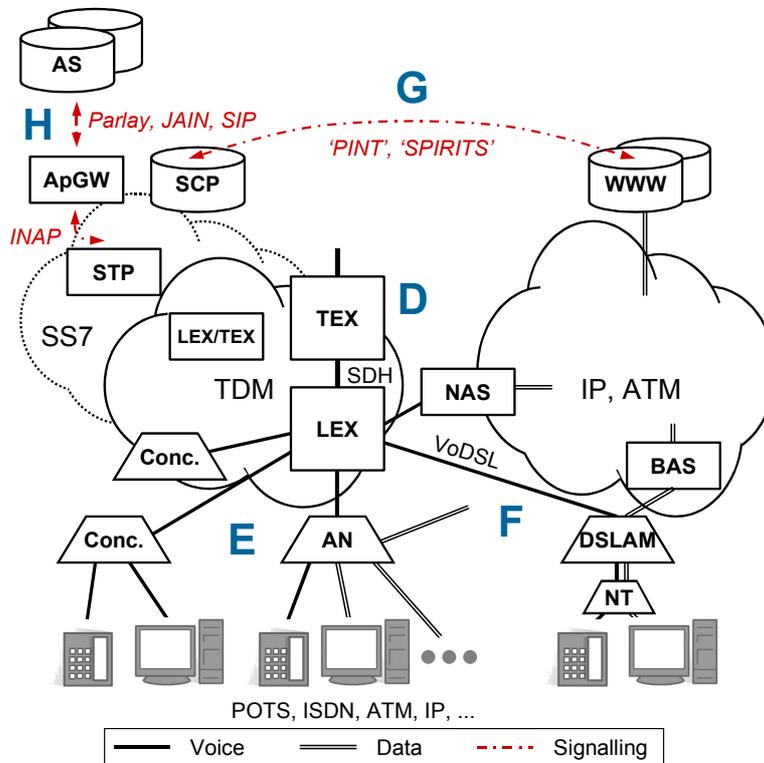


II.2.3 PSTN consolidation

Established carriers in growing markets will face major capital expenses (CAPEX) to extend the capacity of their network in line with the growth in subscribers. These investments will be needed at all levels of the network, local level as well as in the long-distance network.

A safe approach (from a technical as well as from an economical viewpoint) is to start with consolidation of the existing PSTN infrastructure while selecting "NGN-ready" products for expansion, and introduction of new services for additional revenue generation as illustrated in Figure 9 hereunder.

Figure 9 – Step 2: PSTN consolidation



II.2.3.1 Switch consolidation [D]

Deployment of a small number of large exchanges (local and transit) with increased switching capacity, and high-speed interfaces (e.g. SDH, ATM) reduce the operator's operational expenses (OPEX) and enable faster deployment of new services. "Redundant" switches may be converted to additional remote access concentrators.

Introduction of new technology with, for example, smaller footprint, or packet fabrics inside the exchanges, allows the carrier to reduce expenses and reuse the switching equipment for new data services.

II.2.3.2 Access consolidation [E] and VoDSL [F]

Adding new Access Nodes and upgrading the existing ones lets the carrier capitalize on his PSTN, while extending the coverage area and the bandwidth offered to individual subscribers (fibre closer to the end user). New access technology provides seamless multiservice access to voice (POTS, ISDN) and data (ADSL, ATM, IP, ...) services and paves the way to next-generation networks.

Optimization of the ADSL access infrastructure is realized through introduction of Voice-over-DSL (VoDSL) loop-emulation services (inverse gateway, with a V5.2/GR303 connection to the LEX).

II.2.3.3 IN-Internet convergence services [G]

Providing an external server to the PSTN and the Internet, the IN Service Control Point (SCP) may be used as a means to integrate voice and data into common applications.

Example IN-Internet convergence applications are Click-to-Dial, Internet Call Waiting, Web Augmented Calling, Unified Messaging, etc. In order to communicate with the Internet servers, the SCP has to adopt some IETF protocol suites (e.g. PINT and SPIRITS).

II.2.3.4 Open service access [H]

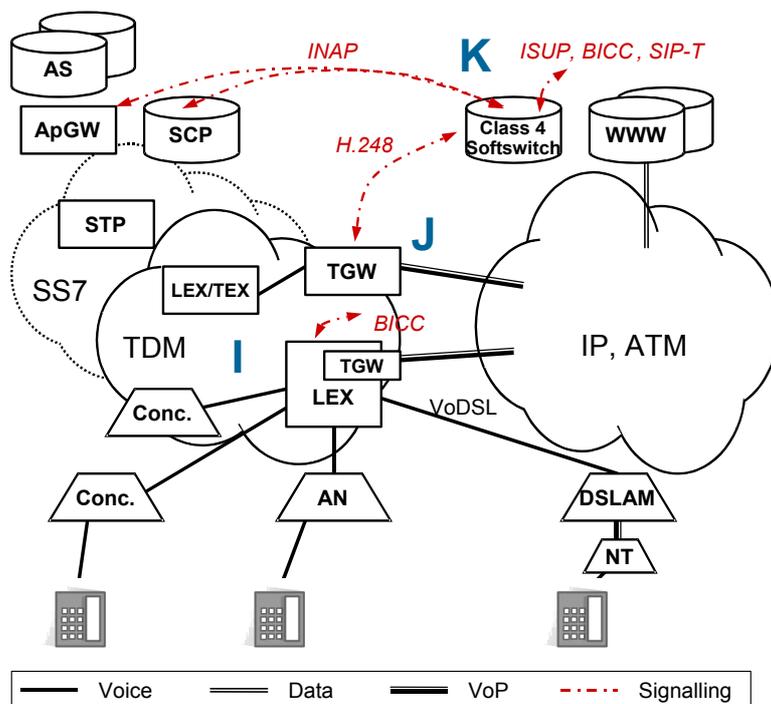
To prepare for the NGN and to get extra revenues from new services, the network operator may deploy Application Gateways (ApGW) with open interfaces (e.g. OSA/Parlay, JAIN, SIP) towards (third party) Application Servers (AS).

II.2.4 Voice-over-packet for trunking

As one of the basic goals of NGN introduction is to move to a unique, packet-based infrastructure (presumed with lower OPEX and CAPEX), voice transport will smoothly migrate to IP or ATM technology.

Initially, carriers will focus on trunking scenarios to offload long-distance voice from their TDM network. This is illustrated by Figure 10 below.

Figure 10 – Step 3: Voice-over-Packet technology for trunking



II.2.4.1 Trunking through integrated gateways [I]

The first step towards VoP migration is extending the existing (local) exchanges with integrated Trunking Gateways (TGW) for converting TDM voice into packets (ATM or IP). This approach guarantees full protection of TDM investments, while providing the operator with a full fledged trunking-over-packet solution, as well as continued access to switch based and IN-based Value-Added Services.

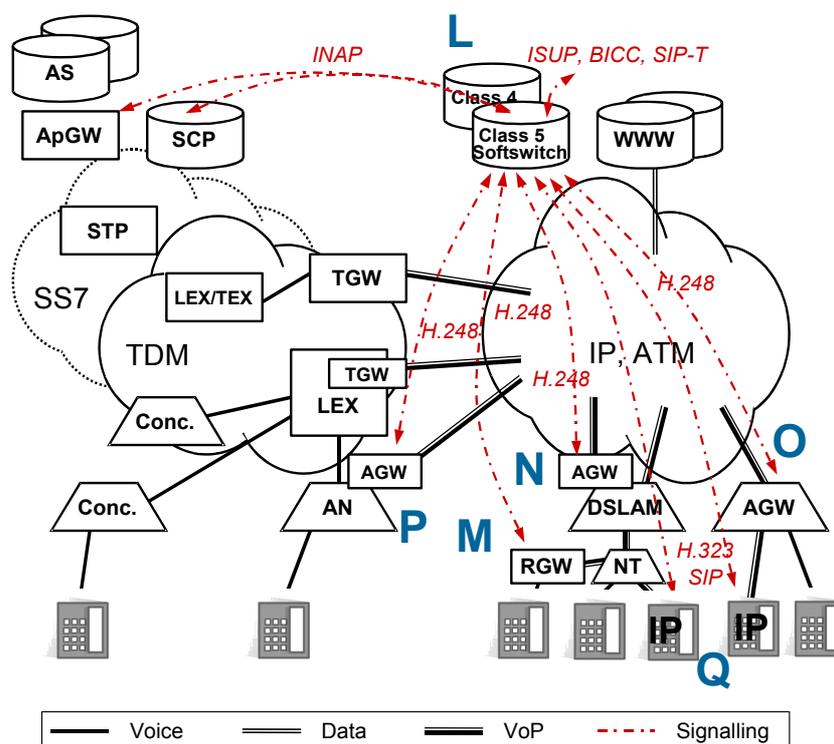
II.2.4.2 TGW [J] with Class 4 Softswitch [K]

In order to address existing switches without integration of a gateway, external trunking gateways (TGW) controlled by a Class 4 Softswitch (through the H.248 or Megaco⁹ protocol – see Section G.1 in Annex G) may be added. From a functional point of view, the Softswitch performs like a Class 4 (Toll/Transit) Exchange, with similar features (e.g. screening and routing), signalling interfaces (ISUP, INAP) and access to Value-Added Services (IN).

II.2.5 Voice-over-packet for access

In fast-growing markets or in markets with aggressive deployment of broadband access (ADSL, LMDS, cable), operators may introduce voice-over-packet technology to capture growth in the access network, or as a means to offload the Local Exchanges from DSL.

Figure 11 – Step 4: Voice-over-Packets for access and CPE



⁹ For early deployment, the MGCP protocol – which is a precursor to H.248/Megaco protocol – might be used due to the availability of gateways supporting it.

II.2.5.1 Class 5 Softswitch [L]

The Class 5 Softswitch with local features (e.g. CLASS, custom calling) will be a shared control element, but several alternatives for voice gateways (depending on end-user topology, density, service requirements, etc.) may be deployed [Figure 11]. Just as in the Class 4 case, the Softswitch will address the gateways using the H.248/Megaco protocol (or MGCP).

II.2.5.2 Residential gateway [M]

ADSL subscribers may install a Residential Gateway (RGW) or Integrated Access Device (IAD) with VoP coding capability. Contrary to the ADSL with split-off voice [B] or VoDSL loop emulation [E] solutions, the RGW provides the broadband user with end-to-end voice-over-packet.

II.2.5.3 Access gateway in the DSLAM [N]

As an alternative to upgrading the CPE of its subscribers, an ADSL operator may choose to extend the DSLAMs with VoP gateway functionality.

II.2.5.4 Distributed access gateways [O, P]

Another solution for connecting the voice subscribers directly to the data network is to introduce new Access Gateways [AGW] or to upgrade the existing access nodes with AGW functionality.

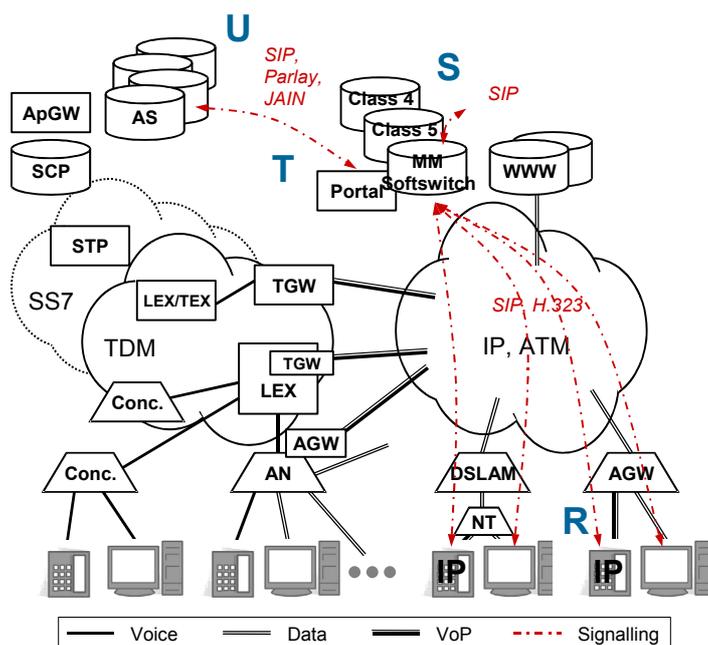
II.2.5.5 IP phones [Q]

In order to address new-generation voice terminals (IP Phones), the Class 5 Softswitch can also terminate emerging user-to-network signalling protocols such as H.323 and SIP.

II.2.6 Introduction of multimedia

It is beyond question that in the near (and even mid-term) future voice will be the predominant service, even in next-generation networks. The introduction of broadband access in the network, however, is enabling the deployment of a new range of data and multimedia services. These new services will allow carriers to differentiate and compete with new entrants.

Figure 12 – Step 5: Multimedia



II.2.6.1 IP clients [R] with MM softswitch [S]

A prerequisite for the deployment of multimedia services is the general availability of appropriate terminals. Today's personal computers are a good starting point, but it is expected that the convergence of computer, consumer and communications technology will result in a number of new multimedia devices.

These new terminals will communicate with the Softswitch through emerging multimedia signalling protocols such as H.323 and SIP.

In order to fully support the new network and terminal capabilities, the Softswitch is extended with mixed-media session and QoS control.

II.2.6.2 Retailer portal and open interfaces [T]

With the introduction of new business models and new players (e.g. Virtual Network Operators, third-party application providers, content providers), there is a need for application access (for authentication, authorization, accounting, roaming, subscriber profiles, etc.) and service brokering platforms (terminal capabilities negotiation, bandwidth brokering, content aggregation, etc.).

Such portals do not only provide the Network Operator with new business opportunities as a Service Retailer, but also clearly separate the network control from the services functionality.

In a full-fledged NGN architecture, applications and network will interface through standardized protocols (e.g. SIP) and APIs (e.g. JAIN, OSA/Parlay).

II.2.6.3 New applications [U]

From an applications (and thus a revenue) viewpoint, "plain vanilla" Voice-over-Packet is not considered as a differentiator. It is even assumed that voice services offered on VoP networks will have fewer features than the ones on circuit networks (especially in an H.323 environment).

Therefore, evolution of the applications portfolio towards data and multimedia is considered an absolute prerequisite for telecom service providers to differentiate, grow and generate new revenues. Typical examples of multimedia applications include:

- Mixed-media calls/conferences
- Real-time data streaming
- Instant Messaging, Presence and Location services

Massive deployment of new applications will be enabled by the availability of application servers and terminals, with easy-to-use service creation tools.

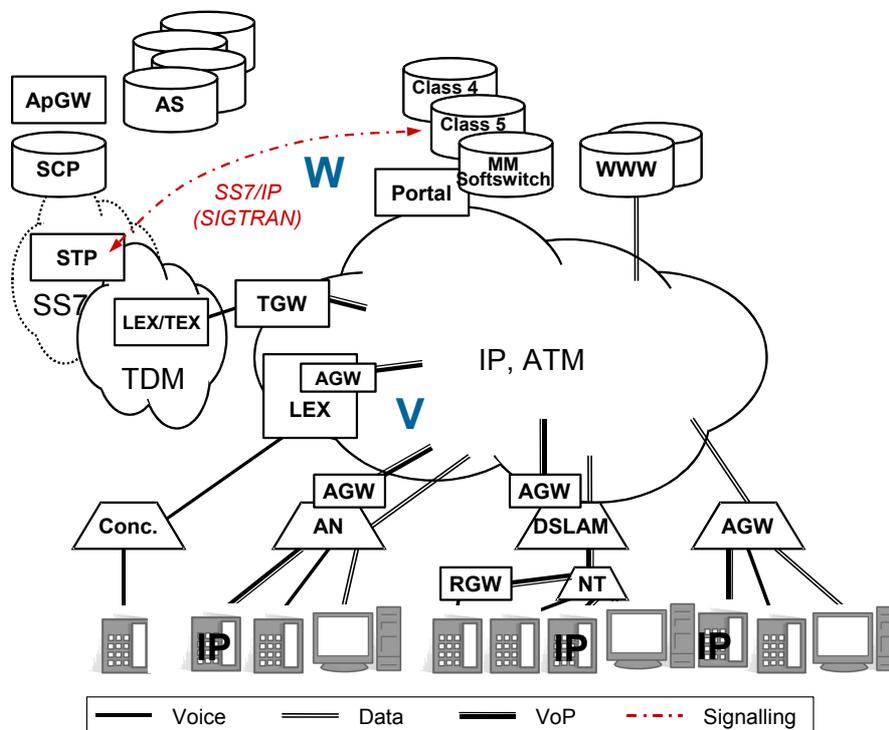
II.2.7 Migration to the full NGN

As a final migration step toward the full NGN, the remaining legacy PSTN equipment is transformed to or replaced by NGN "compliant" network components. The aim of this ultimate (though optional) transformation is to capitalize on existing CAPEX (e.g. access concentrators connected to local exchanges) while further reducing the OPEX (packet-only network for transport and signalling).

II.2.7.1 Replacement of legacy equipment [V]

At the end of their life, remaining TDM exchanges and access nodes are gracefully transformed to or replaced by Trunking Gateways, Access Gateways and Softswitches as outlined in the previous sections.

Figure 13 – Step 6: The full NGN



II.2.7.2 Migration to all-IP signalling [W]

While keeping the upper layers (SCCP, ISUP, TCAP, INAP) intact, the lower layers of the SS7 signalling network are replaced by a packet-based equivalent, as defined by the IETF SIGTRAN working groups.

II.2.8 Alternative migration strategy

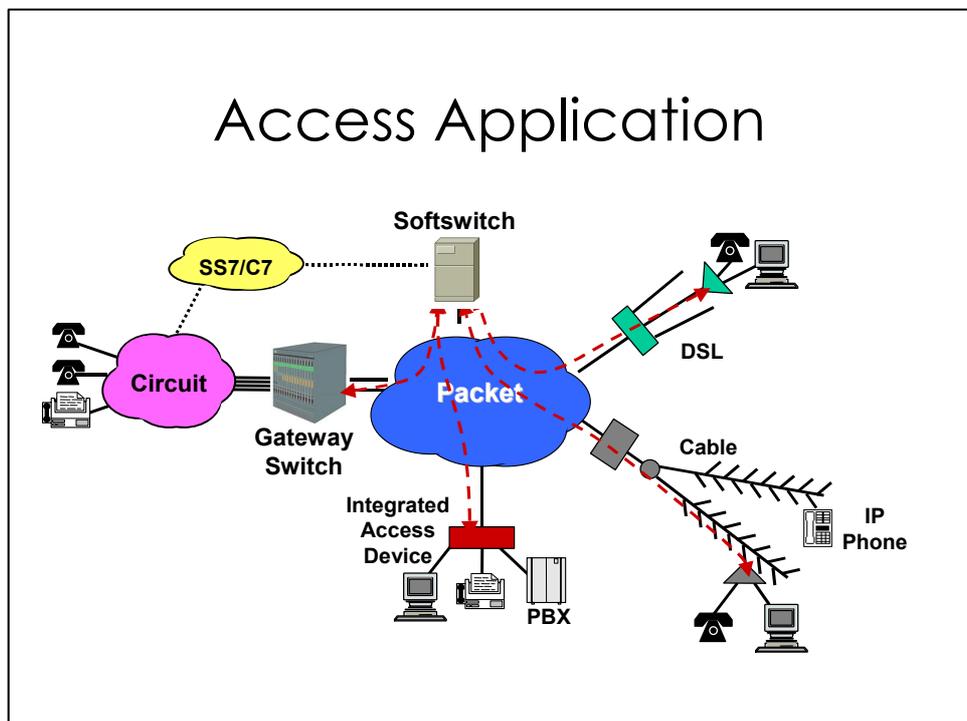
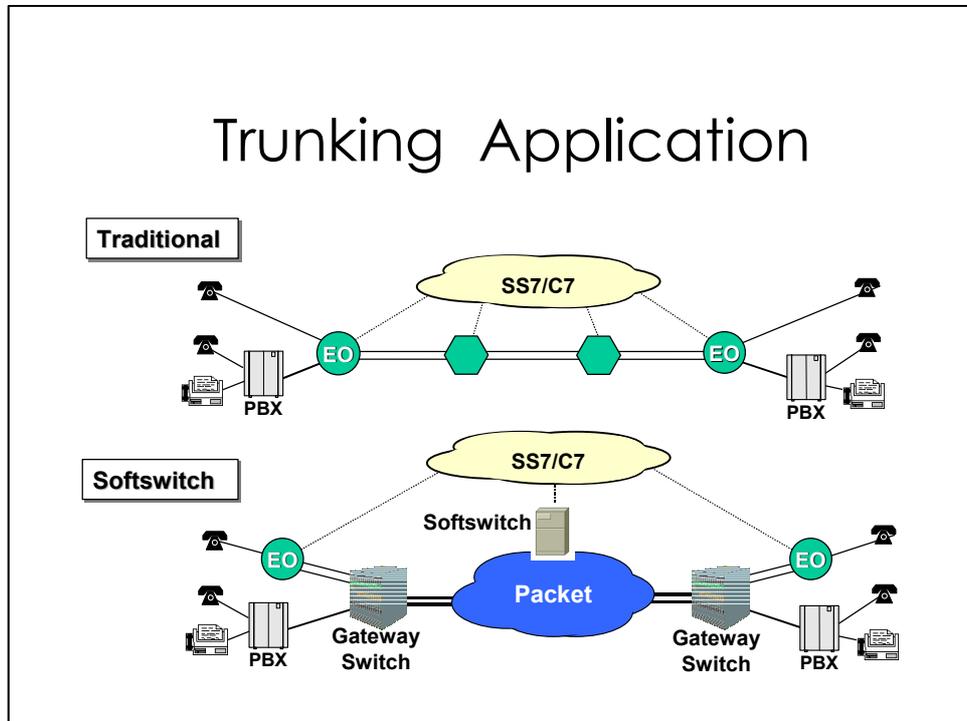
A more simplified migration strategy based on the Softswitch Consortia architecture consists of a two-step approach where:

Step 1 – introduces IP telephony equipment for trunking and access.

Step 2 – introduces new capabilities and applications based on the Internet Protocol and the host of open, standardized interfaces and software languages available to it.

This two-step strategy is illustrated hereafter for trunking and access.

Figure 14 – Alternative migration



Chapter II.3 – Applications

II.3.1 Benefits to end users

In addition to providing the potential for lower cost telephony to end users, IP telephony technology makes it easier to create new applications capabilities because it:

- carries and processes voice, data and multimedia traffic and signalling in the same form;
- utilizes the Internet Protocol and the host of open, standardized interfaces and software languages available to it.
- Examples of such capabilities and applications are:
- IP Centrex – extends traditional Centrex capabilities to not only accommodate voice, but also data and multimedia;
- Unified Messaging – delivers voice, fax and e-mail messages into a single mailbox that users can access anywhere from a web browser, e-mail, or a telephone;
- Pre/Post-paid calling – offers a range of pre- and post-paid calling card capabilities created on an open platform;
- Internet call waiting – enables a single phone line for both voice calls and Internet access;
- Conference call capabilities;
- Call/Contact centres – enables a range of call centre capabilities, e.g. web enabled.

II.3.2 VoIP virtual trunking

In this application, the IP network replaces the TDM trunk network. Calls originating at the PSTN are passed to the IP network at a gateway, which also converts the media stream, carried over the IP network over a pre-provisioned virtual trunk (e.g. layer 2 tunnel) to a gateway at the terminating PSTN, where the media stream is converted back and delivered to the called party. Signalling between the PSTNs uses BICC carried over SCTP.

At present, VoIP, when provided as a public service, is supposed to use E.164-based addressing (in the private trunking application it may follow a private numbering plan), whereas Internet telephony uses web-based addressing. With the progress of work in ENUM (see Annex H), this may change in the future.

II.3.3 Multimedia applications

Delivery of multimedia information and multimedia communication in a unified way is considered as one driver for creating the IPTN+, as it allows the creation of new applications/ services to become a new source of revenue.

Below a brief functional description of the most promising application protocols/frameworks for different types of such services follows:

II.3.3.1 Conversational

The ITU-T Recommendation H.323 is an umbrella standard for specifying a for an IP-based multimedia conferencing system. It refers to a couple of other standards which specify signalling protocols, media coding and call control services. H.323 uses an evolutionary approach to VoIP, which offers a high degree of interoperability to legacy SCN-based telephony services. Drawbacks are its rather high implementation complexity and architectural problems concerning convergence of telephony and Internet services, and lack of scalability and flexibility.

II.3.3.2 Broadcast and streaming

MPEG-2

The ISO/IEC standard MPEG-2 (ISO/IEC 13818, parts 1 to 9) addresses the digital compression of television signals (audio and video), synchronization of media components and their delivery either by a storage media (e.g. DVD) or by a broadcast service. MPEG-2 is the backbone of the Digital TV System, which will replace the analog system in near future. Moreover, it is considered as the standard for delivery of high quality video, e.g. for video-on-demand services in the NGN.

MPEG-4

MPEG-4, a set of standards developed by the ISO/IEC working group MPEG, provides ways for streaming multimedia information to:

- represent units of aural (music, speech), visual (video, photos, graphics) or audiovisual content, called "media objects". These media objects can be of natural or synthetic origin; this means they could be recorded with a camera or microphone, or generated with a computer;
- describe the composition of these objects to create compound media objects that form audiovisual scenes;
- multiplex and synchronize the data associated with media objects, so that they can be transported over network channels providing a QoS appropriate for the nature of the specific media objects; and
- interact with the audiovisual scene generated at the receiver's end.

Although, due to the object-base coding approach, MPEG-4 is primarily predestined for applications which need a high degree of interactivity and flexibility, e.g. for e-learning and advertising, the standard includes as well application profiles, which effectively support conversational and broadcast services, in a similar way to MPEG-2 and H.263.

II.3.3.3 Store and retrieval

MPEG-7

MPEG-7, an ISO/IEC standard developed by MPEG, formally named "Multimedia Content Description Interface", aims to create a standard for describing the multimedia content data that will support some degree of interpretation of the information's meaning, which can be passed onto, or accessed by, a device or a computer code. MPEG-7 is not aimed at any one application in particular; rather, the elements that MPEG-7 standardizes shall support as broad a range of applications as possible.

MPEG-21

The multimedia content delivery chain encompasses content creation, production, delivery and consumption. To support this, the content has to be identified, described, managed and protected. The transport and delivery of content will occur over a heterogeneous set of terminals and networks within which events will occur and require reporting. Such reporting will include reliable delivery, the management of personal data and preferences taking user privacy into account and the management of (financial) transactions.

A multimedia framework is required to support this new type of multimedia usage. Such a framework requires that a shared vision, or roadmap, is understood by its architects, to ensure that the systems that deliver multimedia content are *interoperable* and that transactions are simplified and, if possible, *automated*. This should apply to the infrastructure requirements for content delivery, content security, rights management, secure payment, and the technologies enabling them – and this list is not exhaustive.

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The MPEG-21 multimedia framework, a project of ISO IEC JTC1 SC29 WG11, identifies and defines the key elements needed to support the multimedia delivery chain as described above, the relationships between and the operations supported by them. Within the parts of MPEG-21, MPEG will elaborate the elements by defining the syntax and semantics of their characteristics, such as interfaces to the elements. MPEG-21 also addresses the necessary framework functionality, such as the protocols associated with the interfaces, and mechanisms to provide a repository, composition, conformance, etc.

II.3.3.4 Multicast services

The Multicast traffic is currently only 1% of the total Internet traffic. This may substantially change when, as it is targeted by the source-specific multicast model, any unicast source address may host multiple web-pages which operate as multicast sources. Indeed, many User Multicast Applications can be envisioned: hotel-videos combined with web-based hotel reservation; web-based trade shows; web-based education; web-based TV; etc.

Furthermore, Multicast may serve as LAN emulation and play an important role in network constructions.

Chapter II.4 – Quality of Service

The notion of quality of service as applied to a telephone call between two users is fairly broad in scope. Apart from those parameters that are intrinsic to the network and which enable us to quantify the quality of service provided, other facets must also be taken into account when it comes to measuring that quality from the standpoint of the user, i.e. user-friendliness, service availability, security, service intelligibility, etc.

In this section we shall restrict our remarks to the quantifiable aspects of quality of service, such as transmission time, error rates, etc., as well as aspects relating to the service provision model and organization of the network. Providers engineer their networks/capabilities taking into account both the Quality of Service (QoS) objectives that will satisfy their customers and the impact on network costs since customers will be reviewing both price and performance in making their purchasing decisions.

Other aspects, such as security and the numbering plan, will be dealt with in subsequent chapters.

II.4.1 Quality of Service in the context of the telephone network

II.4.1.1 Technical aspects

In the context of the conventional telephone network using circuit switching, it has been necessary, in order to analyse the quality of speech reproduction, to define the notions of *intelligibility* and *listening comfort*. In the case of the telephone service, these quality criteria are applied "end-to-end" for a complex link (national to long-distance or international, for example, across several telephone exchange and transmission system hierarchies), the objective being to produce recommendations for each of the systems (i.e. links in the chain) involved in this end-to-end connection. Where adhered to by all the operators concerned, such recommendations result in the most complex of calls, both national and international, being usable by the call parties. The main sources of performance degradation lie in:

- the numbering procedure;
- echo experienced by the speaking or the listening party;
- transmission error rates.

Network congestion is not taken into account since, where circuit switching is concerned, it ultimately leads to call unavailability but does not affect the quality of established calls. As for error rates, these tend nowadays to relate solely to the mobile sphere, since fixed transmission media are characterized by an excellent level of quality.

Digitization

Digitization at 64 kbit/s, standardized in the late 1960s, ensures excellent quality, although it needs to be borne in mind that several analogue/digital and digital/analogue conversions can affect call quality. However, with the digitization of transmission now being widespread, the phenomenon of successive conversions is becoming a thing of the past.

For reasons of economy, operators use speech compression systems – circuit multiplication equipment (CME) – on intercontinental links with a compression factor of up to eight and with reliance on the non-simultaneity of activity on the part of the two parties to the call and on redundancy in the voice signal. The level of compression varies according to traffic volume, the maximum compression rate being applied only during busy periods. Such compression is characteristic of a transmission link between two switches. If a call is carried over several links each applying compression, the performance degradation will accumulate; however, the rule describing the perception of such accumulated degradation is complex. Efforts are thus made to limit the number of times that a single call is compressed/decompressed.

Echo and delay

Any transmission involves a propagation time, which can be as much as a fraction of a second (geostationary satellites). In the case of transmissions by cable or optical fibre, it is a little over 3 ms/km. The distance as the crow flies must be increased to take account of detours due to the terrain and also to the additional paths that have to be used in order to protect against interruptions.

Conventional circuit switching involves a delay in the order of one quarter or one half of a millisecond per switch passed. All in all, a connection that does not go via satellite will experience a delay of a few milliseconds, increased by approximately 8 ms per thousand kilometres. Where voice transmission by ATM is used, and one virtual circuit is used for each channel, a delay of at least 6 ms, corresponding to the "insertion into cells", must be added.

This delay creates a difficulty for the party who is about to speak, particularly where both parties begin to speak at once and need to decide who will proceed. This situation does not arise in the case of national land connections using circuit switching between two parties, since here the delays are sufficiently short. However, where delays amount to a fraction of a second (as is the case with a geostationary satellite hop), the problem begins to become noticeable, and calls for a measure of discipline on the part of those at each end of the call, particularly in the case of a telephone conference. In principle, one does not come across connections involving two geostationary hops.

The phenomenon of echo is due to insufficient separation between the two directions of transmission at the interface between the analogue subscriber line and the time switch. The transmission delay has the effect of rendering more noticeable, and hence more annoying, a coupling between the two directions of transmission. It is for this reason that connections involving high propagation times are equipped with echo cancellers, these being devices which auto-adaptively produce a signal which counterbalances the echo signal.

II.4.1.2 Aspects relating to network organization

Over and above the technical elements referred to above, it is important to note that, of all the telecommunication network characteristics that contribute to quality of service, the main one is "recognition" of the semantics of the application (speech transfer) by all of the active components within the telecommunication network (i.e. the switches).

This recognition of the application's semantics goes beyond the mere reservation of a resource in the form of a 64 kbit/s circuit throughout the duration of the call¹⁰, the quality being ensured above all by the signalling dialogue that is exchanged end-to-end between all of the network's active components. This permits the necessary resources to be reserved and maintained throughout the duration of a call. The cost of the resources mobilized for a given call is as much due – if not more so – to the maintenance of a "state" for that call among all the network's active components as to the transport resources actually mobilized (circuits in the case of today's telecommunication networks).

Indeed, even where a packet mode of transport is used – see the discussion on data networks below – it would be difficult to do without the call-by-call reservation of adequate network resources if the aim is to be able to ensure quality of service.

Likewise, this way of operating could just as well apply to a packet transfer mode as to a switched circuits mode. The BICC protocol (See Section G.2 in Annex G) recently defined by ITU-T demonstrates the possible transposition of the initial approach developed for the circuit transport mode (ISUP protocol) in telecommunication networks to a new mode of transport by packets (ATM or IP).

¹⁰ Such reservation is often criticized for consuming more resources than a packet mode in which resources are used only as required and according to the traffic generated.

II.4.2 Quality of Service in data networks

Data networks were created in order to enable communication between applications running on computers. These were originally large centralized computers. The development of mini- and then of micro-computing together with the development of computer applications based on the client/server model have made a major contribution to the deployment and interconnection of data networks. The equipment connected to a data network therefore essentially takes the form of highly distributed computers that are more or less complex and run several applications.

The problem of quality of service in data networks was from the outset different to that in telephone networks, owing to the fact that the quality of service expected by data network users is not associated with a particular application provided by the network, but rather with properties relating to their points of access to the network. These properties together define what is commonly referred to as the Service Level Agreement (SLA).

The user's access point to a data network – whether public or private – enables communication to take place between one or more machines and all of the other machines connected to the network. The quality of service requirements at this access point form part of the SLA¹¹ and are generally expressed in terms of authorized speed (average and peak), transmission time (average and residual) or relative priority of data in the event of congestion. As can be seen, these are properties that are expected from the network for the transport of data in support of one or more applications of which the network has, *a priori*, no knowledge.

As is the case with user access points, the points of interconnection between networks are also governed by application-independent SLAs whose SLS express only transport properties.

The transport mode most often used by data networks is thus the packet mode, a choice which stems from the sporadic nature of the data transmitted by computer applications. The data presented to a network access point are assembled into packets of fixed or variable size according to the nature of the network. In each case, the packet header contains the destination address, thereby enabling the network components to route packets towards their final destination – or the nearest point thereto – without taking any account of the content of the packet, which will be processed by the appropriate application in the destination machine. It is possible to effect a packet transport mode by means of virtual circuits (as, for example, in ATM networks), thereby better ensuring quality of service¹². However, IP networks do not, at the basic level, provide for the virtual circuit mode. Protocols such as IntServ or MPLS enable such an extension on top of an IP network, but are still far from having been deployed in all IP networks.

Thus, where an application uses one or more data networks, the resulting quality of service end-to-end between the two machines working on the application in question will depend on the quality of service ensured by all the networks used. It is for this reason that Internet quality of service is often criticized. It needs only one network to have a less than acceptable quality of service for the end-to-end quality to suffer.

It is for this reason that when companies use the Internet to interconnect remote machines for their strategic applications, they often turn to the services of virtual private network (VPN) providers. The VPN provider sets up, by specially engineering the signalling arrangement on top of the Internet, a virtual network that ensures acceptable quality-of-service properties among all of the company's

¹¹ Actually, it is more proper here to use the term Service Level Specification (SLS) rather than SLA, the former being the technical specification of the latter which can be considered as the legally binding agreement.

¹² In this case, all the packets pertaining to a single application may use an identical route – rather than a random one according to congestion – on which a number of resources have been reserved, thereby making it possible to specify the behaviour of the network with respect to quality of service properties.

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access points, which can include certain dynamic access points for remote users. A VPN can of course only be set-up by reserving resources in all of the physical networks that support it; such a service must therefore be paid for, and is currently intended only for business customers¹³.

II.4.3 Quality of Service of an IP network used for telephony

One of the main difficulties in IP telephony is to achieve a quality of service similar to that which users are accustomed to in telephone networks.

This difficulty stems, on the one hand, from technical considerations that are specific to the data transfer over IP networks mode and, on the other hand, from those relating to the organization and mode of provision of service over data networks in general, and IP in particular.

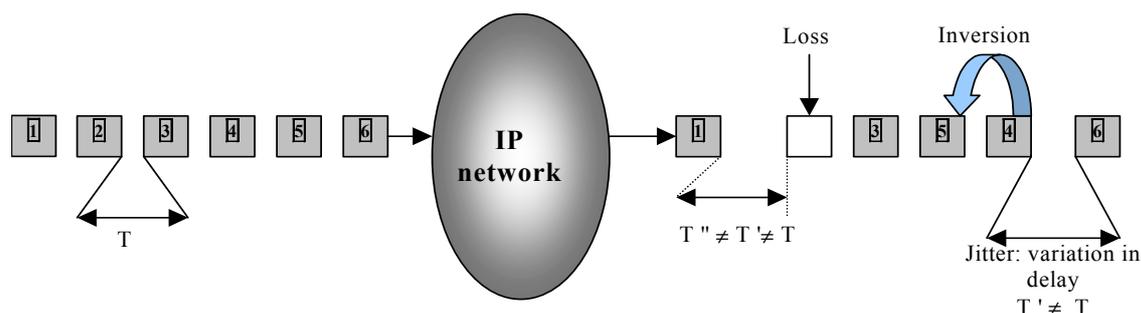
II.4.3.1 Technical difficulties

The packet mode of data transmission used by IP networks introduces degradation factors into the quality of communication. We may list four main sources of difficulty associated with the packet mode of transmission that have an impact on the transport of voice over IP.

- **Loss:** possible disappearance of packets during the communication. The incidence of this factor will be moderate if the loss ratio is low.
- **Delay:** this refers above all to the overall transit time, including the time taken to reconstitute the order of the packets upon arrival and compensate for fluctuations in transit times (this overall transit time must be lower than 400 ms if interactive conversation constraints are to be respected).
- **Jitter:** variation in the packet arrival delay.
- **Echo:** this refers to the delay between the transmission of a signal and receipt of the same signal as an echo.

Figure 15 below summarizes the above-mentioned difficulties and Annex B discusses in more detail each of these factors.

Figure 15 – Main difficulties in IP telephony transmission



¹³ It is appropriate to mention at this point the complaints that are made by private users concerning Internet quality of service, even when the communication with their ISP – in some cases high speed, for example by cable or ADSL – is not the problem. The contract between the user and the ISP relates solely to access by the former to the network of the latter, and has nothing to do with any other networks the user's packets may transit in order to reach their final destination. This is a clear illustration of the key difference with respect to the telecommunication network service provision model, where the operator's responsibility extends all the way to the final destination of the call, wherever that destination may be located.

II.4.3.2 Technical solutions for provisioning QoS over IP networks

The Internet Engineering Task Force (IETF) has defined many protocols and methods for the provisioning of QoS over IP networks; among them one can mention:

- Network over-provisioning or for corporate networks switched LAN – may not be economically viable
- Quality feedback (IETF RTCP) – application centred, no network impact, admission control/traffic management
- Priorities (e.g. for queuing resources, DiffServ)
- Resource reservation (RSVP, IntServ)
- Traffic segregation (transport and routing)
- Traffic engineering

In addition to the Internet Protocol capabilities such as RSVP, IntServ, DiffServ, IP telephony QoS can benefit from features in suppliers' equipment that offer a variety of queuing, traffic shaping, and filtering technologies for implementing traffic priority and controlling congestion end-to-end across the network. Some examples are:

Custom Queuing (CQ) handles traffic by assigning different amounts of queue space to the various classes of packets and then servicing the queues in a round-robin fashion. While a particular protocol, user, or application can be assigned more queue space, it can never monopolize all the bandwidth.

Weighted Random Early Detection (RED or WRED) combines IP precedence and Random Early Detection (RED) capabilities to provide differentiated performance characteristics for different classes of service. RED provides the ability to flexibly specify traffic handling policies to maximize throughput under congested conditions.

Committed Access Rate (CAR) provides the means to allocate bandwidth commitments and limitations to traffic sources and destinations, while specifying policies for handling traffic that exceeds bandwidth allocations.

An outline of some of the major protocols defined for QoS is given in Annex C.

Of course, none of the above protocols and methods would be sufficient alone to provide an acceptable quality for voice unless in very specific and favourable circumstances. A network engineering involving a combination of methods and protocols is likely to be required. On the other hand, though many of these protocols are implemented within actual products, there is still a lack of a comprehensive packaging that ensures a seamless guarantee of QoS for voice applications. A case-by-case study is therefore needed.

Still, the above are techniques applicable for a given subnetwork domain; how can we ensure the appropriate QoS provisioned end-to-end for a voice call traversing multiple operator domain¹⁴? This necessitates a discussion on how so-called next-generation networks exchange information about an incoming call so that resources can be appropriately reserved within each involved subnetwork to ensure a proper QoS end-to-end.

II.4.3.3 Aspects relating to the organization and model of service provision by IP networks

The very notion of *telephony* over IP – or over any other packet-type mode of transport – implies the provision of an interactive voice dialogue of an acceptable quality between parties.

As was illustrated in the first chapter, service offers which provide IP telephony to users connected to legacy telephony services require the intervention of a provider operating a managed IP network (i.e. one in which quality of service is guaranteed) and a gateway to the conventional telephone world. We

¹⁴ It seems that IETF is becoming aware of this inter-domain consistency of Quality of Service and a specific working group on that subject started work in December 2001 and is named "Next Steps In Signalling (NSIS)". See <http://www.ietf.org/> for more details.

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have also discuss in II.4.2 above, the service level specification which currently governs access to and the interconnection of data networks. That specification does not relate to the semantics of a particular service provided by the network, such as telephony, but to a set of basic properties characterizing the transport of data by the network.

One might therefore ask the following question: from the strict standpoint of quality of service assurance, and assuming that a generalized access in native IP mode becomes widespread, under what conditions will those constraints imposed by telephony be ensured consistently over a succession of interconnected IP networks?

The response to this question is fairly complex, while an appreciable number of technical considerations that form part of that response are currently at the research stage or have not yet been deployed to any significant degree in data networks.

It is nevertheless possible, on the basis of simple observations based on common sense, to identify the following evolutions at the level of IP network organization that will be necessary in order to be able to speak of a genuine IP telephony service according to the above definition:

- There can be no denying that the "free" Internet model only makes sense if the interconnection between the networks is based either on barter – i.e. equivalent traffic in the two directions – or on all-inclusive interconnection models without guaranteed quality of service. Quality of service has to be paid for, whatever the mode used to achieve it, even if that mode involves an Internet overlay (see the example of virtual private networks (VPN) discussed in II.4.2).
- Given that voice requires the strict guarantee of a certain number of quality-of-service parameters, its provision as a service provided by a third party cannot be free of charge. It could one day be provided on an all-inclusive basis, but a commercial evolution of this type has already taken place in the conventional telephone networks.
- If it is to honour the service it provides to a customer who must pay for that service, the network operator, in addition to ensuring the quality of its customer's connection to its network and, if need be, within that network, as is currently the case with Internet access providers, must also be able to extend that quality through all of the networks used by a given communication, right up to the final destination.
- In order to ensure end-to-end quality, there has to be a dialogue between networks, and that dialogue has to take place for each new call, as is the case in the current telephone network.
- Some people may take the view that a sufficiently over-dimensioned SLA between operators could ensure this "propagation" of quality of service without the need for call-by-call signalling and reservation. Such an arrangement would be economically viable, and hence foreseeable, only if voice – or any other type of traffic requiring the strict reservation of resources – did not exceed a minimal proportion of the traffic carried between the two networks.
- In the event that the configuration described above for the sharing of traffic between voice and data was not feasible and that the network was essentially used for the transport of voice, call-by-call signalling for the reservation of resources would thus be necessary. The BICC protocol standardized by ITU-T proves that such a signalling arrangement is possible on top of an ATM or IP packet transport network.

Chapter II.5 – Security

In the telecommunication sector, security has become an increasingly imperative requirement. The opening up of the global telecommunication market to competition, on the one hand, and the evolution of transport technologies in telecommunication networks, on the other, have served to accentuate the importance of security for the various players, i.e. users, who require that their communications be kept confidential in order to safeguard their private lives; network operators, who need to protect their activities and financial interests; and finally, regulatory bodies, which require and impose security measures by publishing directives and issuing regulations to ensure the availability of services.

The drawing up, in a manner that is formally correct, of a well-defined set of requirements in respect of security services remains a fairly abstract notion, since each network has its own characteristics and, above and beyond the technology used, security solutions depend on a variety of factors. However, we are able to point to two call protection strategies. One of these consists, for the users of communication services, in ensuring for themselves the protection of their calls. In this case, the public network does not intervene. This type of protection is known as end-to-end protection. The second strategy consists in wholly or partly delegating responsibility for the protection of calls to the public network, which must ensure protection on portions of the network, each of which is located between two sets of public network security equipment.

The most important features characterizing network security are the following:

- **Confidentiality**, whereby a call between two correspondents is protected against illegal tapping by an unauthorized or ill-intentioned third party.
- **Authentication**, whereby an entity can be sure that the data received actually come from the stated transmitting entity.
- **Access control**, which is a service whereby access to the network resources (server, switch, router, etc.) is restricted in accordance with the security policy in force. Otherwise, if an ill-intentioned individual manages to obtain unauthorized access to one of the network resources, he or she is then able to launch attacks such as illegal tapping or denial of service, which consists in the continuous transmission of data to network elements such that no resources remain available for other network users.
- **Integrity**, whereby an entity can be sure that the data received have not been modified in any way during the course of their transfer. It is possible, by means of this service, to eliminate the risk of data corruption as a result of deliberate and ill-intentioned manipulation.

II.5.1 Security in the context of the telephone network

As we have already seen (see the chapter on network architecture), one of the main differences between the telephone network and IP networks lies in the concentration of intelligence and in processing within the network at the level of switching nodes. Thus, in the case of the telephone network, protection is fully the responsibility of the network, and users play no part whatsoever in that regard. The fact that the intelligence is located within the switches considerably reduces the risk of ill-intentioned attacks since, in order to make the network malfunction, an ill-intentioned individual must have access to the public switches. However, despite that obstacle, we could hardly claim that the conventional circuit switching telephone network is today totally free from criminal activity or hacking. An illustration of this lies in the fact that, some ten years ago, the telecommunication network experienced a fundamental change with the introduction of the intelligent network using Signalling System No. 7 (SS7). This development, while it provided greater flexibility to the network through the introduction of new services, at the same time increased its vulnerability to the misuse of those services, an example of which is the freephone service. Certain services, moreover, are more open to misuse since their utilization requires that users have access to management information.

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As regards the security features referred to above, we may point out the following in respect of the telephone network:

- Where confidentiality is concerned, the telephone network provides full confidentiality that is limited solely by the legislation in force (tapping of a telephone line by the national authorities).
- A telephone call can only be established if the calling party can be properly identified by the network, such authentication being an essential factor in service billing. With the telephone network, it is therefore possible at all times to know the two parties to a call (i.e. the called and calling parties).
- The fact that the switches are generally housed in well-protected locations (telephone exchanges), it is a simple matter to establish an access control system designed to minimize the risk of attack by an anonymous individual. Furthermore, switches generally allow for the safeguarding of all actions initiated from a maintenance console, access to which is in the majority of cases protected by passwords.
- The circuit switching used in telephone networks, whereby a 64 kbit/s (or 32 kbit/s) circuit is reserved throughout the duration of a call, facilitates the task of ensuring call integrity.

It should perhaps be noted here that there are two types of flow within a telecommunication network, the first being the flow of voice calls, the second being the flow of signalling and management information. The latter flow, which is crucial to the proper functioning of the network, is for the most part made up of signalling messages (as in the case of SS7) and is transported over a network that is – at least functionally – parallel to the one used for transporting the user traffic. There are a number of signalling network topologies. In most cases, the signalling points (SP) are a physical part of the switch, which is why it is important to ensure the effective protection of the switches and to pay great attention to the signalling network's points of entry into other networks.

II.5.2 Security in the context of the IP network

In IP networks, most of the processing needed to establish calls is delegated to the user terminal equipment. The intelligence is hence deployed to the ends rather than to the nodes of the network, as in the case of telecommunication networks.

It is therefore clear that the security functions as well will to a large extent be ensured by users, and, as the case may be, by the end routers, and not by core equipment within the network.

Two scenarios emerge when we look at security issues in IP networks. The first is that of a proprietary network, otherwise known as a managed IP network, where the management, maintenance and operating functions are the responsibility of a well-identified party exercising the role of operator. In this case, the network manager could introduce protocols and equipment for the purpose of implementing security services within the network, whereupon the responsibility of ensuring communication security would be borne in part by the network. The second case is that of the Internet, which is, in fact, the interconnection of a very large number of IP networks worldwide. The lack of any party having overall responsibility for this "network of networks" requires that users take full responsibility for ensuring the security of their communications.

Furthermore, security issues were not taken into account at the design stage of the IP protocol. This is why it has been necessary, in the interests of ensuring the protection of communications carried over such networks, to subsequently add security services to the mass of network protocols already in existence. Two solutions have prevailed for the security of traffic transported over IP, namely the TLS (Transport Level Security) protocol, which provides security within the transport layer, and the IPSec protocol. The TLS protocol is implemented on top of the TCP and is therefore only able to protect applications traffic transported over TCP, whereas IPSec is applied at the IP level and is hence more generic than TLS and can be used to provide security for any type of traffic over IP, including the UDP transmissions used by IP telephony. IPSec protocol is discussed in more detail within Annex D.

There are two modes for the provision of security for IP packets using IPSec: the transport mode and the tunnel mode:

- The **transport mode** applies one or more security functions (essentially authentication and encryption) to the IP packet to be transmitted. These functions do not provide complete protection for the header fields. The mode of transport is applicable only to terminal equipment, particularly the end routers. An intermediate router might not apply the IPSec transport mode to an IP packet it is relaying on account of fragmentation and reassembly problems.
- In **tunnel mode**, a new IP packet is created by a method involving the opening of an IP tunnel in IP. The security function(s) which is (are) applied to the external IP packet therefore protect the integrity of the original internal IP packet (header and data), since this constitutes the "data" part of the external packet. This is obviously the best mode for creating secure VPNs and ensures better protection against traffic flow analysis.

With the use of the IPSec protocol in managed IP networks and VPNs, we come back to the telecommunication approach, which consists in entrusting security services to the network, without however removing all responsibility from users, who must continue to ensure a part of those services at the applications level.

The next-generation security architecture defines the overall framework for placing and locating security measures tying together the infrastructure components, the networking components and the applications with services.

- The IP telephony network (IPTN) requires balanced security services on network/transport level and on application level with defined interaction for access security for various application security demands; continuous security across all layers from basic infrastructure IP services (DNS) to the application is the scope of the IPTN security architecture. Secure distributed, de-composed systems will be the major challenge.
- The IPTN security architecture should provide means of how to overcome the limitations of firewalls: decentralized (personal) firewalls or de-composed multimedia firewalls with firewall controllers and firewall gateways that are application aware might be appropriate solutions. Security inter-working among the various secured IPTN and entities will be an issue (e.g. through security gateways).
- IPTN shall provide security interfaces and security APIs.
- Meshing of security islands towards recognized trust relationships. Public-Key Infrastructure (PKI) serves as a security domain of managed trust. PKI will gain crucial importance for scalable security and supporting worldwide trust-relationship. IPTN will have to deploy PKI for provisioning security. PKI concepts and systems need to be enhanced with respect to the multimedia and real-time requirements of IPTN.
- The IP telephony security architecture shall incorporate improved security using state-of-the-art cryptographic algorithms. It is required to provide a security upgrade facility to improve security step-by-step over time.

IPTN security addresses such a broad scope that the architectural concepts must be built in conjunction with the overall IPTN architecture and be mutually aligned. On the other hand, some on-going security activities in the various bodies are important for the IPTN development. The IPTN architecture discussion should probably be one of the first items to start with.

II.5.3 Lawful interception for IP telephony

Lawful interception requirements for wireline and wireless telephony are established in each administration based on national law and are not established by ITU.

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In telephony networks, lawful interception is somehow facilitated by the fact that all incoming/outgoing calls towards a given subscriber have to go through his serving local exchange. There is therefore a trusted single point within the network that can send a "duplicate" of all calls where this subscriber is involved towards the legal entity that asked for the interception.

In telephony over IP, just like circuit-switched wireless or wireline telephony, lawful interception can be more or less complex depending upon the specific requirements that are established in each sovereign country and the model(s) used for provisioning IP telephony.

Chapter II.6 – Coding

The purpose of coding is to transform a voice signal, generally analogue, into a digital signal of a given speed and quality. The first coding operation involves sampling of the analogue signal at a given sampling frequency and with a given accuracy, that accuracy being characterized by the number of bits used to code the amplitude of each sample. It is clear that the choice of frequency and of the number of bits used represents a compromise in terms of the speed/quality of the coded signal. The greater the quality required, the greater the speed obtained after sampling.

The sampling theorem states that an analogue signal may be reconstructed from digitized samples if the sampling frequency is at least twice the bandwidth of the original signal. The human ear being capable of perceiving a frequency range in the order of 20 Hz to 20 kHz, high-quality audio coding uses sampling frequencies higher than 40 kHz.¹⁵

II.6.1 Coding technologies used in the context of the telephone network

The frequency domain (bandwidth) that can be transmitted over telephone lines is officially set at between 300 and 3 400 Hz. The modern codecs (coder/decoders) used in today's telephone exchanges have a bandwidth in the order of 200 to 3 700 Hz, and this had led to a general improvement in the quality of subscriber lines. A bandwidth filter is therefore applied to the outgoing signal, thereby restricting the frequency space allocated to transmission of the signal on that link.

However, despite that limited bandwidth, the frequency of digitized telephone speech – at 32 to 64 kbit/s – is considered fairly high since, for digital storage applications, higher frequency implies more memory while, for digital transmission applications, higher frequency implies higher bandwidth, power and cost. It is in order to remedy this situation that coding systems which allow for compression of the signal have been introduced, particularly in long-distance transmission systems used in switched telephone networks.

Today's circuit switching telephone networks use, for the most part, coding systems based on the time technique, which is characterized by preservation of the waveform of the signal to be coded. According to the quantification method used, we can identify two types of coding: simple PCM coding and differential coding.

II.6.1.1 PCM (pulse coded modulation) or MIC coding

This is the most simple of the coding algorithms used for coding speech in the switched telephone and ISDN networks. It comes down to sampling an analogue signal at a fixed sampling frequency of 8 kHz and quantifying the samples by a value of 8 bits, representing the amplitude of the signal at that very instant on the basis of non-linear compression standards (law of A or μ). Given that the digitization process assigns a specific binary number to each amplitude of the signal, and that there are only 256 amplitudes for 8-bit digitizations, it is probable that the number assigned will not correspond exactly to the true value of the signal. This error is known as the quantification error and produces a quantification noise in the output signal. This coding system corresponds to ITU-T Recommendation G.711 and to a data signalling rate of 64 kbit/s.

II.6.1.2 DPCM, ADPCM and ADM differential coding

Differential coding (DPCM: Differential Pulse Code Modulation, ADPCM: Adaptive Differential Pulse Code Modulation, ADM: Adaptive Delta Modulation) is based on the observation that successive samples from an audio source are highly correlated. It is therefore more advantageous to encode not the samples themselves, but the difference between the successive samples. ADPCM encoders encode the samples differentially with a component estimated by extrapolation from the preceding values. This coding system, which corresponds to Recommendation G.721, uses only 32 kbit/s per voice channel.

¹⁵ 44.1 kHz for CD coding and 48 kHz for DAT coding.

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The other two differential coding methods (DPCM and ADM) are characterized by the method used to predict the value of the following sample on the basis of the value of the preceding one. There are differential coding options which give data signalling rates of 16, 24 and 40 kbit/s; however, the speech quality deteriorates very rapidly when the rate falls to 16 kbit/s. For more details on the above protocols, see Annex E.

II.6.2 Coding technologies for telephony using an IP network

The audio quality obtained over the Internet is part and parcel of the service it provides. That service comes down to providing applications with a transmission channel whose characteristics, such as delay, bandwidth or loss rate, may vary considerably over time. Generally speaking, two approaches, which are often used simultaneously, may be envisaged in order to harness that quality. The first approach consists in adapting the network services to the requirements of the applications, which actually comes down to modifying the protocols and mechanisms used by the network to provide new services that are tailored to the requirements of the applications (telephony in our case). To that end, it is necessary to define a range of services and deploy mechanisms such as the reservation of resources or allocation of resources within the routers, which will then be able to provide the required services. The second approach consists in adapting the applications to the network services, i.e. arranging things in such a way that the application itself compensates for the undesirable effects of a network of the best effort type. This comes down in practice to engineering things in such a way that the application adapts its behaviour in accordance with the characteristics of the connection over which it is transmitting its packets. In the case of voice transmission, the aim of the adaptation exercise is to achieve the best possible sound quality at the receiving end, given the status of the network. It is within the framework of this latter approach that coding techniques of a far higher efficiency than the time technique have been developed and are being used in audio-video transmission over IP networks.

The speech encoders currently used in IP telephony applications may be grouped according to three major coding techniques:

- Time techniques (bit rates between 16 and 64 kbit/s);
- Parametric techniques (bit rates between 2.4 and 4.8 kbit/s);
- Analysis-synthesis techniques (bit rates between 5 and 16 kbit/s).

The first category has already been presented in the above paragraph, and encoders based on the time technique are widely used in conventional telephone networks. The latter two are discussed in Annex E.

The latter two categories of coding techniques (parametric and synthesis) afford the advantage of low bit rates. However, according to the well-known principle whereby any process has a cost in terms of time, the greater the compression rate, the longer the delay resulting from the processing stage will be. An optimum compromise needs to be found between the bit rate and the associated processing delay.

The following table groups together, for the majority of coders mentioned above, the main characteristics in terms of: bit rate; speech quality as an MOS (Mean Opinion Score), the average MOS being established in a standardized manner on the basis of five categories (1 = poor, 2 = mediocre, 3 = fairly good (average), 4 = good, 5 = excellent), for clean speech conditions; complexity of execution (in MIPS DSP fixed 16 bit); and coding/decoding delay.

Table 1 – Characteristics of speech coders for IP telephony

Coder	Standard/ Recommendation	Bit rate	Quality of speech (MOS)	Coder/ decoder delay	Complexity (MIPS)
Time PCfM	G.711	64 kbit/s	4.2	125 μ s	0.1
Time ADPCM	G.726	32 kbit/s	4.0	300 μ s	12.0
Analysis-synthesis RPE-LTP	ETSI – GSM 06-10	13 kbit/s	3.6	50 ms	2.5
Analysis-synthesis CELP	DD FS1016	4.8 kbit/s	3.5	50 ms	16.0
Analysis-synthesis LD-CELP	G.728	16 kbit/s	4.0	3 ms	33.0
Analysis-synthesis CS-ACELP	G.729	8 kbit/s	4.0	30 ms	20.0
Analysis-synthesis MP-MLQ-ACELP	G.723.1	6.3 and 5.3 kbit/s	3.9 to 3.7	90 ms	16.0
Parametric LPC	DOD LPC10 FS1015	2.4 kbit/s	2.3	50 ms	7.0

We may thus conclude that coding systems have seen a tremendous evolution in recent years, and that this has considerably reduced the need for bandwidth for various telecommunication services, particularly voice transmission. As things stand, these coding systems have reached maturity, and work is still under way on the development of new and even more efficient coding systems. Today, with coding systems using the analysis-synthesis technique, these being the most efficient systems for the application of IP telephony, great progress is being made toward guaranteeing quality of service for this type of application over IP networks.

Chapter II.7 – Accessibility

II.7.1 Access to the telephony network

As is well-known, the value of a network grows exponentially with the number of its subscribers. Today, the worldwide telephony PSTN/PLMN network has 1.2 billion subscribers; the potential of communication services that can be sold by any subnetwork operator is directly related to the number of its own subscribers but also to the above figure of worldwide telephony subscriber number.

It is for this reason that ubiquitous access to the telephony network – especially in developed countries – is the result of a virtual circle where improved access leads to improved demand thereby leading to improved access. Another consequence of this is that the cost of access can fall dramatically in such a case; operators might be willing to subsidize access or even give it for free provided that they can expect to sell services through it; this statement holds even in wireless PLMN networks where the cost of equipment – that can be comparable in technological complexity to a PC – is relatively high.

Two fundamental technical aspects of access to telephony networks – that are consequences from the above economic considerations – are therefore:

- a clear delimitation of the responsibility of the user equipment regarding the communication application offered by the network: the user equipment, however sophisticated, is a passive entity whose responsibility is strictly limited to request a communication service from the network; it never mixes or intervenes with the communication application logic;
- a clear separation between the customer premises equipment (CPE) and the network (the so-called T reference point): this separation is useful not only for insulating the network from failed user equipment but also determines a reference point delineating the responsibility domain of the network operator.

Technically, access to a telephony network is considered from the user standpoint as cheap and reliable; access equipment for both fixed and mobile network is standardized and can be purchased independently from the operator. Only the access to the *subscription* can be expensive – especially in developing countries with low teledensity – mainly because the expected revenue from provisioning the access is not in accordance with its costs¹⁶.

II.7.2 Access to data networks and to the Internet

Data networks are not determined, *a priori*, by any specific application that they support; they are basically used as a universal transport media for a multiplicity of applications hosted by the computers they help connect to each other.

Access to data networks was, up to recently (early nineties), *de facto* limited to industry, government and academia for their own personnel connected through computers (mini- and later micro-computers or PCs) to their organization's local access network (LAN). Even public data networks were restricted to customers belonging to the above categories for leased lines or virtual private network services. Basically, access to data networks was historically not open to the public at large, paradoxically not because a data network is inherently costlier than a telephony one (it is rather the reverse) but because a data transport service – even when sold through a public operator – is not as high on the value chain (for the general public) as the person-to-person communication service provided by the telephony network.

It was only the advent of the Internet that allowed the first large-scale interconnection of data networks worldwide, the emergence of Internet applications having an appeal for the general public (web

¹⁶ As a case in point, one can observe the high success of mobile telephony even in low-income developing countries where the relative lower cost of provisioning access (with respect to a fixed network), on the one hand, and the higher expectation of revenue (attraction of the mobile service), on the other hand, led to a rapid explosion of such networks.

browsing, electronic mail), and advances in personal computer technology that led to a demand for public access to data networks – and more specifically to the Internet.

Still, paradoxically, it is only through the ubiquity of the telephony network, especially in developed countries, that Internet access became feasible for the general public. As discussed earlier in this document (see chapter I.1), access to the Internet for members of the public is still dominantly made using a telephone line – generally a fixed one – through a normal phone call towards an ISP. The result of this operation is that during the phone call the home computer is allocated an IP address by the ISP; in fact, it becomes connected to the ISP data network – and can use all sorts of Internet applications, including Internet telephony.

Future access to the Internet by the general public will eventually evolve towards so-called "broadband access" where, essentially, the user is connected to the Internet in "native" data mode and is permanently allocated an IP address. Many technologies exist for such a broadband access, such as DSL (over the copper wire of the legacy PSTN), wireless local loop or cable networks, to cite the most important. Moreover, new user appliances are emerging that allow access to the Internet without the need of a general-purpose computer type equipment.

II.7.3 Access to IP telephony and next-generation networks

There are mainly two types of access to IP telephony: through a normal telephony handset or using computer equipment.

As illustrated in Scenarios 2 and 3 of chapter I.1, using telephony equipment is possible only when an operator has deployed a gateway on the side of the telephony user that is still managed by a subscriber telephony switch (or local exchange). Telephony over IP refers here to the usage of a transport over IP technology for a more or less substantial portion of a long-distance call. However, in future evolutions, a subscriber media gateway – as discussed in section 4.4 – controlled by a media gateway control function replacing (or as an extension/evolution of) a legacy local exchange might allow a closer extension of a packet transport to the end user.

Ultimately, however, and especially when broadband access technologies in native IP or packet mode will predominate, it can be foreseen that a new kind of PC-to-PC¹⁷ communication mode might dominate (though with a necessary interconnection with the legacy base through gateways). This mode could be driven not by the replacement of the old telephony service but also by new possibilities of communication services (for example multimedia). Nevertheless, in order for this mode to develop and eventually have the same ubiquity and acceptability as the current telephony network, it is necessary to harmonize the functionality of terminals, especially the way they request communication services from the network.

Today the PC-to-PC mode is characterized by the use of end-to-end protocols like H.323 or SIP that allow the communicating parties to enter into communication with each other with the help of some network servers. Still, despite the presence of those servers, the communication – and even the agreement on the coding used for voice or video for instance – is made end-to-end.

This raises questions about the universality of this communication mode (see also the numbering chapter hereafter) because it necessitates that communicating parties use compatible versions of the same protocol.

Many equipment manufacturers are seeking to develop a new breed of multimedia call control engines that will solve this problem by allowing end users to direct their communication requests towards this engine – and not directly towards their peers – in whatever protocol of choice (H.323, SIP or others). The advantage of this scheme is that not only the protocol compatibility issues are solved but, most important, this engine is capable of provisioning the needed resources within the network such that the communication service is provisioned with a good quality (see QoS chapter above).

¹⁷ As noted earlier in chapter I.1, the term PC may denote here any type of appliance capable of running a VoIP application.

Chapter II.8 – Addressing and numbering plans for telephone services for native IP subscribers

One of the main advantages of the worldwide telephony network is the existence of a universal numbering scheme for subscribers that is agreed upon internationally under the auspices of ITU (Recommendation E.164).

This universal numbering plan allows the provision of the universal communication service: any subscriber of a subnetwork telephony operator can be reached from any subscriber of any other subnetwork operator in the world using the same E.164 number allocated upon his/her subscription to his/her operator. This number is universal and, most important, its integrity and uniqueness is guaranteed worldwide. One can even say that this feature – together with the contractual network guarantee of quality of service – will become the key characteristic of the telephony service in the age of decreasing transmission costs.

One of the technical challenges raised by the ever-closer integration between circuit-switched and packet-switched networks is how to address calls that pass from one network service to another. Generally, it is assumed to be desirable that an integrated global subscriber access plan exists. For example, the same ITU-T E.164 telephone number would reach a subscriber regardless of whether IP-based or PSTN network technologies are used.

It is now widely possible to originate calls from IP address-based networks to other networks, but it is uncommon to terminate calls from other networks to IP address-based networks (except in very particular cases of IP PABX as illustrated in chapter I.1). Rather, calls are generally terminated on the PSTN, so the called party can only use a terminal device connected to those networks. In order to access a subscriber on an IP address-based network from the PSTN, some sort of global numbering/addressing scheme across both PSTN and IP address-based networks needs to be developed and implemented.

ITU-T Study Group 2 (SG2) is currently studying a number of possible options whereby users in IP address-based networks can be accessed from/to PSTN users. One option is the assignment of E.164 numbering resources to IP devices. Another approach is to support service interworking between different subscriber addressing systems in the PSTN and IP networks; for example, using the IETF's ENUM protocol (see Annex H for more details). ENUM defines a Domain Name System (DNS)-based architecture and protocol for mapping an E.164 telephone number to what are known as Uniform Resource Identifiers (URIs). URIs are strings of characters that identify resources such as documents, images, files, databases, and e-mail addresses. For example, <http://www.itu.int/infocom/enum/> is the URI for the ITU website providing an overview of ENUM activities.

There are still today unresolved technical issues related to ENUM. They essentially deal with the consistent behaviour that has to be ensured between the telephony and Internet segments of a call established between a telephony subscriber and an IP one.

In other terms, how can end-to-end consistency be ensured regarding Quality of Service, security or billing, to cite the most important issues?

On the telephony side, there is a party responsible for the call, i.e., a telephony operator, that delivers the above-mentioned attributes in accordance with the telephony model where a network operator is responsible for the call seen as a network service for which it is responsible.

On the Internet side, the IP subscriber is managed by an ISP, or ITSP (Internet Telephony Service Provider). If we take the Quality of Service issue for instance, to what extent does the ISP/ITSP have control over the quality of the communication towards the "ENUM subscriber" if it is not responsible for all of the traversed IP networks within the IP segment of the call? What entity can be held responsible – towards the telephony operator – for the quality of the terminated call? If E.164 ENUM numbers should not – *a priori* – be discriminated against normal telephony E.164 numbers, then appropriate solutions to the above issue and others have to be found.

Chapter II.9 – Conclusions to Part II: Technical Aspects

In this report we have sought to describe the technical characteristics of IP telephony, the main issues and the associated solutions.

After a brief introduction in chapter I.1 of the scenarios under which IP telephony is used today, a working explanation of IP telephony drawn from ITU-T SG2 is proposed. After the checklist in chapter I.2, a comparison is made in chapter II.1 between the respective architecture of data and telephony networks and the rationale for the use of data network type of technology (packet IP networks) to transport both voice and data applications.

Chapter II.2 is dedicated to the discussion of migration scenarios of telephony networks towards the so-called next-generation networks that are based on packet or IP transport. In this discussion, a general step-wise approach has been proposed and each step is discussed both in terms of network evolution as well as the capabilities for services and/or applications that can be offered through it. As a complement to chapter II.2, chapter II.3 discusses some services and applications that can be offered by next-generation networks.

Chapters II.4 to II.8 are dedicated to the discussion of specific technical issues related to IP telephony respectively in terms of Quality of Service, Security, Coding, Accessibility and Numbering. For each of these issues we have tried to offer a comparison between the approach taken by telephony networks, the one generally used by packet IP networks and the one that is needed from IP networks supporting IP telephony.

Finally, it should be mentioned that we found it useful to alleviate the text by introducing a list of annexes explaining details of protocols and methods needed to make IP telephony a working reality. Those annexes do not constitute an exhaustive coverage of all past and ongoing efforts in this domain; they illustrate however that IP telephony cannot be seen as merely a "laboratory reality" and that the technical seeds for a wide-scale deployment of voice over IP transport can be a reality. The main technical issues for the deployment of IP telephony can be summarized as follows:

- While it is unquestionable that technical solutions are well-defined today for the transport of voice over an IP technology, one can hardly ignore the reality that a large-scale deployment of such solutions is still lacking even within developed countries. This has an evident consequence on the maturity of the solutions and of the products supporting them.
- Notwithstanding the above, many major operators and global carriers are reporting a shift to packet or IP transport for the whole or parts of their networks. This IP transport will support both data and voice applications. However, this evolution essentially concerns for the time being the backbone part of their network and does not address end users' access. Only other operators or enterprise networks that have deployed – at their own expense – a unified data/voice access to their users can connect to such networks.
- For the majority of telephony operators in developed countries, the main short-term driver to move their backbones to IP transport is related to the increase of the data traffic they are handling due to the use of their network as an Internet access mean. However, migrating their end users to IP telephony just for the sake of replacing the current telephony service is not considered today as a cost-effective alternative.
- Developing countries are suffering not only from a lack of broadband access but also from a lack of basic access to any telephony network.

PART III

IP TELEPHONY – ECONOMIC ASPECTS

Chapter III.1 – General economic implications of IP telephony

III.1.1 General remarks

Throughout the world, enormous sums are being invested to establish IP-based networks, both for creating new capacity and for enabling existing narrowband networks and future broadband ones to run IP-based services. It is in this broader context that any consideration of the economic aspects of IP telephony should be rooted. The initial driving force behind this investment has been the desire to widen and improve access to communication networks.

One of the reasons in the motivation to send voice over IP on a dedicated enterprise network (intranet) is the cost advantage that this process offers to organizations due to integration of computer and telephony on the same platform. However, this advantage is not obvious when we are talking about a carrier grade network for carriage of voice traffic on a backbone network of the type deployed by local and long-distance carriers in their country, where QoS is not a trivial issue.

Although reducing long-distance network costs is always a popular topic and provides a possible reason to introduce IP telephony, the actual savings over the long term are still under scrutiny and debate. Some commentators believe that the savings from lower prices are based on avoiding telephony access charges and settlement fees. Others believe that IP telephony has the potential to benefit the customers through efficient provisioning of converged services on a single network. Saving on bandwidth cost is pronounced where the volume of data traffic is heavy and has overtaken voice traffic.

III.1.2 Comparison of IP (fixed and mobile, access and core networks) with circuit-switched (fixed and mobile) telephony

An integrated infrastructure that supports all forms of communication allows more standardization and reduces the total equipment and spares complement. The economies of putting all forms of traffic over an IP-based network will pull companies in this direction, simply because IP can act as the unifying agent regardless of the underlying architecture. This combined infrastructure can support dynamic bandwidth optimization. However, the present-generation Internet (IP) was designed only for data traffic and does not guarantee any QoS for real-time voice and video. The standards for real-time traffic are still evolving.

For the sake of comparison, let's take a look at the cost structure between the circuit-switched network and the IP-based next-generation network:

Table 2 – Cost structures

Cost components	Costs with circuit-switched networks	Costs with next-generation network
Carriage of voice calls	Strong distance dependent ¹⁸ Strong dependence on call duration	Weak distance dependent Weak dependence on call duration
Access costs	Relatively low fixed cost per basic telephone line (assuming infrastructure access pipe is available)	Same as for circuit-switched (assuming broadband access is not required)
Customer support	Staff intensive, so high cost or low level of support	Automated, so higher level of customer support for the cost of providing customer support on a circuit-switched network

¹⁸ Distance and call duration are functions of the number and capacity of switches used. The cost of IP routers is much less than the cost of circuit switches.

Table 2 – Cost structures (end)

Adding new services	High	Low
Dealing with growth in data traffic	Very high	Substantial ¹⁹ but much less than for a circuit-switched network
Data services	High because of the requirement to run separate overlay networks	Relatively low, because all services – voice and data – run over a single network

From the above table we can see that:

- The cost of voice calls on a next-generation network deploying IP is not sensitive to distance or duration. So there are opportunities for a next-generation network operator to move to a flat-rate pricing for IP telephony calls.
- Access pricing may be a key issue for next-generation networks. How can they recover the high cost of providing integrated broadband access without deterring customers?
- There are opportunities for next-generation network operators to price services at levels below those of their circuit-switched competitors.
- The next-generation network operators need to think about how they will recover the cost of meeting the data traffic growth generated by their customers. They may expect their network components to double their price: performance ratio over 18 months (as per Moore's Law), but data traffic will increase by as much as ten times over the same period.

Connection-oriented networks are the natural choice of voice/video. In data communication also we have connection-oriented protocols like ATM/FR. In the case of ATM, various bit rates such as CBR, VBR, ABR, etc., have been defined which offer different quality of services and the tariffs associated with each category of service is different. According to ITU Study Group 3, in the case of IP environment, subscribed QoS class (best, high, medium, best-effort) is defined as an IPDR attribute in accordance with the ETSI TIPHON TR-101-329 and DTS-101-512 guidelines, in terms of call set-up time, end-to-end delay, transmission quality and conversation quality. Actual QoS rating for an IP telephony call can be compared against the subscribed QoS class for SLA confirmation and billing adjustment.

III.1.3 Investment costs and cost of operations and maintenance

The case of investment has been discussed from the point of view of operators who wish to take investment decisions for IP telephony, particularly utilizing next-generation networks based on core IP technology. The choice is primarily based on the fact that it generates significantly lower cost. With up to 70 per cent savings on both capital expenditure and operating costs, NGNs make the business plans of operators, particularly CLECs, more attractively to investors, while allowing them to address market segments that were previously unprofitable. The investment in an NGN enables a major increase in potential revenue per customer. It allows CLECs to offer a wider range of value-added services than conventional networks. At the same time, it provides a competitive advantage over rivals with traditional networks. Potential customers are attracted by the future-proofed solutions offered by these networks. Once captured, the customers are easy to retain. The value-added and tailored nature of the services that are possible on the NGN platform help raise the cost for the customer of changing to other service providers, and thereby reduce customer churn. Moreover, the service providers are able to attract more investment as they find it easier to attract capital investment if they propose using IP rather than circuit-switched network technology in their business plan.

¹⁹ Given that traffic is growing at 200% per year (source: OVUM, *The business case of Next-generation IP Networks/* Chapter F).

III.1.4 Human resources including training of personnel in IP networks

IP telephony offers a plethora of services to the users which adds to the complexity at the core of the network. The IP platform is by and large programmable and automated. Most of the existing personnel are attuned to a circuit-switched kind of environment. Their interaction and exposure to IP-related equipment and their operation and maintenance could pose a challenge, particularly to the developing and least developed countries. Since handling and troubleshooting of IP telephony network equipment both require highly skilled IT staff, which is a scarce resource, they add to the cost.

In addition, some of the users might be using IP phones, which are quite different in function, and people in rural and remote areas, where the literacy rate is low, might encounter operational difficulties.

Therefore, administrations may want to consider taking steps to position themselves to take advantage of the IP telephony technology and services by suitably educating the existing technical staff and also by inducting new qualified personnel, which can act as a change agent for imparting the latest know-how required to deal with IP-related products and services. Training courses/seminars can be organized. ITU is in a good position to help the Member States and Sector Members in this regard and the Centre of Excellence (CoE) Programme is one such example where requisite modules are available for the administrations needing help.

Any measure taken by the operators/administrations to train their staff to deal with IP technology adds cost to the overall cost of IP telephony services. Since the training of personnel is necessary in this fast-changing technological world, irrespective of the technology deployed, thus, in the case of IP telephony, the challenge of training can be viewed as an opportunity to invest in precious human resources.

Chapter III.2 – General costing and pricing issues

III.2.1 Costing methodologies on IP telephony

III.2.1.1 General remarks

In a connectionless IP telephony network, it may be difficult to categorize the network elements as "local" or "long-distance" and allocate their costs towards the charges for the local, national long-distance and international calls. Efficient costing and pricing models for PSTN have evolved over the years. Similar models are not available for IP networks, as these networks were mostly engineered for private and business communication or Closed User Groups. Only in the recent past have engineers and economists joined forces to identify the cost drivers of data networks such as IP telephony, but much needs to be done in this regard. A possible issue in this effort is whether the charging for IP telephony can be made on miles and minutes as for PSTN voice or whether it should be based on only the Minutes of Use (MOU).

III.2.1.2 Costing models

Since a data network service can be categorized as a transport media, it is natural to charge for the volume of the transported data or even employ a flat rate that generally gives right to send and/or receive a given amount of data. It is important that all pricing models select the correct cost drivers, e.g. voice telephony cost should include all relevant transmission cost elements.

From the strictly economic point of view, the need to clearly understand the flat rate and volume-based charging models, together with their impact on both the operator and service provider revenues, has emerged. The merger of the two communication models places the revenue value chain at the heart of the debate, as a prerequisite for the successful, widespread merging of IP-based transport and user applications in the current telecom environment.

III.2.1.3 Element-based costing

A change in technology from circuit-switched to IP-based technology can effect the cost of providing and running networks and can alter the basis on which the costs of some key interconnection functions such as call origination and call termination have been based. This may result in regulators needing not only to revise the figures that they determine but also to revise the basis of their determinations to take into account the changes in the nature of the networks. For example, where element-based costing and charging are used, new elements may need to be defined and their costs elaborated.

III.2.1.4 Market forces

Generally, in fully liberalized markets where there is full competition in all the segments of the telecom services including local, domestic long-distance and international, there is little role for regulatory intervention in respect of determination of costs for the services. Another important fact of such markets is that the exercise for tariff rebalancing may have been accomplished in different phases. For example, Hong Kong adopted a market-driven approach whereby individual public telecommunication operators would, according to their business incentives and market competition, develop and implement the required IP telephony services and networks to cater for the market demands.

III.2.1.5 Traffic rebalancing: Cost-based tariffs

In India (as per the TRAI Act), the regulator is required to fix tariffs for various telecom services. The Telecom Regulatory Authority of India completed a transparent tariff fixation exercise in 1998-99. Based on extensive public consultation, the Telecommunications Tariff Order 1999 was issued in March 1999 (TTO-99), fixing the rental, local call and long-distance call charges. The basis of fixing such charges was the underlying cost of the network elements involved in setting up of a local call, a national long-distance call and an international call, in addition to cross-subsidization. Since the

network elements are fixed and identifiable as Local Loop (LL), Local Exchange (LE), Transit Exchange (TE), Transmission System (TS), etc., such an exercise has been relatively simpler. The tariff fixation was done in the light of the National Telecom Policy '99, which lays down affordability and a significant increase in teledensity, as the main objectives of the Government for the next decade. Therefore, rentals have been kept much below the underlying capital costs of providing the local network. Significant cross-subsidy has been inbuilt in the tariff fixation exercise, i.e. cross-subsidy from the long-distance segment of the PSTN to the local network segment has been permitted, till rebalancing is completed in a phased manner. Thus, long-distance call charges have been kept above cost in the interest of affordability of local service, so that people get connected to the network at affordable cost to achieve the teledensity target.

In some jurisdictions, access deficit contributions (ADCs) have been levied on competing call providers so that they contribute to the cross-subsidization of connection and access charges. ADCs are usually a pence-per-minute uplift to wholesale access charges and, in theory, should ensure that all calls (whether they are provided by the incumbent or a competing provider) contribute to the cross-subsidization of access in a competitively neutral and non-discriminatory way. However, in practice, it is difficult for regulators to ensure that ADCs are not harmful to the market (as they will be if their impact on the market is not competitively neutral). For example, ADCs were an element of the interconnection regime in the UK in the early days of liberalization there. However, the ADC system was considered by many in the UK to be overly complex and lacking in transparency, and there was little confidence by new entrants that the ADCs levied by British Telecom (BT) were truly cost based and non-discriminatory. The UK regulator (OFTEL) abolished ADCs in 1995 at the same time as giving BT more freedom to rebalance its tariffs.

In jurisdictions where pricing arrangements include cross-subsidies from long-distance and international call prices to access prices, the relevant authorities may wish to consider whether and how IP telephony will impact on this. However, it should be noted that the most accurate price signals for efficient market entry and investment in liberalized markets can only be provided where tariffs are aligned with costs and no cross-subsidy exists. Re-balancing of tariffs – as is happening in India for example (see above) – will therefore result in more efficient competition and hence increase customer benefit.

III.2.2 Billing issues

Another major issue relates to the payment of carriage charge based on the usage of resource of one operator by another, in a multi-operator long-distance call. These are based on an exact measurement of traffic flowing from one network to another at the NNIs in terms of miles and minutes of use. Sophisticated inter-carrier charge billing systems based on CCS7 signalling have been implemented under the aegis of the regulators in some developed countries such as Japan. Such systems may require extensive exchange of information between network elements of the operators in a multi-operator environment based on CCS7 signalling. Some believe that IP telephony products are particularly deficient in this regard. For this purpose, IP telephony gateways capable of network management, security and number translation or directory function will be required. They may need to be capable of production and Call Detail Records (CDRs) on a real-time basis that include information such as call duration; dialled number; number of packets sent and received; destination; etc. These are required for proper accounting settlement between operators in a multi-operator environment.

In ITU-T Study Group 3, Dr E. Yam introduced a study on standardization of IP detail records (IPDR), carried out within the IPDR.org organization (see below). In contrast to traditional connection-based billing based on call detail records, IPDR.org is now exploring concepts such as content-based billing, QoS-based billing or value-based billing, using the new IP detail records (IPDRs). An IPDR contains information pertaining to the when, what, who and where of an IP-based session/call.

To address standardization issues of next-generation business support systems (BSSs), an industrial initiative has been formed, called IPDR.org. Its stated objective is to define the essential elements of data exchange between network elements, operation support systems and business support systems. It will provide the foundation for open, carrier-grade IP-based support systems that enable next-

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generation service providers to operate efficiently and cost-effectively. The IPDR NDM-U (network data management – usage for IP-based services) document defines the technical specifications of an IPDR reference model and associated IPDR attributes and formats. The newest NDM-U version 2.5 was released in April 2001.

The service model of NDM-U version 2.5 VoIP consists of end-points (phones), service elements (gateways, gatekeepers, soft-switches, and so on), probes, mediation systems and the BSS system. The service scenarios include the inter-working of IP, PSTN and wireless phones. IPDR records are generated as each service element involved in the VoIP call is transferred to the mediation system for aggregation and correlation, and finally they are passed to the BSS systems for billing and recording purposes.

Additional information for a VoIP call, such as QoS parameters, can be collected as IPDR attributes by independent probes and service elements. IPDR QoS attributes include call clarity index (P.562), voice quality MOS, call set-up success and delay, plus IP-domain parameters of packet loss, out-of-sequence packets, and packet delay variation and latency delay.

Subscribed QoS class (best, high, medium, best-effort) is defined as an IPDR attribute in accordance with the ETSI TIPHON TR-101-329 and DTS-101-512 guidelines, in terms of call set-up time, end-to-end delay, transmission quality and conversation quality. Actual QoS rating for an IP telephony call can be compared against the subscribed QoS class for SLA confirmation and billing adjustment.

IPDR records are normally generated at the end of the call. Alternatively, they can be generated while a call is in progress, triggered by certain significant events such as detected frauds or extra long duration.

Special attention is being given for harmonization of definitions and concepts between legacy PSTN services and the new IP and wireless services, for example, E.164 phone numbers versus IP phone addresses, post-dial delay (PDD) versus call set-up time, and so on.

III.2.3 Payback period

The payback period method is a tool to determine how long it would take to recover an investment from the returns attributable to that investment. Generally, it is agreed that, under the payback period method of appraisal, the shorter the payback period, the better.

Carriers that have deployed traditional voice systems within their environments may need to consider the cost of replacing that technology. Although IP telephony offers a rapid return on investment, most traditional voice systems are calculated based on a long-term return on investment (ROI). This leaves the carriers with limited options when determining the overall "cost" of IP telephony deployment. One option is to determine the current outstanding investment on the traditional voice system (PSTN), if any, and account for it as a loss. Another option may be to add any non-returned investment to the cost of IP telephony deployment, which will increase the ROI calculations for IP telephony.

III.2.4 Pricing basis

III.2.4.1 General pricing structure

Circuit-switched traffic is basically distance sensitive and various distance slabs are available for different types of tariffs. The concept of miles and minute may not be relevant in the case of IP telephony traffic because the routed packets may take different paths. There could be two options for pricing: one is based on volume of traffic in terms of transacted bits/bytes and the other could be flat-rate pricing. As an IP call routes through multi-operators, it may be essential that peering arrangements at regional level need to be in place. In addition to the above, the call may be routed through a PSTN operator, in which case there has to be a charge billing system in place, which may exchange CDRs for the billing purpose.

In addition to the above, ISPs typically have the option to negotiate a variety of commercial arrangements in today's market place.

III.2.4.2 End-user pricing

Data gathered in case studies indicate that the new pricing scheme offered by the emerging IP telephony service is another important component affecting the current pricing regime. Data from different countries show prices for IP telephony have turned out to be between 30 to 50 per cent lower than for traditional voice telephony. In Peru, for example, a call to the USA placed through an IP telephony service provider, like Net2phone, offers significant savings compared with the rates of the incumbent carriers. For a PC-to-phone call from Peru to the USA, the tariff per minute is USD 0.15 via Net2phone, compared the USD 0.66 charged by Telefónica del Perú.

In Thailand, IP telephony rates of PhoneNet during standard rate calling hours were between 29 per cent (to South and Central America) to 33 per cent (to Europe and East Asia) cheaper than the rates of traditional PSTN voice telephony offered by the Communications Authority of Thailand (CAT).

In addition to lower rates, the IP service providers claim that there are other benefits to users. Firstly, the service is not charged in blocks or units of time, as in the case of traditional calls, but on the number of actual seconds used. Thus the price of the service will be effectively lower than in the simple comparison. Secondly, with calling cards, users can more easily control their expenditures. Thus it seems that this service will be more attractive for business users who have less flexibility over choosing the best time for making calls.

The evidence from various available case studies indicates that where IP telephony has been allowed, the competition introduced has contributed to a substantial reduction in the prices for international and national long-distance calls, which in turn has brought about considerable consumer benefits. It is likely that the lowering in prices has been to stimulate demand, which will partially offset any possible revenue loss.

Chapter III.3 – Experience from developed and developing countries

III.3.1 Corporate experience

Some carriers have already announced plans to use IP platforms for voice traffic. For the most part, IP platforms are being deployed on private managed networks to enable carriers to provide voice applications to business customers. The advantages of this approach are likely to be in the ability of carriers operating in business markets to serve all the communications needs of their customers (data and voice) on a single IP network, and to provide more value-added and enhanced voice functionality than is possible on circuit-switched networks.

Carriers are also thinking about how IP telephony can be deployed in the PSTN. The economics of migrating PSTN voice traffic to IP platforms can be different to that for deployment of IP telephony in private managed networks serving business customers because the advantage of offering a diverse range of data and voice applications on a single network is unlikely to be a key driver for PSTNs. The case for deployment of IP telephony in PSTNs is likely to focus more on cost savings relative to continued use of existing circuit-switched systems. Carriers considering deployment of IP telephony in a PSTN may also need to consider other issues like how to manage quality – particularly when IP telephony is available across network interfaces rather than, as now, within a single network environment.

III.3.2 India

The Government of India is committed to monitor development in the field as well as the impact of IP platforms on national development, and will review the issue at an appropriate time. At present, national long distance has been opened to full competition and various operators have started applying for the licences. At the same time it may be noted that international long distance is the monopoly of the incumbent operator VSNL (Videsh Sanchar Nigam Limited) and the rights of VSNL for ILD (International Long Distance) went up to the year 2004. However, the Government decided to open the ILD field two years before the schedule and this was due to be done after 31 March 2002.

The Government has referred the issue of the opening up of IP telephony to the regulator of its recommendations. However, the incumbent operator VSNL has plans to use IP technology for real-time service for transit traffic between TAXs and bypassing PSTN TAXs on an experimental basis at six locations in the country. This network will use a separate IP network. At present it is not envisaged to connect this to public Internet. Only domestic long-distance traffic is proposed to be experimented with IP telephony. No IDD calls are proposed in this experiment. This experiment will cover voice real time and fax services, whereas the data services will be transacted through public Internet.

The IP system for transit is planned by VSNL as a pilot project. It will follow ITU-T Recommendation H.323. The system will deploy media gateways, routers, media gateway controllers, signalling gateways, gatekeeper, management server and billing server.

The equipment and the network design is based on highly reliable carrier grade solutions with a high degree of redundancy. The network architecture is evolved keeping expansion and scalability in mind to make any future update as easy as possible. The architecture utilizes a centralized signalling gateway in media gateway controller and gatekeeper and distributed media gateways at six locations, viz. Delhi, Mumbai, Calcutta, Chennai, Pune and Bangalore. The media gateway handles conversion of media stream from circuit format to packet format and vice-versa. The media gateway controller controls and manages the call connection in package network and the signalling gateway shall constitute the interface to the circuit-switched network through an in-band (one of the time slots of E1) or out-of-band signalling network. The routers will route the IP packets between the media gateways. The systems shall also have an IVRS (Interactive Voice Response System) to interact with the customers by sending them voice messages and receiving from them DTMF (Dual Tone Multi Frequency) tones/pulse digits.

III.3.3 Hong Kong

IP telephony is permitted for the operation of networks and services within Hong Kong and for communications between Hong Kong and the rest of the world. Hong Kong adopts technology-neutral approach in licensing of network and services. All forms of IP telephony operations are permitted including PC-to-PC, PC-to-Phone and Phone-to-Phone. Hong Kong follows a market-driven approach that takes into consideration the customers' demands. Hong Kong is not concerned about the bypass of the conventional networks and services by IP telephony. It believes that different types of services should compete against each other on a fair basis. Since January 1999, the external services market has been fully liberalized and ISR (International Simple Resale) has been permitted in bypass of services based on the international accounting rate system. Since then, there has been a steady decline in the accounting rates and the consumer prices for external telecommunication services, much to the benefit of the consumers.

Hong Kong has implemented the re-balancing of the tariffs for domestic and external telephone services. As a result, the tariffs for domestic telephone services can cover costs and the subsidization from the external services to domestic services has been, by and large, removed. Bypass of the external services under the international accounting rate system by services using other technologies, such as IP telephony, would have minimal impact on the development of the domestic telephony network.

III.3.4 Singapore

In Singapore, prior to full liberalization of the telecommunication market in April 2000, only SingTel could provide IP-based telephony service. The only exception was computer to computer telephony calls between two Internet users. With market liberalization, a new Internet-based Voice and/or Data Service class licence was created. Any organization can offer Internet-based voice/data services provided they have a licence and abide by a minimum quality of service. By the end of 2000, as many as 70 companies had been licensed to provide Internet-based voice services. The incumbent operator Singtel has launched a couple of IP telephony services that are significantly cheaper than its normal international tariffs. eVoiz allows users to make a call from their PC to telephone subscribers in selected countries. A one-minute call to the US costs USD 0.05 compared to USD 0.23 per minute per International Direct Dialling (IDD). SingTel estimates that eVoiz will add 10 million minutes of international traffic. SingTel's VO19 service, launched in August 2000, allows any telephone user to make an international call over IP-based networks by dialling a special number. A one-minute VO19 call to the US costs USD 0.11, almost half the normal IDD charge.

Chapter III.4 – Economic impact of IP telephony

III.4.1 Impact on incumbent operators' revenues

III.4.1.1 General remarks

Although no authentic study is available on comparison between IP telephony and PSTN circuit-switched telephony in terms of cost but it is widely believed that IP telephony is a cheaper option. Least cost routing is likely to provide a tool for savings in the transmission and switching cost for the operators deploying IP telephony. In this context it may not be possible to offer a uniform solution for all operators to meet the challenge of IP telephony services. It may depend on the different situation of economy, degree of telecom liberalization in the market, types of services offered by the operators and also by the size of service provider, and the market i.e. the mix of data and voice traffic.

III.4.1.2 Decrease in existing revenues

It is widely accepted that the IP telephony is still in the nascent stage of development and ITU and IETF are jointly working towards improving the quality of service and addressing the interoperability issues between PSTN and IP networks. The cost of an IP telephony gateway switch is four to five times higher than that of a PSTN switch and also the density of port of an IP telephony gateway is smaller than that of a PSTN switch. Because of the increase in carried voice traffic, though IP telephony saves 40 to 60 per cent in transmission cost, mainly because of compression on the backbone, the increase in cost of gateway may be quite significant as compared to the savings.

Least cost routing may reduce the profit margins of PTOs and in developed countries with competitive markets, there is an ongoing price war, which might force the long established, incumbent PTOs to cut their cost base in order to compete against new entrants and resellers. The problem can be more severe for incumbent operators in the case of developing countries, which may rely on international settlement payments for a significant part of their revenue and profits.

The latest settlement rates as of October 2001 provided by the US Federal Communications Commission (FCC) are as follows:

- a) China 36 cents
- b) Korea 19 cents
- c) India 42.5 cents
- d) Malayasia 19 cents (ISR)
- e) Philippines 19 cents (ISR)
- f) Singapore 15 cents (ISR)

The settlement rate figure for India reflects the last one filed with FCC but it has since expired. The US Government does not tract IP telephony traffic and as such does not have a method to extrapolate if there are any lost settlements.

PTOs are caught in a dilemma. If they do not reduce their prices for international service towards cost, then they may lose some market share to IP telephony service providers. On the other hand, if they do reduce prices towards actual cost, their profitability could be adversely affected though the increased volume may partially offset this. A pricing structure based on utility of telephone calls to consumer may be a more logical way forward than a narrow reliance upon cost trends.

III.4.1.3 Eventual creation of new revenue opportunities by including converging data and telecom models for revenue generation

Some believe that the consumers like integrated services even more, given the developing trend of telecommunications and computer communication. With the fast development of IP technology and IP network, more and more users admit that IP will play a very important role in integrating voice, data

and video together. There have been discussions claiming that IP network can provide voice service as good as that of PSTN, if only the IP network can provision sufficient resources, specially bandwidth, for IP telephony services. In fact, it may indeed become possible if the network is designed accordingly and the resources are suitably provisioned.

Providing only a few services with a versatile network may not be a good way for service providers, because by this means the cost of operation and maintenance is too high. With the development of IP technology, it is expected that new value-added service with lower cost could be provided and also create more demand. This is why – although IP telephony systems are still evolving – most operators are paying significant attention to it.

It is widely accepted that PSTN cannot be replaced in one day. Therefore, PSTN and IP telephony will need to exist together for a long time. During this time, the market for IP telephony may grow fast and the market for PSTN may drop down slowly. Presently, the main service method of IP telephony (Phone-to-Phone) depends on PSTN for the access and local exchange port, as the IP version of class V switch is not available, though the kind of PC-to-PC telephony with LAN and Intranet do not need PSTN at all. For some time to come, we may live with a hybrid network, i.e. one based on PSTN and another based on IP telephony, the former offering high QoS in terms of latency and jitter and the latter a lower QoS with a lower price.

III.4.1.4 Economic strategy for incumbent operators

Historically, huge investments have been made in the traditional PSTN network and infrastructure and this cannot be ignored/dumped. In order to protect the investment, the PSTN operators and regulators may have to focus upon drawing up a strategy for survival of PSTN (incumbent) and its existence side by side with IP telephony. Below, a strategy for operators based on four different categories of development is described. While examining the strategies for operators, it may be noted that these are not meant to be prescriptive in approach and as such should be treated rather as guiding strategies for operators. The first three categories have been taken from an APT document, whereas the fourth category has been added, as suggested subsequently.

Category "A" is for developed economies where the telephone penetration density is over 50 per cent and the countries are in the development stage of new technologies and services. In these countries, the telephone market is saturated and data traffic is the predominant traffic on the backbone. Such operators are primarily aiming at data, value-added services and the integrated service market, while at the same time paying more attention to the commercial market. The following suggestions to traditional PSTN operators can be offered:

- Make full use of the current PSTN network resources.
- Lower the cost of PSTN service by effective management, reducing operation costs, and adopt newer technologies.
- Lower, or even stop, the investment on PSTN, especially long-distance transmission network.
- Speed up the development of broadband IP network, so as to provide integrated service and new value-added services, such as IP telephony, broadband Internet access, VPN, VOD, videoconference, er-commerce, etc. Building an IP telephony network just for voice service is not a good strategy.
- Recommend broadband IP service to key consumers to help them to lower the cost of communication and make it easy to do business with others.

Category "B" economies constitute those economies whose telephone penetration rate is about 10 to 20 per cent and is growing at a very fast pace. These countries may want to observe the advancement of new technologies to incorporate these for boosting the PSTN services. The following suggestions to traditional PSTN operators are offered:

- Seek special protection for investments already made so that premature redundancy can be avoided.

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- Make full use of the current network resources to protect their huge investment. PSTN today offers an array of tele and supplementary services as well as IN services, which are far superior to what IP telephony can offer at present.
- Adjust the current PSTN tariff of IDD and domestic long-distance calls. If the competition from ITSPs is very serious, an IP telephony network can be built, but its scale should not be too large. The size of the IP network to be built will depend upon the conditions existing in the country/region in question.
- Lower the cost of PSTN service by effective management, reducing operation and maintenance cost, and adopting new technologies.
- Lower the investment for long-distance PSTN.
- Introduce innovative services so that the basket of services available to the customer is enlarged.

Category "C" are those economies where the telephone penetration rate is between 3 to 5 per cent but increasing rapidly. These countries may be committed to developing PSTN infrastructure. The governments in this category may want to encourage PSTN carriers to continue the development of PSTN. In addition, greater attention should be paid towards the development of IP-based networks and IP telephony including standardization, performance, quality of service, management, regulation, etc., and to introducing them at the proper time in a phased manner. The existing Basic Service Operator/National Long-distance Operators may be encouraged to use IP telephony technology in their backbone network. New entrants to provide IP telephony should be treated as similar to PSTN operators as far as the contribution to USO is concerned.

Category "D" economies are those economies where the teledensity is below 3 per cent. The prime need for such economies is to improve access in the first place. They should seek to improve the global reach of their network since the value of a network exponentially improves with the number of people connected to it. In order to deal with unregulated IP traffic, the long-term solution is thought to be reduced reliance on revenues from International traffic. It may be useful to introduce future next-generation interconnect agreements. It is feasible for telecom operators to manage their own gateways from IP telephony to/from their subscribers. Moreover, using the telecom charging model based on flat rate/volume of traffic transacted, the telecom operator can help make flourish new Internet-based service markets created by new actors by offering a service control platform that will better ensure the necessary revenue streams for all involved. The operator can therefore have, for example, interconnect agreements – in packet mode – with IP telephony service providers to terminate calls in their own network or offer such services for outgoing calls to their own subscribers.

Another idea is to migrate the core network of the network transport technology on to a packet transport for voice services which can go gradually without jeopardizing existing investments. Modern switches will support a migration from TDM circuit-switched transport to packet or IP transport with the same level of service. Communication protocols such as BICC, recently released by ITU-T and currently implemented by all major telecom manufacturers, allow two switches to relay a voice call over an ATM or IP network with the same level of services as over a legacy circuit infrastructure.

III.4.2 Impact of IP telephony on consumers

The economic advantage of IP telephony for consumers can be described very simply: it is invariably cheaper than the traditional alternative, especially for IP telephony. If all other factors (quality, convenience, reliability, etc.) are equal, the choice to use whichever type of IP telephony is cheapest is economically a rational one. In addition, IP telephony offers a plethora of converged services to the users that provide tremendous flexibility and option to the consumers. This fact is becoming more and more important for consumers because it also reduces the inconvenience of using different devices for different applications.

In terms of availability of options, the long-distance operator may choose to offer two types of network in parallel, i.e. one based on circuit switching and another based on IP telephony technology, to offer the so-called "*economy class service*" based on a second "dial tone" from his IP telephony

gateway and a different dialling scheme. The consumer may need to be provided with different tariffs so that he/she can exercise the option from among the available baskets of services offered by the service providers. Tariffs for PSTN long-distance calls are sensitive to both time and distance (miles and minutes), whereas IP telephony calls may have to be charged based on volume of data transported and hence different pricing principles are to be followed. This may pose a real challenge to the regulator who will have to evolve a methodology to fix the rates for these two "classes of service" for the same distance. Similar challenges may be encountered in respect of numbering, seamless dialling, equal ease of access, etc., in a hybrid IP/PSTN multi-operator network.

Consumers may be interested in the quality of service to be provided by the service providers. Where long-distance service providers deploy IP telephony technology to engineer their networks, they must offer the quality of service that is acceptable to the customers and is comparable to a PSTN-based network. In markets which are fully open to competition, the issue of tariffs is decided by the market forces, but in the absence of full competition, a specification of the quality of service by the regulator may be needed, as quality of service may have to be linked to the tariff paid by the customer. Even a lower quality of service such as "economy class" may need to be specified and guaranteed.

III.4.3 Impact of IP telephony on international settlement rates

Internet traffic is generally focused on larger backbone providers and most of the ISPs tend to connect to Network Access Points (NAP), which provide backbone connectivity to the servers located there. Historically, in the case of PSTN circuit-switched international calls, there are different accounting rates for different countries based on international traffic volumes sent and received. Under the international settlement system, the operator(s) in the country that originates a call has traditionally made a compensatory payment to the operator(s) in the country that terminates the call. Payments are made when traffic in one direction is greater than the traffic in the reverse direction. The level of payment is based on bilaterally negotiated "accounting rates". The net settlement payment is usually made on the basis of excess traffic minutes, multiplied by half the accounting rate. Net settlement payments, primarily from developed countries, have grown larger as traffic flows have become less balanced.

Operators that send more traffic than they receive may have an incentive to develop alternative routing procedures. These operators may choose to route the traffic through alternate routes to avoid settlements based on the above cost accounting rates and instead pay interconnect fees at local access rates or below. This may be one reason as to why more and more traffic is being delivered using IP backbones instead of PSTN circuits to deliver traffic. Other reasons include the fact that IP networks allow operators to provide economically voice, data and other applications over a single, integrated network that offers capabilities for new revenue opportunities.

Price arbitrage opportunities have grown little in domestic long-distance or international calls between low-price markets.

III.4.4 Economic implications of USO on IP telephony

In many countries, the largest or dominant operator has an obligation to provide basic services directly to any customer who reasonably requests it, or to ensure that all citizens have access to certain services. In some cases, this is accompanied by specific obligations to extend network infrastructure to unserved areas. These requirements are sometimes referred to as universal service or universal access obligations. Because universal service/access obligations require provision of services to customers in areas which are uneconomic to serve, the universal service/access provider faces costs as a direct result of its obligation. In some jurisdictions (e.g. the USA), other operators contribute to the funding of these costs through competitively neutral funding arrangements. Elsewhere (e.g. Australia and the UK), costing studies have established that the costs of universal service provision are offset by benefits accruing to the universal service provider directly as a result of the universal service obligation – where this is the case, there is obviously no need for universal service funding arrangements.

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In Hong Kong, the operators of external telecommunication services have the obligation of sharing the cost of providing universal service of the domestic telephone network in accordance with the volume of traffic handled. This obligation is not dependent on the technology used. Thus operators of external telecommunication services based on the IP telephony technology are also required to pay their share of the universal service contribution (currently about 10 Hong Kong cents per minute). Furthermore, where the calls are delivered through the domestic telephone network, a Local Access Charge (currently about 13 Hong Kong cents per minute) is payable to cover the cost of transmission over the domestic network. Again, this is not technology dependent and IP telephony operators are subject to the same obligation as operators using other technologies.

The deployment of IP telephony has the potential to reduce the cost of universal/service access. If IP telephony is used extensively in jurisdictions where contributory funds operate, regulators may wish to review funding arrangements. In these cases, the cost of universal service/access obligations will need to be re-assessed to reflect changes to the cost base resulting from the deployment of IP telephony.

Certain universal service funding schemes may present the possibility of inequalities, where only some providers of functionally equivalent services are taxed in respect of those services while others are not, based solely on the technological platforms which they employ. Worse, it creates incentives for those operators that have traditionally paid into universal service schemes to switch to alternative platforms, reducing even more the volume of traffic on which payments are payable.

The universal service funding schemes of Uganda and Nepal offer a creative solution to the problem of this issue. In both the countries, ISPs are required to be licenced and to contribute a small portion (1-2 per cent) of their revenues to the universal service fund. These funds are intended to be allocated through competitive tendering. In this way, the possible cost advantage enjoyed by ISPs is somewhat lessened because universal service charges cannot be avoided.

III.4.5 Interconnection issues

To promote competition, based on a level playing field, it may be required to ensure equitable and non-discriminatory interconnection between service providers. Interconnection regulation involves unbundling of network elements, definition of technical interfaces, such as user network interface (UNI) and network-to-network interface (NNI). Technical regulation also involves specification of "Quality of Service" on each of the technical interfaces, such as UNI and NNI, so that end-to-end quality of service can be guaranteed to the customers in a multi-operator environment.

A change in technology from circuit-switched to IP-based networks may affect the cost of providing and running networks and may alter the basis on which the costs of some key interconnection functions such as call origination and call termination have been based. This may result in regulators needing not only to revise the figures that they determine but to revise the basis of their determinations to take into account the changes in the nature of the networks. For example, where element-based costing and charging are used, new elements may need to be defined and their costs elaborated.

Technology is steadily reducing the cost of networks whilst the costs of billing and customer service may be reducing more slowly. This is creating economic pressures for retail prices to move approximately in the following sequence:

- Distance dependent, charge by the minute
- Distance independent within the country, charge by the minute
- Distance independent internationally, charge by the minute
- Distance independent, charge per call
- No call charges, all charges by subscription

IP technology is increasing this pressure and flat-rate Internet access is part of this progression. Despite the wish of some regulators to avoid influencing the markets, interconnection rates may influence the ability of service providers to set retail prices. If the above trend is to proceed, then changes in retail prices may have to be reflected by changes in the interconnection rates, in particular the introduction of rates that are less dependent on, or independent of, call minutes.

The following table illustrates the basic differences between interconnect in the circuit-switched and Internet (packet-switched) networks:

Table 3 – Interconnect differences

Aspect of interconnect	Approach with circuit-switched interconnect	Approach with Internet interconnect (IP)
Core transit network	National incumbent in each country	United States backbone operators
Regulation of interconnect supply conditions	Strong regulation of the incumbent; e.g. incumbent must charge cost-based prices for unbundled services	None
Publication of interconnect charges	Required for interconnect with incumbent	None
Control of network boundaries	Gateways for charging and security functions	Very limited
Location of POIs	Agreed on bilateral basis	Increasingly located at Telecom Hotels ²⁰
Charging arrangements	Charge per minute	No charges, or charges based on BW of IC link
Principles underlying charging arrangements	Cost-causation	Cost and value of interconnect to each party
Is Call Detail Record provided?	Yes	No
Possible to distinguish traffic by origin?	Yes	No

The following points are worth noting:

- A high proportion of Internet content and capacity is generally provided by large backbone ISPs. ISPs with smaller networks may pay these backbone operators – directly or indirectly – for global connectivity. The backbone ISPs usually act as the core transit network through which a significant proportion of traffic may pass. In contrast, circuit-switched network interconnection is based on the network of the national incumbent. Small operators start by interconnecting with the incumbent and only interconnect directly with other small networks when the volume of traffic justifies the expense. In this case, the national incumbent acts as the transit network.
- Interconnect within the Internet is generally unregulated and non-transparent. Interconnect arrangements between ISPs are based entirely on commercial negotiation; the major ISPs also require interconnecting parties to sign non-disclosure agreements on the terms agreed. Interconnect between circuit-switched networks is very different. Interconnection to the fixed network incumbent has been strongly regulated. In particular, interconnect charges set by the dominant fixed incumbent are usually unbundled, cost-based and published.
- Interconnect within the Internet was initially designed to provide a single "best efforts" service for transporting packets from one IP address to another regardless of the network to which the

²⁰ A Telecom Hotel is a site where several communications services could house network equipment and link more easily to one another. Telecom Hotels house switches of several telecom companies in one place. Shared quarters save money and let the companies plug into each others' networks. Telecom Hotels provide access to regional, national and global telecommunication backbones, which are the primary means of carrying voices or data from one destination to another.

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host sites are attached. Packets pass across the boundary between IP networks from one router to another with minimal control. In contrast, circuit-switched operators build gateways at the boundaries of their networks, and use these to provide charging and security functions.

- Increasingly, interconnect within the Internet occurs at common Telecom Hotel (see footnote 20) facilities – whether the ISP is interconnecting on a bilateral basis or through an Internet exchange on a multinational basis. The sheer volume of traffic exchanged, and the rate at which this traffic is growing, may make the use of Telecom Hotels a viable option. In contrast, the physical points of interconnect (POI) between traditional circuit-switched networks are located at a large number of points, as agreed on a bilateral basis by the interconnecting operators.
- An originating circuit-switched network operator usually charges the user to make a call and then pays another fixed circuit-switched network operator to terminate it. This is done for each call and follows the principle of cost-causation (that is, each network pays for the link capacity required to deliver or receive calls generated as a result of its sales activities). When two networks interconnect within the Internet, they determine charging arrangements through commercial negotiations, taking account of both the cost and value that each creates for the other over the aggregate of traffic exchanged.
- Call detail records (CDRs) are created by circuit-switched operators on either side of the point of interconnect and are used to calculate and check interconnect charging between the operators. There are no CDRs for interconnect within the Internet. Usually, one ISP pays the other a flat rate per month for carrying packets, or peers with it on a settlement-free basis.
- Autonomous systems (networks) on the Internet cannot distinguish between traffic originating on their own network and traffic originating on another network that is being carried as transit traffic. Circuit-switched networks can distinguish transit traffic by looking at the calling number in the CDR. This means that a circuit-switched network can charge an originating network separately for transit traffic and terminating traffic. An IP network within the Internet cannot do this. IP Network A can treat IP Network B as a customer and charge it to deliver both traffic terminating on A's network and traffic transiting through A. Alternatively, it can peer with B and deliver B's traffic free of charge. However, if A agrees to a peering arrangement, it will not accept transit traffic from B, which must find another way of delivering this component of its traffic. Circuit-switched networks usually share the cost of interconnect links on a cost-causation basis. ISPs negotiate interconnect link charges. This generally means that small ISPs pay large ISPs for the full cost of the interconnect link.

The following table provides comparison between Interconnect charging in circuit-switched and packet-switched networks.

Table 4 – Interconnect charging: circuit-switched networks versus the Internet

Interconnect arrangement for charging	Circuit-switched networks	Internet (IP)
Terminating traffic	Charge per minute	No charge between IP networks of the same size Large ISPs charge small ISPs on the basis of the BW of the IC link
Transit traffic	Charge per minute	Charge on the basis of the BW of the IC link
Interconnect link	Cost sharing on Causation basis	Smaller IP networks normally bear the cost of the link in full

Source: OVUM (The business case for Next-generation IP Networks).

Interconnect charging arrangements:

Interconnect arrangements between major ISPs are commercially negotiated on a bilateral basis. There are no hard and fast rules. In virtually all the cases, the resulting arrangements are commercially confidential. The present scenario for the payment of an interconnect link is as follows:

- Each ISP usually bears its own costs when providing a link from its network to an Internet exchange for either bilateral or multilateral interconnect
- Small ISPs use point-to-point links and normally pay the full cost of these links.
- Cost sharing of point-to-point links between backbone ISPs is commercially negotiated and there are no formal rules.

Under current conventions, smaller ISPs bear the full cost of links to backbone ISPs. This service is known as "Transit" which provides full connectivity to the Internet. Interconnect is usually performed via an Internet exchange. This system was established in the early 1990s when non-US ISPs were interconnecting to backbone providers in the United States. But now many small ISPs claim that they should share the cost of the link with the backbone ISPs rather than pay the entire cost.

The main interconnect charging arrangements are as follows:

- Settlement-free peering
- Payment for global connectivity (payment for transit service)
- Settlement-based peering

Settlement-free peering²¹:

Major ISPs usually peer and exchange traffic with each other on a settlement-free basis. This arrangement may have the advantage that there is no need to measure and bill the traffic flowing across the point of interconnects. This may cut operating costs significantly. Whether one ISP will peer with another in this way is a matter for commercial negotiation. The general rule is that ISPs will use settlement-free peering if they are roughly equivalent in terms of the size and reach of their backbone networks, the QoS in terms of packet loss, etc., and their ability to interconnect with each other at a minimum of three or four POIs.

Paying for global connectivity:

An ISP might pay another ISP to provide it with global connectivity when it cannot peer on a settlement-free basis. For example:

- A small ISP that does not operate any significant IP network might use this method for delivering all its traffic.
- Large European ISPs might use settlement-free peering with other large European ISPs, but opt for a US-based backbone ISP to provide them with connectivity to destinations such as the Asia-Pacific region, Latin America and the United States.

The charge for global connectivity will depend upon the volume of traffic generated. An ISP will try to minimize the volume of traffic delivered in this way by reaching peering arrangements with other ISPs wherever possible.

Settlement-based peering:

ISPs exchange traffic and pay one another according to the net flow across the point of interconnects. Such arrangements are still relatively rare and many Internet exchanges only allow settlement-free peering. However, the London Internet Exchange recently changed its policy to allow settlement-based peering.

²¹ Peering: The Whatis.com site defines "peering" as the arrangement of traffic exchange between Internet service providers (ISPs). Larger ISPs with their own backbone networks agree to allow traffic from other large ISPs in exchange for traffic on their backbones. They also exchange traffic with smaller ISPs so that they can reach regional end points. Essentially, this is how a number of individual network owners put the Internet together. To do this, network owners and access providers (ISPs) work out agreements that describe the terms and conditions to which both are subject. Bilateral peering is an agreement between two parties. Multilateral peering is an agreement between more than two parties.

Chapter III.5 – Conclusions to Part III: Economic Aspects

- i) By and large, IP telephony promises to provide capability to offer converged and innovative telecommunication services to the end users in a cost-effective manner. Investment in IP-based networks may be regarded as an investment in the future, irrespective of the state of economic development of a particular ITU Member State. The business case for investment in IP would rarely be based on the potential of IP telephony alone, but rather on the wider potential of IP-based networks to carry data, text and video traffic as well as voice. Almost all the carriers across the globe are planning various strategies to gear up to facing the challenge posed by packet-switched telephony such as IP telephony.
- ii) Net settlement payments have been declining worldwide since the mid-1990s, and arguably this would have happened even without IP telephony. This trend is principally the result of increased competition and pressure from countries that make net settlements. As retail prices fall and more traffic is routed via least cost routes, settlement rates are forced downwards. This market change is particularly affecting those PTOs that have traditionally relied upon revenues from international service to cross-subsidize their local access networks. It is forcing the pace of tariff rebalancing.
- iii) Operators have traditionally used profitable long-distance and international services to cross-subsidize in part the functions of network access and local calling. In increasingly competitive markets, such hidden cross-subsidies can no longer be sustained. In the future, operators will need, instead, to address new challenges that may require substantial tariff rebalancing and a greater reliance on locally generated revenues.
- iv) While IP telephony SPs may bypass certain parts of an incumbent operator's network, they will not eliminate the need for local networks. Indeed, insofar as Internet telephony is a new "killer application" and makes access to the Internet even more popular, it may actually increase the volume of local calls. Already, in some Member States, as much as a third of all local calls are to the Internet, though IP telephony represents only a small proportion of this demand. Furthermore, dial-up Internet access is on a steeply rising curve while circuit-switched traffic is slowing down. Competition will drive prices closer to costs and, where IP telephony offers the lowest cost alternative, it may be the preferred solution.
- v) For ITU Sector Members that are equipment vendors, the development of new IP-based product lines may be essential to future growth and profitability. Third-generation mobile networks (IMT-2000), which can also be IP-based, offer vendors additional opportunities to propose new products including customized and personalized location-based information services that will most likely resemble the Internet client/server model rather than the traditional telecommunication model.

PART IV

IP TELEPHONY – POLICY ASPECTS

Chapter IV.1 – Review of the current regulatory framework

IV.1.1 General remarks

The introduction and growth of IP telephony raises a number of important policy issues. ITU-D is challenged with advising and assisting Member States and Sector Members in response to specific concerns and needs of developing countries regarding the policy implications that surround the introduction of "IP telephony".²² In Part IV, expert advice and assistance is provided in three major sections:

- review of current regulatory frameworks;
- country case studies; and
- shared experiences in developing new methods and approaches for the introduction of IP telephony.

This part of the report is meant to serve as a general guide – not a step-by-step plan. The Secretary-General's Report to and the Chairman's Report of the third World Telecommunication Policy Forum (WTPF-01), <http://www.itu.int/osg/spu/wtpf/wtpf2001/index.html>, provide useful background discussions of the many policy issues, as well as a survey of the varied domestic regulatory policy approaches of ITU Member States. The survey reveals that there is no single policy approach, and indicates that the policy issues will continue to evolve as IP telephony technology is enhanced and more widely deployed.

Aspects of the Secretary-General's Report and the WTPF Chairman's Report have been included in this report where applicable. Readers are encouraged to consult the full reports as well as the source documents prepared for the WTPF for additional detail. However, as the underlying technologies and markets evolve, it is important to consider the effects of these changes on policies and to plan for change within the policy-making process.

IV.1.2 Overview

As IP networks and IP telephony become more widespread, policy-makers face the challenge of evaluating whether current regulatory frameworks, developed initially for circuit-based networks, are relevant and appropriate for IP-based networks. This challenge is arising at a time when many Member States are lightening their regulatory regimes and moving to greater reliance on competition to ensure consumers the broadest possible access to telecommunication services.

Due to the very different regulatory regimes created to address particular domestic economic, political and infrastructure challenges, ITU Member States may want to focus their reviews on the rationale behind their policy frameworks, and especially the desired effects in the context of overall economic and social development. In particular, the existing level of network development and state of the communications market generally are issues that most likely will have to be taken into account. Countries that have very low teledensity levels must address the most basic difficulty of building a telecommunication infrastructure.

Within these broad policy frameworks, IP telephony may raise a number of specific questions for policy-makers and regulators that require a careful and informed balancing of different and sometimes competing interests. As a threshold matter, it is useful to understand the short- and long-term economic consequences of any policy decision. It also is essential for regulators and policy-makers to understand that there is no policy model that is universally applicable. A number of approaches may be appropriate.

²² Opinion D <http://www.itu.int/ITU-D/e-strategy/internet/iptelephony/Documents/wtpf2001/Chaireport.html#OPINIOND>.

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It is recommended that ITU Member States consider the benefits of:

- 1) first, defining the broad telecommunication policy objectives for the country, within the context of overall economic development and social needs, and
- 2) second, tailoring the regulatory regime to reach these objectives.

IV.1.3 Areas for review

As the basis for determining policies specific to IP telephony, the Group of Experts believes that ITU Member States may benefit from a review of their more general domestic telecommunication regulatory frameworks with the following in mind:

IV.1.3.1 Achieving policy goals in the context of convergence and existing market conditions

Member States may need to evaluate their policy goals before determining what, if any, regulation is necessary in a converged market. For example, it may be appropriate to limit regulation in a converged, competitive marketplace, employing regulation only when there is market failure.

IV.1.3.2 Encouraging investment, spurring innovation, advancing development and opening up markets

A competitive telecommunication environment allows for competition among multiple service providers and for multiple investors. Experience around the world reveals that competitive telecommunication models have been adopted to attract capital investment for telecommunication and IP-based network infrastructure build-out. It also is evident that policy-makers and regulators have successfully implemented a competitive model by ensuring appropriate safeguards against undue market power. Policies that allow multiple carriers and Internet Service Providers (ISPs) have been shown to stimulate infrastructure build-out and lower prices for business and consumer access.

IV.1.3.3 Customer benefits

Competition has been shown to enhance end-customer choice by providing more options, both in terms of price and quality. Consumer welfare is usually the greatest in an environment where there are no limits on the number of suppliers and services.

IV.1.3.4 Universal service/access objectives for telecommunication services

In some circumstances, the market may not function to provide telecommunication services to certain subsets of users. Universal access can be defined as government-sponsored programmes designed to provide access to specified telecommunication services for a community. Several countries in the developing world have adopted universal access models to provide access to a defined set of telecommunication services in rural and remote areas, and for low-income individuals. Without access to these services, Internet and other advanced IP services access and use is inhibited. Countries that implement universal access programmes may want to consider the following:

- A universal access programme that is created to promote the development of telecommunication infrastructure in rural and remote areas, and for low-income individuals.
- A universal access programme for telecommunications that is operated in a transparent, competitively neutral and non-discriminatory manner.
- Clear identification of the universal service requirements and provider obligations.
- When universal access to local services is funded by a cross-subsidy (for example, from long-distance telecommunications), clear and transparent identification of that cross-subsidy.
- A funding mechanism that is clear as to whether the funds come from taxes or revenues.
- A universal access plan that promotes infrastructure development and is pro-competitive.

IV.1.3.5 Consideration of technological issues such as quality of service

The service and quality capabilities of IP telephony technologies are still evolving. To reach the full range of market needs, it is expected that IP technology will have functional capabilities similar to circuit-switched technology. Policies that allow flexibility in choice of technology and application to address user needs and to permit users to choose among different prices and qualities are more likely to encourage investment and stimulate development.

IV.1.3.6 Interconnection and access policies

Within the context of network transition, one role of policy-makers may be to ensure that existing services remain available as new services are introduced, as driven by market forces. Interconnection policy can play a critical role by ensuring that new and existing infrastructure can coexist, thus preserving and enhancing the value of both. In technology transitions, such as the one from circuit-switched to packet-switched communication transport mechanisms, there is usually a period of coexisting technologies. Policies that recognize transition by allowing multiple network platforms and encouraging their interconnection are preferable.

IV.1.4 Agency contacts

ITU maintains a contact database of the regulatory agencies and key contacts by Member State. Contact information can be found at <http://www.itu.int/GlobalDirectory/index.html>. An additional source of country-by-country contact information is available online at: <http://www.totaltele.com/links/list.asp?CategoryID=267>.

Chapter IV.2 – Case studies and sharing experience

IV.2.1 Introduction

While some developing countries have policies prohibiting IP telephony, others have policies embracing it. Some do not regulate IP telephony at all, while others have chosen to include it in a positive manner within their telecommunication regulatory framework. These countries may be motivated by a desire to encourage and stimulate emerging technologies that can lower costs, increase total revenue opportunities and promote innovation and national economic growth. These policies may be linked to concerns about imposing regulations on technologies that are not fully mature. Limitations placed on IP telephony may be seen as inconsistent with approaches designed to stimulate the deployment and migration to IP-based networks. Regulators also may be hesitant to intervene in new markets unless there is evidence of market failure. Decisions to prohibit, regulate or not regulate IP telephony are often coupled with long-term policy objectives for the development of the communication infrastructure/network.

IV.2.2 Results of policies embracing IP telephony

The World Bank has created an Economic Internet Toolkit for African Policy Makers addressing many of the above issues in the context of developing economies. This toolkit presents a model of the likely impacts of the Internet on African telecommunication companies and Internet service provider revenues, models of the cost structure and potential reach of Internet service, data on the extent of Internet development in Africa and examples of its current use. With this background, the toolkit goes on to discuss policy choices faced by countries that hope to expand Internet use within the context of needed telecommunication reform and government/private partnerships involving universities and NGOs. It is available in five pdf. files and an Excel spreadsheet containing the model itself. It can be accessed online at: <http://www.infodev.org/projects/internet/010toolkit/afprelim.pdf>.

IV.2.3 Policies consistent with transition/convergence of networks

Case studies can provide useful insight into the impact of regulation on the development and expansion of the telecommunication market within a particular economy. Caution should be exercised, however, in extrapolating the findings to economies that do not share the basic characteristics of the studied economy. However, the methodologies used in these studies can be particularly useful to others conducting their own case study. ITU has completed case studies of five Member States: Canada, China, Colombia, Korea and Peru. These studies are available online through the ITU website at: <http://www.itu.int/osg/spu/wtpf/wtpf2001/casestudies/index.htm>.

Over the past four years, the Organization for Economic Co-operation and Development (OECD) has undertaken in-depth reviews of the telecommunication regulations in a number of economies. This work is aimed at producing, for each country reviewed, a multidisciplinary review of progress on regulatory reform based on international benchmarking, self-assessment and peer review. The recent reviews of the Czech Republic, Hungary and Poland are specially noteworthy. The reviews on Hungary (<http://oecdpublications.gfi-nb.com/cgi-bin/oecdbookshop.storefront>) and the Czech Republic (<http://oecdpublications.gfi-nb.com/cgi-bin/oecdbookshop.storefront>) are available through OECD publications (as of 2001).

IV.2.4 Sharing experience in developing new methodologies and approaches

IV.2.4.1 General remarks

Countries have taken widely differing policy approaches toward IP telephony, which may be related to different prevailing market conditions or degrees of liberalization. The sharing of these different approaches can help policy-makers define and evaluate options to address issues specific to the environment in their country.

IV.2.4.2 Approaches to "technology-neutral", sector-specific regulation

Technological neutrality is a principle that is invoked by some policy-makers and regulators when addressing IP telephony and other emerging communication technologies. This concept can be generally characterized as an effort to apply regulations in an identical manner to like services, regardless of the technology used to provide these services in a competitive market. Unless other policy imperatives take precedence, the purpose of this concept is to support competition policy by ensuring that one provider is not given more favourable regulatory treatment than another when providing equivalent services. Others believe that policy-makers should not be indifferent to technology. They assert that emerging technologies might benefit from a "window", i.e., a form of asymmetric regulation during an introductory phase, which would allow them to develop and grow outside traditional regulation.

The principle of technological neutrality was widely discussed at the WTPF meetings and the Group of Experts meetings. No consensus was reached. However, many believe that:

- 1) a country first must have effective competition in order to apply a principle like technological neutrality;
- 2) technological neutrality is a legitimate consideration in policy and regulatory deliberations, but it should not override broader pro-competitive objectives.

The European Union (EU) has concluded proceedings to create technology neutral regulation. The Directive on access to, and interconnection of, electronic communications networks (COM(00)384final of 12 July 2000) harmonizes the way in which EU Member States regulate the market between suppliers of communication networks and services in the Union. The Directive lays down a framework of rules that are technologically neutral, but which may be applied to specific product or service markets in particular geographical areas, to address identified market problems between suppliers of access and interconnection.

A summary of the Directive "Interconnection and access in the new EU regulatory framework for electronic communications services" is attached as Annex P. Additional material is available at the EU website: http://www.europa.eu.int/information_society/topics/telecoms/regulatory/new_rf/index_en.htm.

IV.2.4.3 Application of domestic telecommunication regulation establishing effective competition, universal service/access obligations including any other further obligations and other experiences

- a) The United Kingdom's independent regulator, OFTEL, has had extensive experience with different regulatory approaches, including price caps, and of adjusting the scope and intensity of regulation to take into account the level of competition in the market and technological change. Its website is: <http://www.oftel.gov.uk/>.
- b) In the United States, there has been extensive competition in certain sectors of the telecommunications market since the 1970s, especially in long-distance and enhanced or value-added services. In 1996, national legislation was adopted that specifically opened the local telecommunication market to competition. An overview of the U.S. experience is provided in: <http://www.itu.int/ITU-D/e-strategy/internet/iptelephony/Seminars/2ndEGM/documents/policy/IPTel-21.pdf>.
- c) India is experimenting with IP telephony in limited applications. Under the NTP 1999, "Internet Telephony" is not yet permitted in India. The government of India has committed to monitor the development of IP telephony and its impact on national development and will review the issue at an appropriate time. The Government, at present, is working on various issues relating to IP telephony. Meanwhile, the incumbent operator BSNL has plans to use IP technology for real-time service for transit traffic between tandem exchanges, bypassing the tandems on an

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experimental basis at six locations in the country using a separate IP-based network. At present, it is not envisaged to connect these links to the Internet. Only domestic long-distance traffic is proposed for experimentation with VoIP, and no international direct dial calls are proposed in this experiment. This experiment will cover voice real-time and fax services, whereas data services will be transacted through the Internet.

IV.2.5 Potential policy implications of ENUM

Study Group 2 of ITU's Telecommunication Standardization Sector is currently addressing principles and procedures for the administration of electronic numbering (ENUM) as well as defining a framework for the possible role of ITU. A key issue is implementing and maintaining the databases necessary for translation of the E.164 numbers into the Domain Name System so as to safeguard the integrity of the E.164 numbering system. National regulatory authorities or policy-makers may wish to consider their appropriate level of involvement in these activities at ITU. More information on the ENUM protocol, and the issues related to it, can be found at www.itu.int/osg/spu/infocom/enum/ and in Annex H on ENUM.

IV.2.6 Training and education for regulators and operators

A workshop programme specific to IP telephony was developed by ITU working with the Group of Experts, ITU regional offices and certain Member States. The programme for these workshops, together with additional material, is included in Part V.

A number of institutions provide training for regulators and operators. Well known programmes include the following:

- The ITU *Internet Training Centres Initiative for Developing Countries (ITCI-DC)* is a multimillion-dollar project aimed at closing the gap in Internet and "new economy" skills in developing countries. Under this initiative, ITU plans to establish training centres to provide skills in Internet Protocol (IP) networking and services in existing non-profit institutions in developing countries. It is expected that the centres will also function as incubators to help small and medium-sized enterprises to develop Internet-related services. For further information, please contact: Mr Manuel Zaragoza, Project Coordinator, ITU Telecommunication Development Bureau (BDT), Tel.: +41 22 730 5428, e-mail: manuel.zaragoza@itu.int.
- The United States Telecommunications Training Institute (USTTI) (<http://ustti.org>) is a non-profit joint venture between leading US-based communication and IT corporations and leaders of the Federal government who together provide tuition-free management, policy and technical training for talented professionals from the developing world.
- The Institute for Public Utilities at Michigan State University (<http://www.bus.msu.edu/ipu/frmain.htm>) provides an annual two-week workshop "Camp NARUC" – exclusive to public sector employees and government officials that is endorsed by the National Association of Regulatory Commissions (NARUC). The programme includes a track specific for regulators from outside the United States. Camp NARUC International is designed for officials from developing countries who are responsible for establishing regulatory institutions, implementing policy and designing regulation for infrastructure sectors. The programme is designed to complement the traditional Camp NARUC programme. More information is available at: <http://www.bus.msu.edu/ipu/confsem.htm>.
- The United Nations Institute for Training and Research Project for Law and Cyberspace provides subregional workshops and training courses on "Information Society Regulation".
- There is a European and Regulation Framework Masters Course at the Ecole nationale supérieure des telecommunications. The contact person is annie.blandin@enst-bretagne.fr.

Manufacturers and operators also provide training. Cisco Systems has established the Cisco Network Academy Programme (CNAP) that is operational in more than 130 countries and more than 8,000 locations. The Networking Academy curriculum centres on teaching students to design, build, and maintain computer networks. The programme prepares students for the twenty-first century workplace, while serving as a valuable model for e-learning. More information is available at: <http://www.cisco.com/warp/public/779/edu/academy/>.

ITU has signed a cooperation agreement with Cable & Wireless (C&W), through the C&W Virtual Academy, to provide training in communication management to telecommunication professionals from less developed countries within the framework of the ITU's Global Telecommunications University. C&W will provide scholarships to candidates. Training will include themes such as "regulation and policy in communication" and "introduction to IP technology for business". More information is available through the Telecommunication Development Bureau (BDT) at ITU from barbara.wilson@itu.int.

Alcatel, Nortel and Siemens have signed partnership agreements with the ITU-D Centres of Excellence Programme.

Chapter IV.3 – Conclusions to Part IV: Policy Aspects

The policy implications of IP telephony should be examined within the context and complexity of the changes in the market environment. Developing countries face the additional challenge of addressing relatively low teledensity levels. As IP networks and IP telephony become more widespread, policy-makers may face the challenge of evaluating whether current policy frameworks, developed initially for circuit-based networks, are relevant and appropriate for IP-based networks. As the basis for determining policies specific to IP telephony, ITU Member States may benefit from a review of their more general domestic telecommunication regulatory frameworks with the following in mind:

- ITU Member States may need to evaluate their broader policy goals before determining what, if any, regulation is necessary in a converged market.
- Experience around the world reveals that competitive telecommunication models have been adopted to successfully attract capital investment for telecommunication and IP-based network infrastructure build-out.
- Customer benefits are usually the greatest in an environment where there are no limits on the number of suppliers and services.
- In some circumstances where the market may not function to provide telecommunication services to a certain subset of users, government-sponsored universal access/service programmes may be helpful.
- Policies that allow flexibility in the choice of technology and its application to address user needs and to permit users to choose among different prices and qualities are more likely to encourage investment and stimulate development.
- Consider, in competitive markets, whether to take a technology-neutral approach by applying regulations in an identical manner to like services, regardless of the technology used to provide these services.
- Policies that allow for the coexistence of multiple network technology platforms and encourage their interconnection are preferable.

Countries have taken widely differing policy approaches toward IP telephony, which may be related to different prevailing market conditions or degrees of liberalization. No policy model is universally applicable. A number of approaches may be appropriate. The sharing of these different approaches can help policy-makers define and evaluate options to address issues specific to their country.

Training for policy-makers, regulators and operators is essential to help understand the implications of new technologies, new market structures and alternate regulatory models. A number of institutions, organizations and companies provide training programmes. ITU Members are encouraged to take advantages of these programmes. ITU Members are also encouraged to contact each other and to share first hand their experiences.

PART V

IP TELEPHONY – WORKSHOPS AND TRAINING ASPECTS

Chapitre V.1 – General considerations

V.1.1 Introduction

In addition to its tasks, the first meeting of the Group of Experts on IP Telephony (Opinion D, part 3) identified the following three topics in order for BDT to implement actions related to Opinion B:

Topic 1: IP networks and the introduction of IP telephony

Objectives

- i) To inform decision-makers and raise their awareness with regard to essential issues.
- ii) To outline a proposed procedure for the implementation of IP networks.

Topic 2: Training workshops for technical personnel

Objectives

- i) To train technical personnel in the design of IP networks.
- ii) To provide modern tools for dimensioning IP networks.
- iii) To put in place a policy for capacity building and the development of human resources.

Topic 3: Training workshops on regulatory matters associated with the introduction of IP telephony

Objectives

To create the necessary competences to lay the foundations for implementing IP networks and ensure that they are operated in optimum fashion.

Chapter V.2 – Seminars and workshops held

V.2.1 General remarks

After the World Telecommunication Policy Forum on "IP telephony" and in response to Opinion B of the Forum, regional or subregional workshops were carried out by BDT. The following workshops were organized:

- 1) Bamako, Mali (18 to 20 April 2001): West Africa subregional workshop on "IP telephony" focused on technical issues and ITU-T Rec. H.323.
- 2) Dakar, Senegal (26 to 30 June 2001): West Africa subregional seminar on IP telephony focused on technical issues, ITU-T Rec. H.323 and policy issues.
- 3) Lima, Peru (18 to 21 July 2001): Latin America subregional seminar on "Internet and IP telephony" focused on technical issues, ITU-T Rec. H.323, IP projects, and policy and economic issues.
- 4) Dakar, Senegal (23 to 25 July 2001): Forum on telecommunication development in Africa. Results: An action plan for the École supérieure multinationale des télécommunications [Multinational Higher Institute for Telecommunications] (ESMT), for setting up training programmes on new technologies in general and IP telephony in particular (the ESMT in Dakar is an ITU Centre of Excellence).
- 5) Brisbane, Australia (10 to 13 October 2001): ITU regional seminar in partnership with the Pacific Islands Telecommunications Association. The purpose of this seminar, on the Internet and IP telephony, was to define regional needs and propose the general outlines and subjects of training programmes for IP network administrators.
- 6) Damascus, Syria (7 to 10 January 2002): Regional workshop on IP telephony for the Arab Region, with the followings recommendations. Note the emphasis the participants put on the third point.

V.2.2 Arab regional workshop: Recommendations

- To call upon all Arab telecommunications administrations and their staff to deal with IP telephony in order to assimilate and benefit from it, and from the value-added applications it provides to the national economy and at the individual level to the Arab end user.
- To call upon them to take full advantage of what the Information Technology provides in general, to use them in the various applications of e-learning, including distance learning, e-commerce, e-banking, e-business and telemedicine, in addition to all modern and new applications in the various fields.
- To encourage the Arab administrations to carry out live trials as soon as possible in the field of IP telephony at the national, regional and international levels, taking into consideration the particular situation of each administration, and to exchange expertise among the Arab administrations in this field and to also benefit from the BDT experience.
- To call upon the BDT Regional Office for Arab States and the Arab Centre of Excellence to follow up activities in this domain and treat it as one of work priorities in the region; and to convene a future workshop to analyse the trials performed at the Arab region level; and to consider the possibility of setting up an Arab network for IP telephony in cooperation with BDT, taking into consideration the results of studying the report of the Group of Experts to the World Telecommunication Development Conference (WTDC-02).
- To encourage administrations to benefit from training courses to be organized by BDT in the fields of policy, regulatory and technical issues of IP telephony.

- To call upon Arab administrations to benefit from all initiatives provided to Arab and non-Arab Sector Members in the field of studies and training, as well as to benefit from the supplementary activities of BDT in the field of E-Strategies, such as:
 - Internet Training Centres
 - E-commerce project
 - E-government project
- To call upon these administrations to benefit also from the direct assistance of BDT in the field of training and trainers for the IP telephony technologies.

Chapter V.3 – Strategies for developing an IP telephony training policy

V.3.1 Organization of workshops

Taking as a basis the objectives of the three topics identified for implementation of Opinion B, the following outlines and subjects, within the framework of the ITU-BDT human resources development initiative (e.g. Centres of Excellence), are proposed:

V.3.1.1 Topic 1: IP networks and the introduction of IP telephony

Outline:

- i) Presentation of views of WTPF-01
- ii) IP network opportunities
- iii) Challenges:
 - 1) Technical challenges
 - 2) Economic challenges
 - 3) Regulatory challenges
 - 4) Human-resource challenges

Subjects:

- i) Information on progress being made in the area of IP networks and IP telephony within ITU-BDT
- ii) IP network opportunities
- iii) Technical challenges and opportunities
- iv) Economic challenges
- v) Regulatory aspects
- vi) Development of human resources
- vii) Case studies, and examination of actual country experiences

V.3.1.2 Topic 2: Training workshops for technical personnel

Outline:

- i) Traffic engineering and dimensioning
- ii) Architecture
- iii) Interconnection
- iv) Migration from switched telephone networks to IP networks
- v) Protocols
- vi) Security
- vii) Management tools

Subjects:

- i) IP networks: Architecture
- ii) IP networks: Protocols
 - Presentations by equipment suppliers (ITU-D)
- iii) IP networks, with quality of service (MPLS, IntServ, DiffServ, RSVP)
 - Example of IP network with quality of service: IP network/MPLS
 - IP/ATM networks with quality of service
- iv) Voice telephony over IP networks: Challenges, opportunities and risks
 - Voice telephony over IP networks: H.323 network architecture

- Voice telephony over IP networks: Speech coding
- Voice telephony over IP networks: Video coding
- Presentations by equipment suppliers (ITU-D)
- v) Voice telephony over IP networks: Signalling and control protocols
 - H.225/RAS and H.225/Q.931 signalling protocols (with demonstrations)
 - SS7 Signalling
- vi) Voice telephony over IP networks: H.245 applications control protocol
 - Presentations by equipment suppliers (ITU-D)
- vii) Voice telephony over IP networks: Session initiation protocol (SIP) concepts and H.248
- viii) New access technologies: DSL, CATV modem, power line communication (PLC), UMTS, etc.
 - Presentations and demonstrations by equipment suppliers (ITU-D)

V.3.1.3 Topic 3: Training workshops on regulatory matters associated with the introduction of IP telephony

Outline:

- i) Standardization
- ii) Quality of service
- iii) Interconnection

Subjects:

- i) Overview of IP telephony
- ii) Quality-of-service standards
- iii) Interconnection
- iv) Universal service or access
- v) Access strategies (numbers of subscribers, infrastructure deployment)
- vi) Case studies, and examination of actual country experiences

V.3.2 Developing an IP telephony training policy

i) Reinvigorating existing training structures

The five regional Centres of Excellence that have already foreseen establishing a programme on IP telephony or that are in a position to integrate such a programme are:

- ESMT (Dakar)-CoE-AFR
- AFRATI (Nairobi)
- CoE-ARB (Damas)
- CoE-ASP (Bangkok)
- CoE-AMS (Argentina)

ii) Encouraging the establishment of training centres in developing countries

Example:

The following Centres of Excellence were chosen within the framework of ITU-BDT's Internet Training Centres Initiative for Developing Countries (ITCI-DC):

- Universidad Nacional Autónoma de Honduras [National Autonomous University of Honduras] (UNAH)
- National University of Samoa (NUS)
- Ecole supérieure des télécommunications [Higher Institute of Telecommunications], in Tunisia

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- Ecole supérieure multinationale des télécommunications [Multinational Higher Institute for Telecommunications] (ESMT), in Dakar
- Escuela Superior Politécnica del Litoral [Higher Polytechnical Institute of the Coast Region], in Ecuador

iii) Fostering and developing partnerships with administrations, equipment suppliers and operators through projects initiated by ITU-BDT

Example:

- ITU-BDT initiative (the ITCI-DC project)
- Cisco training programme (under the Cisco Networking Academy Programme) in 2001
- Oracle training programme (under the Oracle Internet Academy) in 2002

iv) Seeking out partners able to sustain the occupational training programme

- Nortel and ESMT in Dakar

Chapter V.4 – Conclusions to Part V: Workshops and Training Aspects

The workshops held thus far have been very much appreciated by participants, with particular reference to the challenges and opportunities that IP technologies provide to elaborate new multimedia applications and particularly IP telephony.

The regional workshops should be a forum for the participants to debate the following points:

- the technological requirements for IP telephony;
- how quality of service can be ensured;
- how existing networks can be upgraded;
- how experience acquired in optimizing the methods and solutions in establishing IP networks can be shared;
- replies to questions on cost structures, a mechanism by which prices are set, interconnection, numbering, market, etc.

The partnership role of ITU-BDT together with administrations and the private sector (operators and equipment suppliers) was very important in ensuring the success of the workshops and case studies.

Annex A – Internet Protocol (IP) and User Datagram Protocol (UDP)

A.1 IP (Internet protocol)

The purpose of the IP protocol is to route information across a set of interconnected networks. This is achieved by transferring datagrams (data packets) from one module to another, until they reach their destination. The modules are programmes executed in network servers and routers. Datagrams are transferred from one module to another on a network segment according to the interpretation of an address. Thus, one of the key mechanisms of the IP protocol is address management. The IP protocol forms part of layer 3 of the OSI model. It is completely independent from the underlying layers, with the result that it can be adapted both to a local network and to a global network, which use many and varied media. It is a simple protocol, with no error control.

The most well-known network using the IP protocol is the Internet, which is a universal network comprising a multitude of terminals interconnected by a redundant mesh network. Data transfer over the Internet is carried out on a "best effort" basis. In other words, when two terminals are communicating, data control is carried out by the receiver alone. If an item of data received is thought to be incorrect, therefore, the information has to be re-sent. The communication is thus termed "end-to-end", since two terminals communicating over the network only realize there is an error when the data item has arrived at the other end. It is an asynchronous network, with the sole objective of transmitting a packet to its recipient without any other constraints.

The header of the IP datagram contains a "Type of Service" field, which serves to guide the choice of services when a datagram is transmitting across a network. The priority bits enable the packets to be placed in a hierarchical order, while the Data Transfer Rate (DTR) bits serve to customize the routing according to the desired service. Some networks offer a priority facility, whereby different types of traffic are given preference over others, although in general this comes down simply to accepting and carrying high-priority packets in the event of temporary overload. However, this field is still little used by network modules. In the main, the choice offered is a trade-off between three constraints: small delay, low error ratio and high bit rate.

A.2 UDP (user datagram protocol)

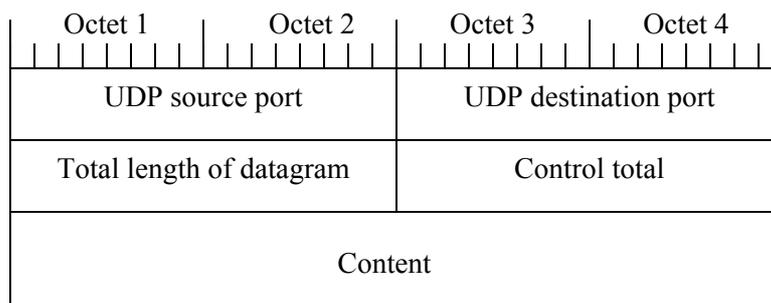
The level 3 IP network protocol is not reliable. It is the higher-layer transport protocol that has to control the transmission. In the Internet, this role is carried out by the transmission control protocol (TCP), which is a reliable protocol that corrects the errors of the underlying protocol. The TCP protocol header contains the sequence number of each packet. It serves to put the data streams back in order at the receiving terminal. Acknowledgements are sent to the source as soon as the packets are received. An unacknowledged packet is re-sent. Practice shows that the recovery of missing packets at least triples transit time. The repeated loss of a single packet can lead to significant time-lags. Since audio and video applications involve constant throughputs that cannot tolerate variations and fluctuations without causing interruptions, the TCP protocol is unsuitable for this type of application beyond a 4 or 5 per cent packet loss rate.

The approach selected for this type of application is to favour continuity over reliability, in other words tolerate packet losses by abandoning the packets in order to safeguard the continuous flow. It is thus the UDP protocol which is generally used for Internet telephony rather than the TCP protocol. The UDP protocol operates in unconnected mode, i.e. by sending datagrams processed independently by the network. The datagrams can take different routes and be received in a different order.

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The UDP protocol is a protocol without error correction (and thus unreliable) and its main function consists in differentiating between the different application services by routing them to the appropriate reception software processing module. This routing is achieved by allocating a port number to each application. The figure below shows the format of the UDP datagram header.

Figure A.1 – An 8-octet UDP datagram header



The UDP protocol is typically used as the underlying protocol for the RTP (real-time transport protocol).

Annex B – Quality of Service for Voice over Internet Protocol (VoIP)

B.1 Loss

The loss of a packet results in missing information when the audio signal is received. According to the number of packets lost, the sound quality at the receiving end may suffer. In the IP approach, packet loss forms an integral part of the concept: routers are obliged (with the random early detection algorithm) to destroy packets in order to avoid possible congestion.

There are four main causes of packet loss:

- Exhausted lifetime (TTL = 0)
- Receiving end delay greater than the jitter buffer
- Destruction by a congested module
- Packet invalidity due to transmission faults

The UDP protocol is used to transmit voice over IP for the advantages of using less overhead and reliance on higher layer protocols (like RTCP/RTP) to provide error or flow control or where "real-time needs" make retransmission – as used by the TCP protocol – inappropriate.

The rate of packet loss will depend on the quality of the lines used and on network dimensioning. If the speech quality is to be acceptable, the rate of packet loss must be less than 20 per cent.

One possible solution for reducing packet loss is to implement error correction systems using redundant and adaptive coding, i.e. variable according to the packet losses statistically observed within the network at any given point in time. Using such systems, it is possible to achieve very high levels of sound quality, even over the Internet. This solution does, however, give rise to an additional difficulty associated with the total transmission delay, which, as discussed earlier, has to be mastered if the network is to be used for telephony.

B.2 Delay

The delay is the time expressed in milliseconds passes between the transmission of speech and its reconstitution at the receiving end. If there is to be an interactive exchange, delay constraints must be applied to the speech transmission. The following figures (drawn from ITU-T Recommendation G.114) are given to indicate the classes of quality and interactivity according to the transmission delay in a telephone conversation.

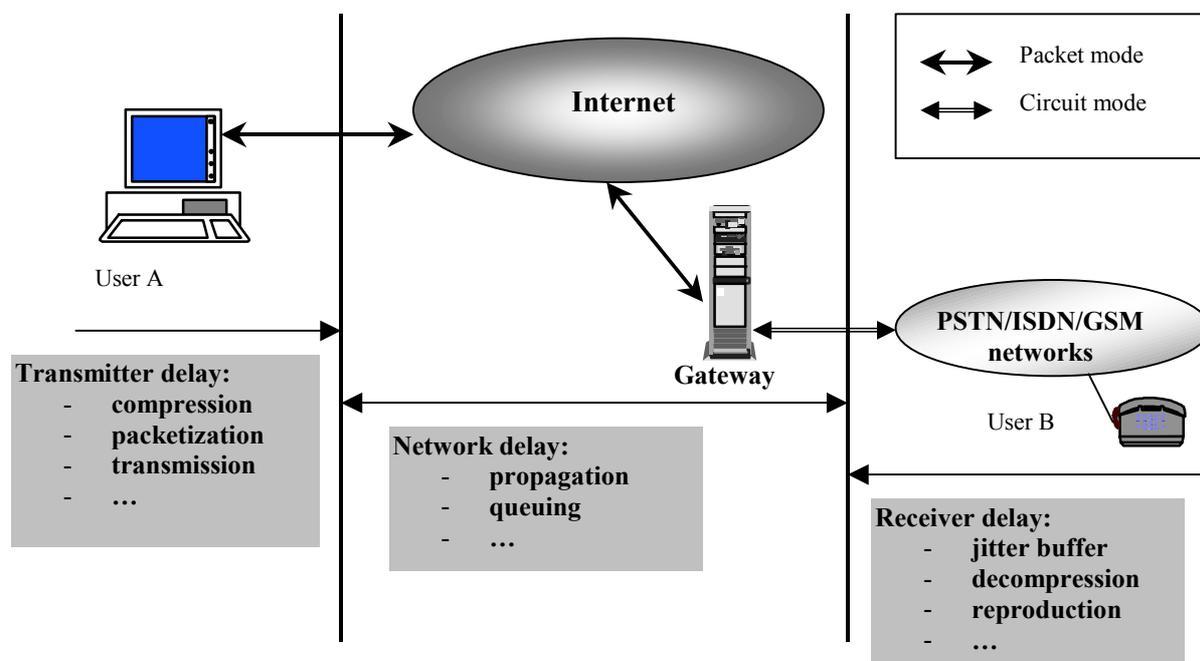
Table B.1 – ITU-T classes of quality according to transmission delay

Class No.	Delay by direction	Comments
1	0 to 150 ms	Acceptable for most conversations; only some highly interactive tasks may experience degradation.
2	150 to 300 ms	Acceptable for low-interactivity calls (satellite 250 ms per hop).
3	300 to 700 ms	Practically a half-duplex call.
4	Above 700 ms	Unusable unless the callers are well-versed in the art of half-duplex conversation (as used in the military).

The delay introduced by the Internet (from 50 ms to over 500 ms, according to the state of the network) is of a much higher order than the delay to be found in a conventional telephone network. To quantify the transmission delay over the network with any reliability is virtually impossible, in view of the great number of unknown factors (routing table, congestion, failures, queuing, ...). However, for

the route that would be taken by a voice transmission, it is possible to detail certain types of delay that are inherent to the network, as illustrated in the following figure depicting the PC-to-phone via Internet scenario (the IP network of the gateway service provider being considered as "ideal" and as not contributing significantly to the overall transmission delay).

Figure B.1 – IP telephony transmission delays



Transmitter delay

At the transmission end, the speech is coded and compressed prior to being encapsulated in IP packets. The size of the packet represents a compromise between the need to reduce the transmission delay and optimization of the bandwidth. The components of transmitter delay are:

- Digitization and coding, i.e. the time taken by a sound card or gateway to digitize and code an analogue signal.
- Compression, which breaks down into three parts:
 - Frame delay: unlike signal digitization, which is carried out in a continuous manner, compression relates to a specific data length. Waiting for that information can involve an appreciable processing time.
 - Coding delay: this delay, which involves compression by synthesis based on prediction, is required by the encoder in order to know, while it is functioning, how the signal is evolving.
 - Processing delay: time taken by the algorithm to compress a frame. This depends on the processor and on the algorithm used.
- Packetization: period of time during which the application puts together a packet (creation of header and insertion of data).
- Transmission: this period of time will depend on the configuration used, i.e. whether the connection is by modem or by direct access on a LAN/WAN.

There are three major categories for the transmission of voice over IP, according to the coding technique used:

- time coding (with speeds between 16 and 64 kbit/s);
- parametric coding (with speeds between 2.4 and 4.8 kbit/s);
- analysis-synthesis coding (with speeds between 5 and 16 kbit/s).

Generally speaking, coding techniques giving low bit rates call for longer processing times, thereby increasing transit time. It is currently acknowledged that the average time for processing speech (compression, decompression and packetization) introduces a delay of some 50 ms for one end of the link.

Network delay

- Propagation: on a wire network, the speed of propagation is 200 000 km/s, making for an appreciable propagation time.
- Routing and queues: according to the nature of the network, different times may be indexed.

In the case of a well-controlled IP network such as an intranet or equivalent, the transmission of packets takes between 50 and 100 ms (propagation and jitter compensation), with the routers introducing a delay of some 50 ms. The total resulting delay therefore lies between 200 and 250 ms from end to end for a well-controlled IP network (intranet). These delays are significantly greater and even indeterminate (at busy periods) in the case of the Internet.

B.3 Jitter

Jitter buffer: this buffer memory allows for resynchronization of packets arriving with variable delays. It therefore serves to compensate for time-lags and to restore packets to their correct order.

- depacketization;
- decompression;
- decoding and digital-to-analogue conversion.

The result of this, under current conditions in terms of the technologies used by the Internet and their dimensioning, is that IP telephony would be feasible only on a controlled IP network of the intranet type, being far more unpredictable over the Internet.

Transmission Jitter

Jitter may be described as the variation in transmission delay. The protocol used to transport voice packets over the Internet (an IP network) is the User Datagram Protocol (UDP). The signalling side uses the Transmission Control Protocol (TCP) layer. The UDP protocol operates in unconnected mode, in which the packets do not necessarily take the same route, whence a variation in transit time. Another cause of transit time variation may be the number of routers encountered and the load borne by each such router. In order to reconstitute a synchronous flux at the receiving end, jitter compensation buffers are installed. However, this buffering increases even further the transmission delay. In order for an acceptable level of quality to be maintained, the jitter must remain below 100 ms.

B.4 Echo

Echo may be described as the time that elapses between the transmission of a signal and the reception of that same signal in the form of an echo. This problem generally arises in the context of PC-to-telephone, telephone-to-PC or telephone-to-telephone communications. It is caused by the electronic components of analogue parts of the system returning a part of the processed signal.

An echo smaller than 50 ms is imperceptible. Above that level, the speaker will hear his or her own voice just after speaking. Where the aim is to provide an IP telephony service, the gateways must process the electrical echo generated by the transfer from two to four wires. If such processing is not carried out, it will not be possible to use the service with conventional analogue sets. To resolve the problem, high-performance echo cancellers are installed at the gateway stage of the network.

Annex C – Protocols for providing VoIP with a good quality of service

C.1 RTP (real-time transport protocol)

The RTP protocol is a transport and control protocol geared to applications with real-time characteristics. It is covered in IETF Recommendation RFC1889 and provides facilities for applications to:

- reconstruct the time basis of audio, video and real-time data streams in general;
- detect packet losses rapidly and inform the source within time-frames compatible with the service;
- identify the content of data and enable secure transmission.

The RTP protocol is independent of the underlying transport protocol and of the networks crossed. It will typically be employed above the simple datagram protocol such as UDP. The RTP protocol operates end to end and does not reserve any resources in the network, since no action is effected on the routers (quality-of-service control is not carried out with the RTP protocol). The RTP protocol is often usefully supplemented by a resource reservation protocol like the RSVP protocol. The RTP protocol provides no reliability. It only offers certain characteristics of a transport protocol. It does not provide for automatic re-sending of missing packets.

Although the RTP protocol does not guarantee delivery time, its contribution to real-time exchanges is significant. The RTP protocol provides information which is extremely useful for the transport of content. It timestamps packets with the time at which they were generated, thus simplifying delivery to the addressee in the right order. It also includes mechanisms for detecting and synchronizing different streams, whereby each packet is immediately recognized as belonging to a specific stream.

The main characteristics of the RTP protocol are summarized in the table below:

Table C.1 – Real-time transport protocol

Reliability	<ul style="list-style-type: none"> • RTP is not reliable if used with UDP or IP, which are themselves not reliable. • RTP can rely on the reliable service provided by the lower layers of networks in connected mode (e.g. ATM layers, AAL3/4 or AAL5).
Congestion control	<ul style="list-style-type: none"> • RTP has no built-in congestion control mechanism like TCP.
Stability of streams	<ul style="list-style-type: none"> • RTP does not guarantee control of transmission times or real-time stream continuity.
Resources	<ul style="list-style-type: none"> • RTP does not reserve any resources and has no direct impact on network behaviour.
Information and tools for the addressee	<ul style="list-style-type: none"> • The RTP header contains various items of information for synchronization and restitution of the signal by the receiver: time stamp, stream and sequencing indices, contributing sources, etc.
Information for the sender	<ul style="list-style-type: none"> • The RTP protocol by itself does not provide any useful information to the sender. It is generally used with the RTCP protocol which provides the sender with comprehensive feedback on transmission quality: packet losses, delays, etc. It enables the sender to modulate its output rate according to available resources.

C.2 RTCP (real-time transport control protocol)

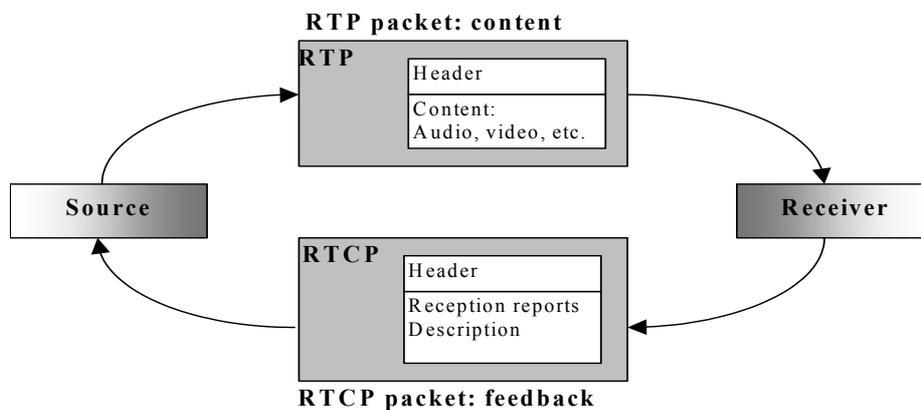
The RTCP protocol is based on periodic transmission of control packets to all participants in a session. It uses the same transmission mechanism as RTP data packets. It is the underlying protocol, in this case the UDP protocol, which multiplexes the RTP data packets and RTCP control packets. The RTCP packet only contains information required for transport control. It does not carry any content. It comprises a set header, similar to that of the RTP packets transporting the content, followed by other elements that depend on the type of RTCP packet. Several types of RTCP packets are defined, for transporting a wide variety of control information. The table below shows the five most common types of RTCP packet.

Table C.2 – RTCP packet types

SR (Sender Report)	Set of transmission and reception statistics from participants who are active senders.
RR (Receiver Report)	Set of statistics from participants who are only receivers and not active senders.
SDES (Source DEscription)	Source description packets are made up of several elements, including the CNAME. They constitute the source's "visiting card".
BYE (End message)	Indicates quitting a session.
APP	Functions specific to a particular application.

The addressees of RTP packets send back information on reception quality, using different forms of RTCP packets, according to whether the addressee is itself a content sender or not. The two types, SR and RR, contain zero or one or more receiver report blocks, intended for each of the synchronization sources from which the receiver has received an RTP content packet since the last report. Gauging the quality of reception is not only useful for the sender, but also the receiver and any network supervisor that might exist. The sender can modify its transmission according to the feedback received; the receiver can infer whether the reception difficulties it is encountering are of local, regional or broader origin. Any supervisors will only receive the RTCP packets, enabling them to evaluate network performance.

Figure C.1 – RTP and RTCP packets for control of reception quality



C.3 RSVP (Resource ReSerVation Protocol)

The Internet Protocol was voluntarily designed to push the intelligence towards the end systems. It is this simplicity and absence of different states that has made the protocol so successful. Nevertheless, despite efforts to adapt terminal systems (transmitters and receivers), critical dysfunctions are still generated within the body of the network. The systems within the IP network still apply the same routing processes irrespective of the source of the packets, which are all treated on an absolutely equal footing according to the FIFO (First In, First Out) principle. It is a simple transport and routing process, easy to implement and entailing minimal processing for the routers. The node determines the path that the packet will follow on the basis of routing tables and in particular of the following two criteria:

- the number of relays or stages to the destination: the shortest route is preferred;
- the installed capacity of the links: the best bit rate is selected.

In practice, the first criterion carries more weight in the routing decision, which explains the tendency for packets to follow one and the same path during a connection session.

On the basis of this principle for the functioning of IP network routers, a real-time stream such as the packets of a telephone call will systematically be placed at the end of the queue in a router, like all other types of packets. This operating principle is thus incapable of coping with the time constraints imposed by real-time transmissions.

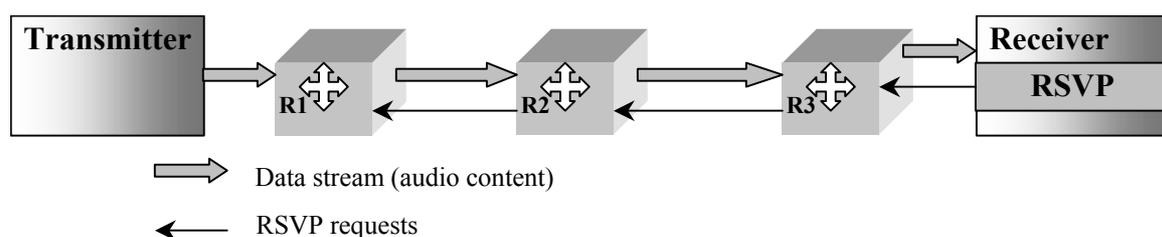
One of the most commonly envisaged solutions consists in building into the IP network routers a dynamic strategy for regulating individual streams.

Since 1989, several proposals by IETF working parties have suggested the introduction in routers of "fair queuing" mechanisms, in proportion to the quality of service required for each application. This reflection led to the development and subsequent adoption of the Resource ReSerVation Protocol (RSVP) protocol, which acts on the network at the level of its routers, so as to channel and discipline their behaviour to make it compatible with the requirements of real-time operation.

The RSVP protocol may be considered as one of the ways of enabling the Internet to become an integrated service network providing both a "best effort" type service and a real-time type quality of service. When a real-time type application requires a given performance level for its data stream, the RSVP protocol will request the routers of the path or paths to reserve sufficient resources to maintain that level of quality.

The RSVP protocol is piloted by the receiver. It is the addressee (and not the transmitter) which files a request for a quality of service corresponding to its needs. This request is routed to the sender in the form of an RSVP message. RSVP messages circulate in the opposite direction to the data stream.

Figure C.2 – Data flow and RSVP requests



When an application requires a given level of quality of service, it transmits the corresponding request to the RSVP software core housed in the router. The router distributes the request to all intermediate routers which the packet goes through from the source. In each node, and according to the quality of service requested, the RSVP protocol places the packets in order on the basis of a decision procedure (admission control). If the packet passes this stage successfully, then the RSVP protocol has established a packet profile, transmitted to the packet classifier, which sorts packets according to their route and profile. Thus, the packet will have a level of priority in the router transmission queue corresponding to the requested quality of service. The reservation of resources is intrinsically inequitable, favouring certain flows and certain receivers over others. The only way to justify this is to charge for it. The RSVP protocol registers a user's order for a given quality of service leading to the reservation of resources, so a means of billing on the basis of consumed bandwidth can be conceived.

C.4 DiffServ protocol (differentiated services protocol)

The service differentiation architecture, currently being standardized in the IETF RFC2475 DiffServ working party, serves to modify the way in which resources are shared in the network. In today's Internet, the network does its best to transport packets without differentiating them. The routers apply the same treatment to every packet. Flow control is carried out end to end, in other words the network leaves it up to the ends to share out the bandwidth. Thus, TCP connections are assumed to each use an equal share of the bandwidth utilized. In the bandwidth service differentiation architecture, packet loss rate and transit delay are influenced by traffic packaging operations carried out at the entrance to the network and by modifications made to the behaviour of routers in the body of the network. With service differentiation, in a congestion situation packet losses can be switched over to certain classes of traffic in order to protect others. It offers no guarantee of flows, since there is never any dynamic admission control that will avoid congestion. Admission control is carried out *a priori* by defining a contract for each class of traffic and dimensioning resources to guarantee that contract.

Two types of routers are distinguished in the service differentiation architecture:

- Edge routers are located at the boundary of a domain and take care of shaping and classifying traffic. One of their functions is to allocate a DiffServe Code Point (DSCP) label to all packets entering the domain. The value of this label for a given stream depends on the service level specification (SLS) attributed to the stream by the network and the instantaneous behaviour of the stream. Once the labelled packet has entered the network, the latter uses the DSCP protocol to select the queue and arbitrate between packets in the event of congestion.
- Routers within the network whose per hop behaviour (PHB) is dependent on the DSCP protocol.

Thus, to include a new service in a service differentiation network, one has to define the behaviour of the routers for each DSCP and the functions supported by the edge routers. In fact, three aspects of a new service can be distinguished. First, the new service must be capable of allocating resources according to the contract each customer has concluded with the network; in other words, it must distribute bandwidth according to the SLS applicable to each stream. Second, the new service must respect the priority attributed to each packet by the source. When the packet starts out, this priority represents the semantic value, but it can be modified by the network's edge routers when streams are aggregated or the behaviour of the source exceeds the limits set under the contract. Finally, resource allocation must remain consistent with the various SLS specifications, both at times of congestion and when the network is being used below capacity.

For now, IETF has defined two DiffServ services (i.e. two PHBs), as well as the default "best effort" PHB (DSCP = 000000). These are:

- expedited forwarding (EF) PHB
- assured forwarding (AF) PHB.

Packets marked for EF per hop behaviour (DSCP = 101110) receive a qualitatively better forwarding service than "best effort". To this end, it is ensured that the EF aggregated initial bit rate is greater than or equal to its arrival bit rate. Then, EF traffic generally encounters a queue which should be short and rapidly processed so that latency time, jitter and packet loss are kept low. EF can offer a sort of virtual "leased line" type service.

AF per hop behaviour is intended for more general services. The AF specification defines four classes and three levels of rejection priority (DP) characterizing the relative importance of a packet in a particular class in the event of congestion. Each class may be seen as a separate queue using a certain proportion of network resources. For each class a queue management algorithm is used that takes account of packet discard priority. In the event of congestion, the algorithm discards the least important packets first. Thus, the degree of assured forwarding each packet enjoys in a particular AF class depends primarily on:

- allocated resources
- available load for the class
- packet discard priority.

For flows using AF behaviour, the packet DSCP reflects the packet's class and discard priority. While packets in one and the same stream must belong to the same class in order to avoid being disordered, they may have different discard priorities. These latter priorities can then be used to differentiate between streams or to differentiate between different information within the same stream.

Table C.3 – Access codes for AF (assured forwarding) differentiated services

	Class 1	Class 2	Class 3	Class 4
Low rejection priority	001010	010010	011010	100010
Medium rejection priority	001100	010100	011100	100100
High rejection priority	001110	010110	011110	100110

C.5 MPLS protocol (multiprotocol label switching protocol)

In addition to conventional data, the Internet now has to transport voice and multimedia data. The resources necessary for these new services, in terms of bit rate and bandwidth, have led to a change in Internet infrastructure. The exponential growth in the number of users and the volume of traffic adds a further dimension to the problem. The classes of service (CoS) and quality of service (QoS) have to be taken into account in catering for the different needs of each network user.

The MPLS protocol is a proposed solution to respond to the problems posed by current networks. It has emerged as a solution for organizing the combination of bandwidth management and service needs for new IP networks. The MPLS protocol proposes solutions linked to the scalability (adjustment to network scale) and routing (based on QoS and QoS measurements). It can be adapted to ATM and Frame Relay networks. The MPLS protocol will play an important role in routing, switching and packet transfer over new-generation networks.

Initial Internet deployment was geared to data transfer over the network. To that end, a simple software router and network interfaces sufficed. As the possibility of supporting high-speed transmissions emerged, elements capable of switching at level 2 and level 3 have had to be deployed within the hardware. These solutions respond to rapid packet transfer needs across the network, but do

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not cater for service needs for the information contained in the packets. Furthermore, most of the routing protocols currently deployed are based on algorithms geared to the fastest possible transfer through the network, but do not take into account other factors, such as delays or congestion, which can significantly diminish network performances. Traffic management is an objective for network administrators.

C.5.1 MPLS components

MPLS is standardized by IETF. It fulfils the following functions:

- It specifies the mechanisms for administering traffic flows of several types, such as flows between different hardware, different machines or even different applications.
- It is independent of the layer 2 and 3 protocol.
- It interacts with existing routing protocols such as RSVP (resource reservation protocol) and OSPF (open shortest path first).
- It supports the level 2 layers in IP, ATM and frame relay networks.

In MPLS, data transmission is carried out on label-switched paths (LSP). These are a sequence of labels at each node in the path from the source to the destination. LSPs may be control-driven (i.e. set according to the type of data transmission) or data-driven (i.e. set after detection of a certain type of data). The labels, which are specific identifiers for the low-layer protocol, are distributed according to the label distribution protocol (LDP) or RSVP protocol. Each data packet encapsulates and transports labels during their routing. High-speed switching is possible, since the fixed link labels are inserted at the very beginning of the packet or cell and can be used by the hardware to effect more rapid switching.

C.5.2 LSR and LER

The elements involved in the MPLS protocol mechanisms may be separated at label edge routers (LER) and label switching routers (LSR). An LSR is a high-speed router within an MPLS network which is involved in setting LSPs. An LER is an element at the edge of the access network or MPLS network. LERs may support several ports connected to different networks (ATM, frame relay or Ethernet) forwarding traffic over the MPLS network after setting LSPs. The LER plays a fundamental role in assigning and deleting labels as traffic enters and exits the MPLS network.

C.5.3 FEC

Forward equivalence class (FEC) is the representation of a group of packets having the same transport needs. All the packets of such a group receive the same treatment during their routing. Unlike conventional IP transmissions, in MPLS a packet is assigned an FEC once only, when it enters the network. FECs are based on needs in terms of service for certain groups of packets, or even a certain address prefix. Each LSR builds a table in order to know how a packet should be transmitted. This table is called the label information base (LIB).

C.5.4 Labels and association of labels

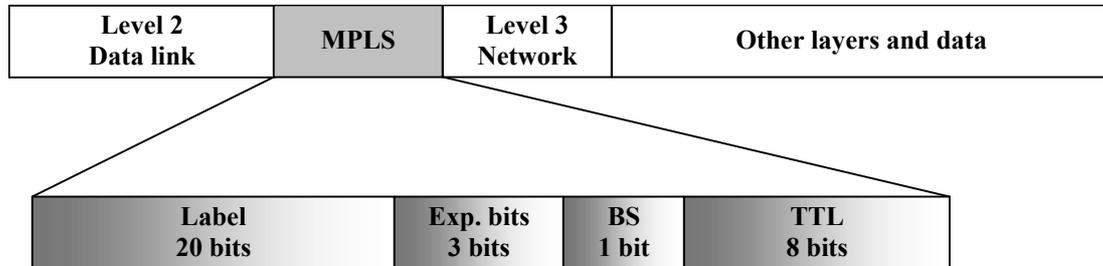
A label, in its simplest form, identifies the path which the packet has to follow. A label is transported or encapsulated in the packet's level 2 header. The router receiving it examines the packet to determine the following hop according to its label. Once a packet is labelled, the rest of its journey is based on label switching. The values of the labels have purely local significance. These values may indeed directly determine a virtual path (DLCI in frame relay or VCI and VPI in ATM).

The labels are associated with an FEC according to a certain logic or policy determining such association. The decision can be taken on the following criteria: unicast routing to destination, traffic management, multicast, virtual private network (VPN) or QoS.

C.5.5 Basic form of MPLS labels

The generic format of a label is illustrated in the figure below. The label may also be located in the layer 2 header or between layers 2 and 3.

Figure C.3 – Generic label format



Annex D – The IPSec protocol

There are several approaches for securing exchanges over an IP network:

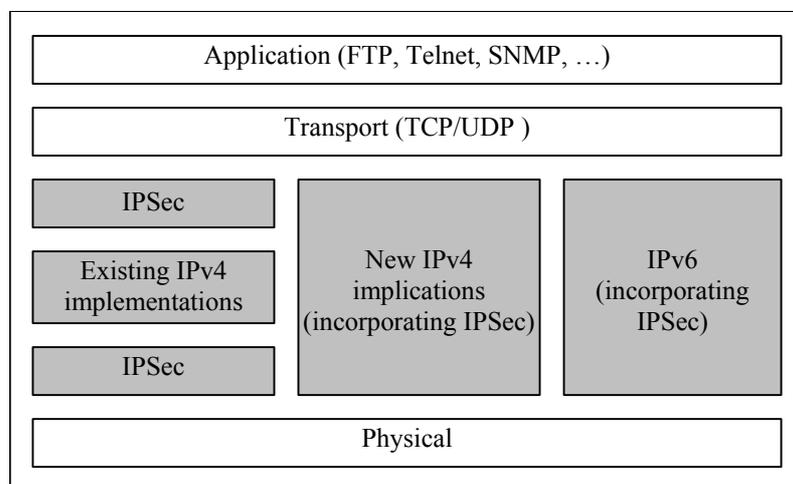
- Application level (PGP)
- Transport level (TLS/SSL, SSH protocols)
- Physical level (encryption units)

IPSec aims to secure exchanges at the network layer level. Since the IPv4 network is widely deployed and full migration to IPv6 will still take some time, an obvious interest emerged in defining security mechanisms that would be common to both IPv4 and IPv6. These mechanisms are commonly designated by the term IPSec (IP Security Protocols). IPSec therefore provides:

- confidentiality and protection against traffic analysis;
- data (and source) authentication;
- data integrity (in unconnected mode);
- protection against rejection;
- access control.

IPSec is a security extension for the IP protocol. It can be implemented on all network equipments and many suppliers are already building it into their product. Example of use: VPN (virtual private networks), for securing remote access to an Intranet.

Figure D.1 – Positioning of the IPSec protocol in the IP pile



SA: "Security Association". The IPSec security association is a connection which provides security services for the traffic it transports. It is a data structure for storing all the parameters associated with a given communication. An SA is unidirectional; thus, protecting the two directions of a conventional communication requires two associations, one in each direction. The security services are provided through the use of either AH or ESP. The role of an SA is thus to consign, for each IP address with which the IPSec implementation may communicate, the following information:

- SA index called the security parameter index (SPI) selected by the receiver;
- sequence number, an indicator used for the anti-rejection service;
- an anti-rejection window: 32 bits counter;

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- sequence overrun;
- authentication parameters (algorithms and keys);
- encryption parameters (idem);
- SA lifetime;
- IPSec protocol mode (tunnel or transport);
- ...

Each association is uniquely identified by means of a triad comprising:

- packet destination address;
- security protocol identifier (AH or ESP);
- SPI.

SAs contain all the parameters required for IPSec, including the keys used. Management of keys for IPSec is only linked to the other IPSec security mechanisms through the SAs. An SA can be manually configured in a simple case, but generally speaking it consists in using a specific protocol enabling dynamic negotiation of SAs and in particular exchange of session keys.

The SA negotiation protocol, developed for IPSec, is the Internet Security Association and Key Management Protocol (ISAKMP). This is a generic framework permitting the use of several key exchange protocols, meaning that it cannot be used on its own. Within the framework of IPSec standardization, ISAKMP is associated with part of the SKEME and Oakley protocols to give a final protocol called Internet key exchange (IKE).

Authentication header (AH) protocol

AH is designed to ensure integrity in unconnected mode and the authentication of IP datagram sources without data encryption (no confidentiality). The lack of confidentiality means that the standard will be broadly diffusible on the Internet including in places where the export, import or utilization of encryption for confidentiality purposes is restricted by law. Its principle is to attach to the conventional IP datagram an additional field enabling the authenticity of the data included in the datagram to be verified on reception. This data block is called the integrity check value (ICV). Protection against rejection is provided through a sequence number.

Encapsulating security payload (ESP) protocol

ESP can provide a choice of one or more of the following services:

- data confidentiality and partial protection against traffic analysis (in tunnel mode);
- data integrity (in non-connected mode) and data source authentication, partial protection against replay.

Unlike with AH, where one simply adds an additional header to the IP packet, ESP operates on the basis of the principle of encapsulation: the source data are encrypted and then encapsulated.

Annex E – Coding principles and techniques

Audio applications on a network involve two distinct aspects, namely digitization and coding of audio data, and packetization of the data for transmission in the network. The purpose of audio coding is to transform an analogue voice signal into a digital signal with a given bit rate and quality.

The first coding operation is sampling of the analogue signal at a specified sampling frequency and with a specified degree of accuracy, characterized by the number of bits used to encode the amplitude of each sample. Obviously the choice of frequency and number of bits used has to be a trade-off between bit rate and quality of the encoded signal. The higher the quality desired, the higher the bit rate obtained after sampling.

Sampling theory states that an analogue signal can be reconstructed from digitized samples if the sampling frequency is at least twice the bandwidth of the original signal. The human ear is capable of perceiving a range of frequencies from around 20 Hz to 20 kHz, i.e. a bandwidth of 20 kHz. The simplest encoding algorithm comes down to merely sampling an analogue signal and quantifying the samples (in other words converting them from real values to values with a finite accuracy). This coding, called pulse code modulation (PCM) and conforming to standard G.721, is used to encode voice in the telephone network. Since the bandwidth of a twisted pair is around 3.5 kHz, the sampling frequency has thus been set at 8 kHz in order to comply with the sampling theory. Quantification is carried out with a logarithmic scale over 8 bits, which is equivalent to a linear quantification over 13 bits.

E.1 Differential coding (DPCM, ADPCM, ADM)

PCM coding lies at the root of a family of widely used differential codings: it is based on the observation that there is a strong correlation between successive samples of an audio source. It is thus a good idea, rather than encoding the samples themselves, to encode the difference between successive samples. There are three types of differential coding:

- **Principle of DPCM**

A DPCM coder is based on the principle that there is a correlation between successive samples, and it is thus possible to predict sample $n+1$ from samples n , and also $n-1$, $n-2$, etc. It therefore suffices to encode only the difference between a sample x_n and the prediction of x_n . In the case of DPCM coding, the prediction of x_n is simply the value of the previous sample x_{n-1} . In a broader case, linear prediction can be used. We then obtain an ADPCM type coding.

- **Principle of ADPCM coding**

The prediction for this type of coder is obtained by a feedback loop: after quantification, the error will be restored. The adaptive prediction uses the short-term autocorrelation function calculated on portions of signal of duration not exceeding 20 ms. The sample x_n will be predicted by a certain combination of the previous p samples.

- **Principle of ADM coding**

Adaptive delta modulation (ADM) coding with adaptive step is similar to ADPCM coding described above. The main difference resides in the fact that the difference between the current sample x_n and the predicted sample \hat{x}_n is quantified with a certain granularity and thus a variable quantification step. Equivalently, one can say that this difference is quantified at several levels. The quantification steps are made proportional to the variance of the input signal x . The ADM coding used in Internet audio tools is called ADPCM DVI. But it follows the above principle. The algorithm calculates the difference between the current sample and its predicted value, this difference is then quantified over 4 bits, then it is added to the previous predicted value in order to obtain a prediction on the next sample.

E.2 Synthesis coding (LPC, CELP)

Synthesis coding is very different from differential coding. The idea is no longer to manipulate samples in order to eliminate redundancy and correlations between samples, but on the contrary to consider blocks of samples and construct a model which generates samples that are statistically identical (or similar) to the original samples. With this model, therefore, it is possible to synthesize samples with specific statistical properties, whence the term synthesis coding. Two types of synthesis coding may be quoted:

- **LPC coding**

Linear predictive coding (LPC) consists in synthesizing samples from a model of a speech production system and an excitation. For human voice, the speech production system is the lung-vocal chords-trachea-throat-mouth-lips system. In practice, this system is modelled by a set of cylinders of different diameters, 10 in the case of [LPC-10], excited by a signal which is either sinusoidal or white noise. The choice of the excitation function (sinusoidal or white noise) depends on the characteristics – voiced or non-voiced – of the signal. A voiced signal corresponds to a letter such as "a" or "u". A non-voiced signal corresponds to a letter like "r" or "s".

There are thus two phases in the synthesis process, namely:

- identification of the excitation function; and
- identification of the cylinder diameters (or more generally identification of the voice model transfer function).

Each phase is executed every 20 ms (i.e. on a set of 160 samples for a sampling frequency of 8 kHz). The coder generates as output an excitation frequency (encoded on 16 bits), a set of 10 coefficients (coded on 10×8 bits) and a gain (coded on 8 bits). The bit rate of the coder is thus 104 bits every 20 ms, i.e. 5.2 kbit/s. This bit rate is thus only one-twelfth of that of a PCM coder.

- **CELP coding**

Code excited linear predictive (CELP) coding, which corresponds to Standard FED_STD 1016, is an extension of LPC coding. It still comprises two phases, corresponding to the excitation and transfer functions. The identification of the transfer function is the same as for LPC. The excitation function, on the other hand, is not only white noise or sinusoidal, but a linear combination of a stochastic (i.e. noise) and periodic functions. The identification of these functions consumes a large amount of CPU time (indeed CELP encoders are generally implemented with specific signal processing cards), but the quality obtained is much better than with the LPC coder.

Annex F – Application level protocols for Voice over IP

F.1 ITU-T H.323 protocol

H.323 was originally defined by ITU-T Study Group 16 as a variant of standard H.320 for videophone on ISDN adapted in this case to local data networks of type Ethernet, Fast Ethernet and Token Ring. It relates to call control, multimedia management, and bandwidth management for point-to-point and multipoint conferences. H.323 also handles interfacing between LAN and other networks. It is designed for coding and compression of voice and image signals transported by IP networks. The standards it contains can be implemented irrespective of the physical medium on which the IP network is founded: ATM, FDDI, etc., operating in unconnected mode and without quality of service guarantee (no error correction). The H.323 protocol is today considered as *the* Internet telephony standard. Henceforth, this standard applies to all packet networks and no longer only local networks.

H.323 entirely subsumes the RTP/RTCP standard, specifying points of detail for a particular scope of utilization defined by H.225. It sets the type of audio and video content of the RTP packets created by an H.323 application and arbitrates certain conflicts between RTCP and the control protocol defined by H.323 (sub-assembly H.245). H.323 comprehensively describes a videotelephone system on LAN, including advanced functions such as conferencing, access control or stream mixing.

H.323 describes all the units that interact in the functioning of such a system:

- H.323 terminals:

A terminal may be a PC, a telephone set, a specialized videoconference terminal or a fax machine on the Internet. The minimum imposed by H.323 is that it should implement the G.711 speech compression standard, use the H.245 protocol for negotiation of channel opening and the establishment of call parameters, and the Q.931 signalling protocol for call set-up and termination. A terminal also possesses optional functions, in particular for group work, and file sharing capabilities. There are two types of H.323 terminals, one high quality (for use on LAN), the other optimized for low bandwidths (28.8/33.6 kbit/s – G.723.1 and H.263).

- Gateways to conventional networks (PTN, ISDN, etc.)

The H.323 gateways provide interconnection with other networks [H.320 (ISDN), H.324 (modem), conventional telephones, etc.]. They ensure Q.931 signalling correspondence, control signal correspondence and consistency of media (multiplexing, bit rate matching, audio transcoding). Many functions are left to the discretion of manufacturers:

- number of terminals that can communicate through the gateway;
- number of circuit-switched network connections;
- number of simultaneous conferences accepted;
- audio, video, data conversion functions, multipoint functions.

- Gatekeepers, which are kinds of management and registration centres which also control the access of terminals to the IP network:

The role of gatekeepers is to translate addresses and manage authorizations. Address translation is not conventional IP address translation, but the association between an H.323 alias (H.323 user identifier) and an IP address stemming from the terminal referencing. E-mail type addresses are possible, as well as telephone-number type addresses. Management of authorizations enables the granting or refusal of permission to carry out a call, limiting of the bandwidth where necessary and managing traffic on the LAN. Gatekeepers also serve to manage the gateways for H.320, H.324, conventional telephones, call signalling to route calls in order to offer supplementary services or multipoint controller (MC) functionalities. Gatekeepers also serve to manage calls, daily logs and reporting.

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- Multipoint conference unit (MCU), multipoint controller (MC) and multipoint processor (MP) responsible for stream mixing and multipoint conference management:

Setting up a conference involving three or more parties requires the intervention of a multipoint control unit (MCU) which handles the negotiation between the terminals under the H.245 standard. These functions are available in the form of software installed on the local network server or in the form of hardware.

During a connection, several channels are opened, each with its own address (UDP or TCP port according to the type of channel). H.323 is defined for videoconferencing: it is therefore possible to exchange sound or video image. For each type of media exchanged and for each direction of communication, an RTP channel is set-up as well as an RTCP control channel (above the UDP protocol). It is also possible to exchange data on a specific T.120 channel (above the TCP reliable transport protocol). Two other channels are linked to call signalling (similar to the case of ISDN Q.931 signalling) and call control. A final type of channel is linked to optional exchange with a gatekeeper governing the access of terminals to the network. This channel supports the operations of registration, admission and status request to the gatekeeper. In total, a multimedia PC wishing to set-up a voice and data connection with another PC via an IP network will thus have to establish the following channels:

- send channel for an audio stream (on UDP/RTP);
- receive channel for an audio stream (on UDP/RTP);
- receive channel for audio control information (on UDP/RTCP);
- send channel for audio control information (on UDP/RTCP);
- data send channel (on TCP/T.120);
- data receive channel (on TCP/T.120);
- call signalling channel;
- terminal capacity control and exchange channel;
- channel for registration and admission with a gatekeeper.

F.2 IETF SIP protocol

SIP (Session Initiation Protocol) is a signalling protocol used to establish, modify and terminate voice calls and (multiparty) multimedia sessions over IP networks (intranets and/or Internet). SIP is a client-server protocol similar in syntax and semantics to the HTTP protocol used by the World-Wide-Web application. As a matter of fact, Web (http) and SIP servers may even coexist and be integrated together.

The client and server roles are functional; that is, a client may behave as a server and vice versa. In order to establish a call, a client sends SIP requests to a server and the latter listens to call requests and prompts the user or executes a programme to determine the response.

SIP defines three types of servers: Registrar, Proxy, and Redirect. These are functional roles; a given physical entity may simultaneously assume any of them, the protocol being the same. A registrar server receives registrations from clients regarding their current location; this subsequently helps locating them for call completion. A proxy server forwards client requests to the final destination or to one or more other SIP servers. A redirect server redirects users to try another SIP server as the next-hop towards the destination.

SIP consists of seven textual messages; these are:

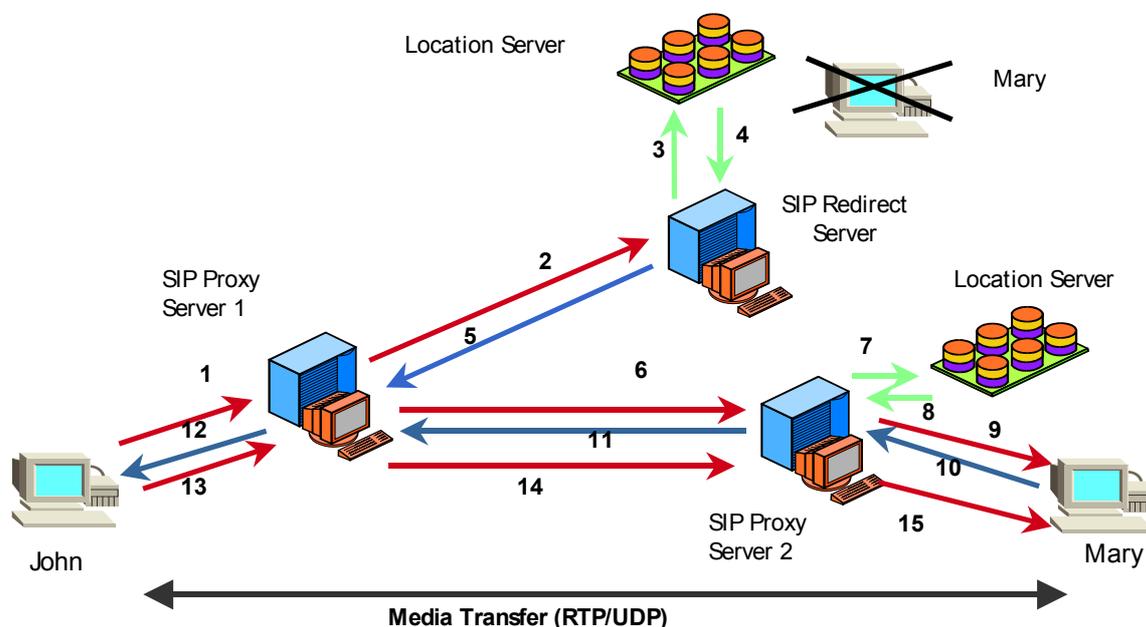
- INVITE: invites a user to a phone call or a conference
- BYE: terminates a connection between two users
- OPTIONS: solicits information about user capabilities
- STATUS: informs another server about the progress of signalling actions that are in progress
- CANCEL: terminates a search for a user
- ACK: confirms that a client has received a final response to an INVITE
- REGISTER: conveys user location information to a SIP server

SIP users are addressed by e-mail like SIP URLs (similar to those of http). These may indicate a user within a domain (sip:user@domain), at a given host (sip:user@host), at a specific host IP address (sip:user@IP_address), or even at a phone number (E.164 number) accessed through an IP/PSTN gateway (sip:phone_number@gateway).

When sending an INVITE message, the client may use the least precise of the above addresses (say the user at domain one). SIP addressing resolution is not only based on the SIP servers described above but also on the existing addressing resolution entities within IP network, that is: DNS and location servers (LDAP server, corporate database, etc.). The objective of locating the address where the called party can actually be reached is based on the usage of SIP servers and the above entities.

The figure below illustrates an example of a call establishment between two users John and Mary. John, using her domain name, addresses Mary. The example illustrates the case where Mary moved from one host to another since the last call it received from John's proxy server.

Figure F.1 – Call establishment using SIP



John sends an INVITE message (1) to SIP Proxy Server 1 providing Mary's domain name address such as sip:mary@domain; the knowledge of that server may already be pre-configured in John's host or determined following a request to a DNS service (not shown in the figure). SIP Proxy Server 1, from the analysis of the provided domain name, relays the INVITE message (2) to the SIP Redirect Server²³. The SIP Redirect Server sends a request (3) to a Location Server, that replies with a message (4) indicating that Mary is no more on the indicated host. The SIP Redirect Server sends a STATUS message (5) to SIP Proxy Server 1 informing it that it should send the INVITE message to SIP Proxy Server 2. SIP Proxy Server 1 sends an INVITE message (6) to SIP Proxy Server 2. SIP Proxy Server 2 checks Mary's host location through an interaction with a Location Server (7 and 8) and sends the INVITE message to Mary's host (9). Mary's host sends a response message (10) to SIP Proxy Server 2 that sends it back to SIP Proxy Server 1 (11) and the latter sends it to John's host (12). If the response message is an OK message, then John's host may send an ACK message (13) to SIP Proxy Server 1 that redirects it to SIP Proxy Server 2 (14) and the latter to Mary's host (15). Mary and John can then directly communicate over the IP network and RTP/UDP protocol.

²³ This server is named here "Redirect" because of the specific role it will play in that particular call.

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It should be noted that, contrary to the H.323 where the network layer is not specifically prescribed, SIP prescribes the usage of the IP protocol. For the upper layers, however, both protocols transfer data over RTP/UDP. Regarding signalling messages and responses, SIP prescribes the usage of UDP. However, if the signalling has to traverse firewalls, then usage of TCP may be necessary.

SIP is a lower-complexity protocol than H.323, but it builds on existing location and DNS services offered by IP networks. The communication model of SIP allows a relatively easy way of establishing a voice or multimedia communication between many users connected to an intranet/Internet host computer; SIP servers may be much less complex than H.323 gatekeepers but they are even less prone to added-value services or any network control over the call.

Finally, regarding the interface with telephony networks, the SIP addressing scheme allows the routing of calls to subscribers of those networks through a gateway device connected to the IP network. A specific extension of SIP, named SIP-T, allows – at the PSTN/IP interconnection point – for calls coming from the telephony network to encapsulate SS7 ISUP messages within SIP so that information necessary for services is not discarded. On the other hand, certain – not all – information is translated from an SS7 ISUP message to generate the corresponding SIP header information in order to facilitate the routing of SIP messages.

Annex G – Network level protocols for Voice over IP

G.1 ITU-T H.248/IETF H.248/MEGACO protocol

The H.248/MEGACO protocol had its origin from earlier protocols defined by the Internet Engineering Task Force (IETF), the most famous of them being MGCP. Since May 1999, IETF and ITU-T Study Group 16 agreed to make a joint specification effort for the H.248/MEGACO protocol whereby a common specification could be agreed by both parties. This annex focuses on the functional principles of the H.248/MEGACO protocol, designated hereafter as H.248.

The H.248 protocol basically consists of a connection model offered by a Media Gateway (MG) to the Media Gateway Controller (MGC); a set of commands that act on the objects of that model; and, a capability for grouping commands into transactions.

The model

The protocol defines a connection model of logical entities, or objects, within the MG that can be controlled by the MGC. Two basic abstractions are used in that model: **termination** and **context**. A termination is capable of sourcing and sinking one or more media. A context is an association between a collection of terminations that make up a single conference. A context with more than two terminations describes the conference bridging properties.

The commands

A context is an association between a number of terminations that describes a conference. An **Add** command is used by the MGC to add a termination to a given context. If no context is specified, the MG should create a new one. A **Subtract** command removes a termination from a context. When the last termination is removed from a context, the MG implicitly destroys it. A **Move** command moves a termination from one context to another. A termination can exist within only one context at a time.

A termination is a logical entity within the MG that source and/or sinks media and/or control streams. Terminations representing physical interfaces to the MG will be created at boot time and are permanent²⁴. Such terminations exist within a Null context, they are removed from that context when the Add command is used to add them to another context and given back to it upon invocation the Subtract command. Terminations representing stream sources/sinks that are instantiated by a higher layer protocol (such as RTP streams) and exist only for the duration of that stream are created/removed by the MG upon invocation of the Add/Subtract commands upon them²⁵.

A termination is described by a number of characterizing *properties*; it may have *signals* applied to it (such as tones and announcements) and may be programmed to detect *events*. The **Modify** command allows the MGC to modify properties, signals and events of a given termination. When a given event occurs on a given termination, the **Notify** command allows the MG to inform the MGC of its occurrence.

Different types of MG may implement terminations that have widely differing characteristics. Such variations are accommodated by the H.248 protocol by allowing terminations to have optional properties, event signals and statistics. Such options are grouped into Packages and a termination realizes a set of such packages, The protocol defines only base-level properties used to describe terminations. Properties not included in the base protocol are defined in *Packages*. An MGC can audit a MG for the current state of properties, events and signals of terminations using the **AuditValue** command. The **AuditCapabilities** command allows an MGC to inquire a given MG about all possible values of termination properties, events and signals allowed by it.

²⁴ One can make an analogy between such terminations and the circuit resources of a telecom switch.

²⁵ Such terminations – as opposed to static ones – represent bearers of networks not based on a connection-oriented model such as IP networks.

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Finally, the **ServiceChange** command allows the MG to notify the MGC that a termination or group of terminations of service is about to be taken or has just been returned to service. This command is also used by the MG to announce its availability to an MGC, and to notify the MGC of impending or completed restart of the MG.

Transactions

Commands between the MGC and the MG are grouped into transactions; each identified by a transaction identifier. A transaction regroups one or more *Actions*, each consisting of a series of commands that is limited to operating within a single context. However, when an action involves a modification of a termination outside of a context (Modify command), or where the MGC requests the MG to create a new context (Add command of the first termination), the specification of a context identifier for a given action is not needed.

Transaction execution semantic follows a classical "all or nothing" rule. A **TransactionRequest** primitive is invoked by the sender and specifies the actions that should be executed by the receiver. The receiver invokes a **TransactionAccept** primitive to indicate the successful completion of *all* actions specified by the TransactionRequest or a **TransactionReject** primitive if *at least one* command related to a given action had failed. The receiver may invoke one or many **TransactionPending** primitives to indicate to the sender that one transaction is actively being processed but has not been completed.

Packages

The H.248 specification provides use case examples of the protocol application for the control of some specific categories of MG. However, the key foundation for the universal applicability of the protocol (i.e., an MGC product from manufacturer A to control an MG from manufacturer B without the need of a joint validation of the two products) seems to lie within the **Package extension**. Packages allow a manufacturer to equip an MG with specific capabilities for handling its terminations and "announce" these capabilities to any MGC through the H.248 Audit command.

G.2 ITU-T BICC protocol

In July 2001, ITU announced the completion of the second set of Bearer Independent Call Control (BICC) protocols. This agreement enables the transition of today's networks from general public switching systems to next-generation server based networks. Utilizing the latest techniques in network design, built on a server, proxy and media gateway network architecture, BICC supports all currently deployed services on circuit-switched, ATM and IP, including third-generation wireless, yet not limiting the future introduction of innovative multimedia, multi-mode services and applications.

Annex H – Electronic Numbering (ENUM)²⁶

The ENUM standard, described in IETF's RFC2916, defines a protocol and an architecture based on the Internet domain name system (DNS), whereby it is possible to obtain a correspondence between E.164 telephone numbers and call service identifiers, with an order of priority (e-mail, website URL, SIP address of an IP telephony server, voice mail, other telephone numbers, etc.) Using the ENUM protocol, therefore, it is possible to find the various addresses of the target user on the basis of a simple telephone number. The end user may also customize the manner in which he or she may be reached, with a single E.164 number. It is easy to add to or modify such additional information without changing the number used for access. The ENUM protocol is thus seen as a technical gateway ensuring correspondence between the Internet and the switched telecommunication network, enabling interworking between the two.

The ENUM protocol and the use of DNS mechanisms do not give rise to any technical problems implying the incorrect functioning of services based on this functionality. However, the ability of the centralized and hierarchy-based DNS architecture to support requests generated by services requiring the transport of information in real time and with high quality will need to be determined at each level of the DNS architecture, according to the load and to the level of availability required by each service.

To find the DNS name of an E.164 telephone number, the RFC2916 standard requires that the following steps be executed:

Table H.1 – Finding the Domain Name System

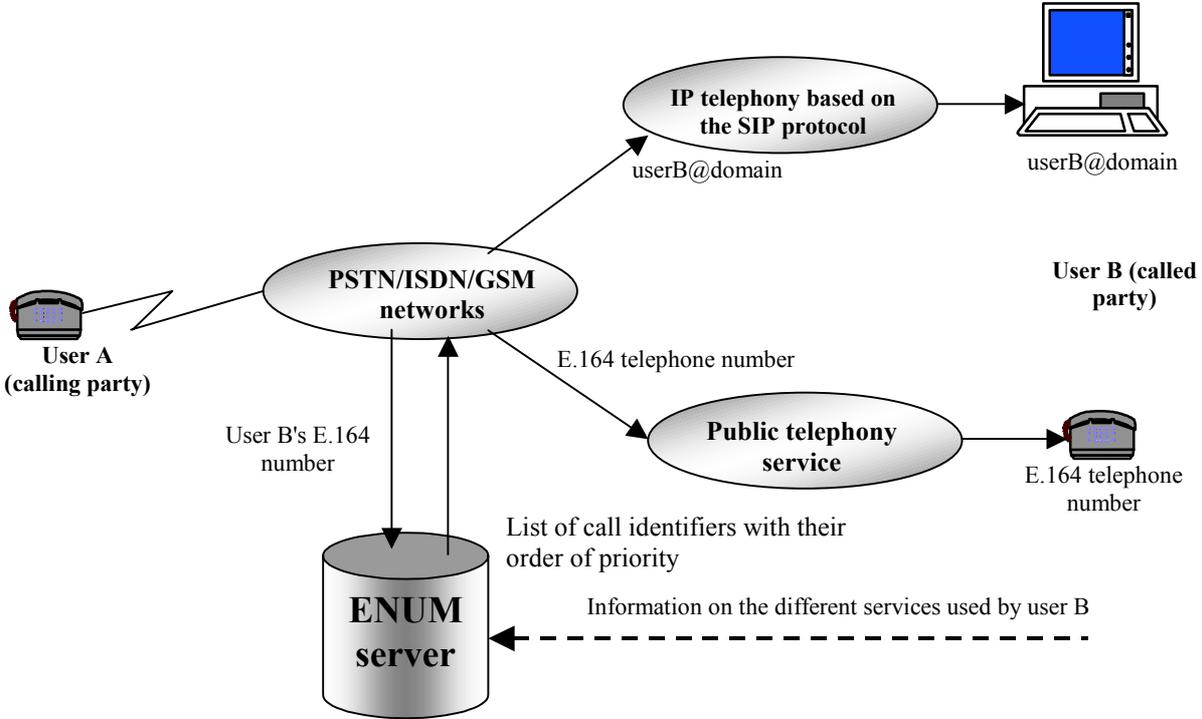
	Step	Example
1	Write the E.164 number in its complete form with the country code (IDDD)	+46-8-9761234
2	Delete all non-numeric characters except for the symbol "+"	+4689761234
3	Remove all non-numeric characters	4689761234
4	Insert a stop (".") between each digit of the number	4.6.8.9.7.6.1.2.3.4
5	Reverse the order of the digits of the number	4.3.2.1.6.7.9.8.6.4
6	Add the string ".e164.arpa" to the end of the number obtained in step 5	4.3.2.1.6.7.9.8.6.4.e164.arpa

Using the domain name obtained in the last step of the above procedure, the ENUM algorithm is applied in order to obtain the order of priority of the call service identifiers. The following figure shows an example of the ENUM protocol in the case of use of a conventional (analogue) telephone set via the switched network.

²⁶ For more information, please consult the following Web pages:

- on the work of IETF: <http://www.ietf.org/html.charters/enum-charter.html>;
- on the work of ITU-T Study Group 2: <http://www.itu.int/ITU-T/com2/index.html>.

Figure H.1 – Example of ENUM usage in the case of the switched telephone network



Annex I – Abbreviations

ADM	Adaptive Delta Modulation
ADPCM	Adaptive Differential Pulse Code Modulation
ADSL	Asymmetric Digital Subscriber Line
AF	Assured Forwarding
AGW	Access Gateway
AH	Authentication Header
AN	Access Node
ApGW	Application Gateway
API	Application Programming Interface
ARIB	Association of Radio Industries And Businesses
AS	Application Server
ATM	Asynchronous Transfer Mode
BAS	Broadband Access Server
BICC	Bearer Independent Call Control
BW	Bandwidth
CAPEX	Capital Expenditure
CDR	Call Detail Records
CELP	Code Excited Linear Prediction
CLNP	Connectionless Network Protocol
CME	Circuit Multiplication Equipment
CPE	Customer Premises Equipment
CPU	Central Processing Unit
CS-ACELP	Conjugate-Structure Algebraic-Code-Excited Linear Prediction
DAT	Digital Audio Tape
DiffServ	Differentiated Services Protocole
DLCI	Data Link Connection Identifier
DNS	Domain Name System
DPCM	Differential Pulse Code Modulation
DSCP	DiffServ Code Point
DSL	Digital Subscriber Line
DSLAM	Digital Subscriber Line Access Multiplexer
DTR	Data Transfer Rate
EF	Expedited Forwarding

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ENUM	Electronic Numbering
ESP	Encapsulating Security Payload
FDD	Frequency Division Duplex
FDDI	Fibre Distributed Data Interface
FEC	Forward Equivalence Class
FTP	File Transfer Protocol
GPRS	General Packet Radio Service
GSM	Global System for Mobile communication
HTTP	Hyper Text Transfer Protocol
ICMP	Internet Control Message Protocol
ICV	Integrity Check Value
IDD	International Direct Dialling
IETF	Internet Engineering Task Force
IKE	Internet Key Exchange
IN	Intelligent Network
INAP	Intelligent Network Application Part
IntServ	Integrated Services
IP	Internet Protocol/Internetworking Protocol
IPSec	IP Security Protocols
IPTN	IP telephony Network
ISAKMP	Internet Security Association and Key Management Protocol
ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
ISR	International Simple Resale
ISUP	ISDN User Part
ITSP	Internet Telephony Service Provider
JAIN	Java API for Integrated Networks
LAN	Local Area Network
LDAP	Lightweight Directory Access Protocol
LD-CELP	Low Delay-Code Excited Linear Prediction
LDP	Label Distribution Protocol
LER	Label Edge Router
LEX	Local Exchange
LIB	Label Information Base
LMDS	Local Multipoint Distribution System
LPC	Linear Predictive Coding
LSP	Label Switched Path

LSR	Label Switching Router
MC	Multipoint Controller
MCU	Multipoint Conference Unit
MGC	Media Gateway Controller
MGCP	Media Gateway Control Protocol
MIPS	Million Instructions Per Second
MOS	Mean Opinion Score
MP	Multipoint Processor
MPEG	Motion Picture Expert Group
MPLS	Multi Protocol Label Switching
MP-MLQ	Multi-Pulse Maximum Likelihood Quantization
NAS	Network Access Server
NAT	Network Address Translation
NGN	Next-Generation Network
NT	Network Termination
OPEX	Operational Expenditure
OSA	Open Service Access
OSI	Open System Interconnection
OSPF	Open Shortest Path First
PABX	Private Automatic Branch Exchange
PBX	Private Branch Exchange
PC	Personal Computer
PCM	Pulse Code Modulation
PDA	Personal Digital Assistant
PGP	Pretty Good Privacy
PHB	Per Hop Behaviour
PINT	Particle Impact Noise Test
PKI	Public Key Infrastructure
PLMN	Public Land Mobile Network
POI	Point of Interconnection
POTS	Plain Old Telephony Service
PPP	Point-to-Point Protocol
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RGW	Residential Gateway
RPE-LTP	Regular Pulse Excitation/Linear Predictive Coding Using Long-Term Prediction
RSVP	Resource ReSerVation Protocol
RTCP	Real-time Transport Control Protocol

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RTP	Real-time Transport Protocol
SCCP	Signalling Connection Control Part
SCN	Switched Circuit Network
SCP	Service Control Point
SCTP	Signalling Connection Transfer Protocol
SDH	Synchronous Digital Hierarchy
SIGTRAN	Signalling Transport
SIP	Session Initiation Protocol
SKEME	Secure Key Exchange Mechanism
SLA	Service Level Agreement
SLS	Service Level Specification
SMTP	Simple Mail Transfer Protocol
SNMP	Simple Network Management Protocol
SPIRITS	Service in the PSTN/IN Requesting InTernet Service
SS7	Signalling System No. 7
STP	Signalling Transfer Point
TCAP	Transaction Capabilities Application Part
TCP	Transmission Control Protocol
TDD	Time Division Duplex
TDM	Time Division Multiplexing
TEX	Transit Exchange
TGW	Trunking Gateway
TLL	Time to Live
TLS	Transmission Level Security
UDP	User Datagram Protocol
URI	Universal Resource Identification
UTRA	Universal Terrestrial Radio Access
VCI	Virtual Circuit Identifier
VoDSL	Voice over Digital Subscriber Line
VoIP	Voice over IP
VoP	Voice over Packet
VPI	Virtual Path Identifier
VPN	Virtual Private Network
WAIS	Wide Area Information Server
WAN	Wide Area Network
WWW	World Wide Web

Annex J – Questions and issues for further consideration²⁷

J.1 There is a general consensus today that offering end users broadband access to IP-based networks is essentially a driver for offering new services/applications – that often have yet to be exactly determined though many ideas about them abound in the literature – that go beyond the basic telephony or Internet access. There are still difficulties facing a massive deployment of broadband access even within developed countries.

J.2 Providing access to the telephony network has always been associated with the revenue that is drawn from it through selling a network telephony service. The difficulty facing operators that deploy broadband access – when they are not subsidized by a public authority or cross-subsidized by a lucrative legacy telephony operation – is that their business model is unsustainable – and many of them went bust recently – if they stick to the ISP model of provisioning only a basic access service without much revenue drawn from network services.

J.3 The above economic difficulties raise the following fundamental technical issue that is still open today: will IP telephony – and later future multimedia communication service – be only an application (Internet model, i.e., without the network active involvement) or a network service like what is happening today in telephony networks? And, if the network will have to be involved in provisioning IP telephony as a service, then to what extent will it have to draw from the technical principles used by the legacy telephony networks and how will such principles be implemented?

J.4 It is recognized that developing countries are suffering not only from a lack of broadband access but also from a lack of basic access to any telephony network. Moreover, IP telephony along with other factors may be decreasing the revenues of their operators, and ISPs are affected by rates for connectivity to IP backbones due, amongst other things, to the volumes of traffic they draw into their network.

J.5 There is no magical technical solution to the above hurdles. There is no question that IP backbones should be deployed by those countries to support the increasing amount of data traffic to the Internet. Still, the short/medium-term urgency is to improve the access to the network at large whether over legacy narrow-band or – preferably – broadband technology to trigger the virtuous cycle of increased revenues with lower and lower access costs. For this to happen, telephony – which is today the only known large-scale network service that brings revenue even in developed economies – should be offered as a network service irrespective of the technology supporting it (whether it is a legacy TDM or a new IP packet transport).

J.6 For telephony to be offered as a network *service* over an IP transport (i.e., IP telephony), there needs to be a change from the way current IP telephony *applications* are offered over the Internet. The above chapters highlighted some of the various issues facing IP telephony to become a reliable network service; they all share a common technical denominator: there need to be network elements that are aware of a service object (i.e., a call) that is at a given point of time provided to an end user with each involved subnetwork providing its contribution to the overall end-to-end service completion and held responsible for it.

J.7 In order to achieve the above objective, and depending on each specific situation, one sensible approach is to consolidate/expand existing telephony networks with future safe equipment and smoothly migrate those networks towards a next-generation network based on a converged IP transport used for both data and voice/multimedia services.

²⁷ This series of "Questions and issues for further consideration" reflects proposals made by experts at the third and last meeting to stimulate discussions, but does not represent a consensus of all experts present at that meeting.

Annex K – "IP telephony" related standards work

A compendium of Internet protocol (IP) telephony related standards work mapped to the WTPF-01 Opinion D Group of Experts outline of technical issues

This document is a compendium of work that has been completed or is being carried out by various ITU-T Study Groups (SG) and other standards development organizations (SDOs), relative to IP telephony technical issues addressed by the Group of Experts. This document is not meant to be comprehensive; however, it provides information about such technical issues that can be used as a resource to aid those implementing a variety of IP telephony technical scenarios. Note that this is a compendium of industry references received by September 2001.

Because of the vast number of regional and international organizations working on IP telephony technical issues, the Group of Experts invites ITU Member States, Sector Members, SDOs and other relevant entities to contribute to this dynamic document by providing input as to where technical work is being done. This in turn may facilitate the compilation of a comprehensive listing.

The information contained in the following pages was contributed by:

- ITU-T Study Group (SG) Vice-Chairmen/Working Party Chairs/Vice-Chairs
- Telecommunications Industry Association (TIA) Engineering Committees and Subcommittees (<http://www.tiaonline.org/standards/> and <http://www.tiaonline.org/standards/ip/>)²⁸
- Committee T1 (T1) Technical Subcommittees (<http://www.t1.org/html/org.htm> and <http://www.t1.org/html/standard.htm>)²⁸
- Internet Engineering Task Force (IETF) (<http://www.ietf.org/html.charters/wg-dir.html>)
- Other comments received from U.S.-based industry, and ITU U.S. National Telecommunications and Information Administration Institute of Telecommunications Institute (NTIA/ITS)²⁸
- ITU-T Website and associated documents (<http://www.itu.int/ITU-T/>)

Third-Generation Partnership Projects (3GPPs):

The 3GPPs embody the benefits of a collaborative effort, while at the same time benefiting from recognition as emanating from a specifications-developing body; they provide easier access of their outputs into ITU after transposition of the specifications in a Standards Development Organization (SDO) into a standard and submittal via the national process, as applicable, to ITU.

3GPP: Develops globally applicable Technical Specifications (TS) and Technical Reports (TR) for a 3G mobile system based on evolved Global System for Mobile communication (GSM) core networks and the radio access technologies that they support (i.e., Universal Terrestrial Radio Access (UTRA), Frequency Division Duplex (FDD) and Time Division Duplex (TDD) modes). The partners have further agreed to cooperate in the maintenance and development of GSM TSs and TRs, including evolved radio access technologies (e.g. General Packet Radio Service (GPRS) and Enhanced Data rates for GSM Evolution (EDGE)).

3GPP2: Develops global specifications for ANSI/TIA/EIA-41, "Cellular Radiotelecommunication Intersystem Operations" network evolution to 3G, and global specifications for the radio transmission technologies (RTTs) supported by ANSI/TIA/EIA-41. 3GPP2 was born out of the ITU IMT-2000 initiative, covering high-speed, broadband, and Internet Protocol (IP)-based mobile systems featuring network-to-network interconnection, feature/service transparency, global roaming and seamless services independent of location.

²⁸ Approved by ITU-T for Recommendations A.5, "Generic procedures for including references to documents of other organizations in ITU-T Recommendations" and A.6, "Cooperation and exchange of information between ITU-T and national and regional standards development organizations" implementation.

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Relevant ITU-T study groups and website links:

SG 2, "Operational Aspects of Service Provision, Networks and Performance"

(<http://www.itu.int/ITU-T/studygroups/com02/index.html>)

SG 4, "Telecommunication Management, including TMN"

(<http://www.itu.int/ITU-T/studygroups/com04/index.html>)

SG 7, "Data Networks and Open System Communications"

(<http://www.itu.int/ITU-T/studygroups/com07/index.html>)

SG 9, "Integrated Broadband Cable Networks and Television and Sound Transmission"

(<http://www.itu.int/ITU-T/studygroups/com09/index.html>)

SG 11, "Signalling Requirements and Protocols" [Lead SG on Intelligent Networks]

(<http://www.itu.int/ITU-T/studygroups/com11/index.html>)

SG 12, "End-to-End Transmission Performance of Networks and Terminals"

(<http://www.itu.int/ITU-T/studygroups/com12/index.html>)

SG 13, "Multi-Protocol and IP-based Networks and their Internetworking"

(<http://www.itu.int/ITU-T/studygroups/com13/index.html>)

Associated document: ITU-T SG 13 IP Project Description Version 5

(<http://www.itu.int/ITU-T/studygroups/com13/ip/documents/IPProjectLastVersion5.doc>)

Associated SG 13-led Activity: Global Information Infrastructure (GII)

(<http://www.itu.int/ITU-T/studygroups/com13/gii/index.html>)

Organization of the IP-related ITU-T Recommendations Y.1000 to Y.1800-series

(<http://www.itu.int/ITU-T/studygroups/com13/ip/y1000series.html>)

The complete list of Y-series Recommendations

(<http://www.itu.int/rec/recommendation.asp?lang=e&type=products&parent=T-REC-Y>)

SG 16, "Multimedia Services, Systems and Terminals"

(<http://www.itu.int/ITU-T/studygroups/com16/index.html>)

Associated document: ITU-T SG 16 Mediacom 2004 Project Description

(<http://www.itu.int/ITU-T/studygroups/com16/mediacom2004/index.html>)

Special Study Group, "IMT-2000 and Beyond" [Network Aspects]

(<http://www.itu.int/ITU-T/studygroups/ssg/index.html>)

ITU-T IMT-2000 and Network Aspects Project

(<http://www.itu.int/ITU-T/imt-2000/index.html>)

ITU-D link:

Guide on Managing and Developing Network Connections and Interconnections to National Internet Nodes (informational inclusion)

(<http://www.itu.int/itu-d/bdtint/Publications/administ.htm>)

1 Working definitions of IP telephony

ITU-T SG 13 Q.15, "General Network Terminology Including IP Aspects"

American National Standard (ANS) T1.523, "Telecom Glossary 2000". This Web-based, hyperlinked document provides authoritative definitions for over 8000 telecommunication terms (<http://www.its.bldrdoc.gov/projects/telecomglossary2000/>). ITU-T SG 13 Q.15 is proposing this glossary for inclusion in an existing ITU-T Recommendation.

TIA Engineering Subcommittee (TR)-41.4, "IP telephony Gateways and Infrastructure". [Project Number (PN)-3-4601] "IP telephony Gateways and Related Control Infrastructure" Terminology related to IP telephony is being collected and defined where necessary under this project. Results were due to be published as a Telecommunications Systems Bulletin (TSB), 5/2002.

2 Network architectures

ITU-T SG 13 (IP-based architectures) and SG 16 (related multimedia-based architectures).

ITU-T SG 9 IPCablecom Project (<http://www.itu.int/itudoc/itu-t/com9/ipcable/index.html>).

ITU-T SG 16 Q.B, "Multimedia Architecture".

Industry Note: Another emerging architecture involves IP-based optical systems.

ITU-T (IP-Related) Y-1200 Series Recommendations.

TIA TR-34.1, "Communications and Interoperability [Satellite-oriented]". Currently, there is an effort to create a WG on IP QoS architectures for satellite networks.

TIA TR-45.2, "Wireless Intersystem Technology". Regarding IP Core Network-Legacy MS Support Domain- (PN-4934) to be published as TIA/EIA/IS-872, "IP Core Network-Legacy MS Support Domain" and IP Core Network-Multimedia Domain- (PN-4935) to be published as TIA/EIA/IS-873, "IP Core Network-Multimedia Domain". Scheduled for ballot approval late 2001.

IETF IP telephony (iptel) WG: Develops solutions that involve problems related to propagation of routing information for VoIP protocols (i.e., SIP and H.323). The iptel group has defined and is continuing to refine a Telephony Routing over IP (TRIP) protocol. Other work involves Language for User Control of Internet Telephony Services, Management Information Base for (TRIP), Call Processing Language Framework and Requirements (RFC 2824) and a Framework for a Gateway Location Protocol (RFC 2871).

2a Current architectures

ITU-T SG 16 Q.2, "Multimedia over packet networks using H.323 systems".

TIA TR-45.6, "Adjunct Wireless Packet Data Technology". Developed the following TSB and Standards for CDMA2000 Packet Data Network Architectures: TIA/EIA/TSB115, "cdma2000 Wireless IP Architecture based on IETF Protocols", December 2000; TIA/EIA/IS-835, "cdma2000 Wireless IP Network Standard", December 2000; TIA/EIA/IS-835-A, "cdma2000 Wireless IP Network Standard – Revision A", 5/2001.

[TIA TR-45.6 developed CDPD Data Network Architecture] TIA/EIA-732, "Cellular Digital Packet Data (CDPD) System Specification". Published 3Q2001.

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2b Open network architecture

ITU-T SG 13 Q.1, "Principles, Requirements, Frameworks, and Architectures for an Overall Heterogeneous Network Environment".

2c Network capabilities

ITU-T SG 13 and SG 16: Recommendations (i.e., H-series) involving IP-based networks, capabilities and interworking.

TIA TR-41, "User Premises Telecommunications Equipment Requirements". Addresses voluntary standards for telecommunications terminal equipment and systems, and networks, specifically those used for voice service, integrated voice and data service and Internet Protocol (IP) applications. Also covers wireless user premises equipment and private network mobility; including service and performance criteria for equipment, systems and networks, as well as the information necessary to ensure their proper interworking with each other, with public networks, with IP telephony infrastructures and with carrier-provided private-line services.

TIA TSB-122-A, "Telecommunications – IP telephony Equipment – Voice Gateway Loss and Level Plan Guidelines", [TR-41] was published July 2000 and covers the transmission (loss and level) requirements for voice gateways. For the purposes of this TSB, a voice gateway is considered to be a device that routes voice between telephones, between telephones and public network trunks or private trunks and between trunks (public network and private trunks). The telephones considered in this TSB consist of two different types: analog telephones assumed to be compatible with the parameters specified in ANSI/EIA/TIA-470-B and digital telephones assumed to be compatible with the parameters specified in ANSI/TIA/EIA-810-A.

TIA TR-41.4: [PN-3-4826] "IP telephony Voice Gateway Transmission Requirements". This document will define the transmission requirements for IP-based voice gateways. It extends the information in TSB-122-A to include other parameters such as frequency response, distortion, noise, delay, and packet loss. It also will provide half-channel loss plan requirements that will allow interworking with European IP voice gateways. The document will be published as a TIA standard and is expected to be ready for ballot after 11/ 2001. ETSI TC "Speech Processing, Transmission and Quality (STQ)" and TIA are developing an equivalent harmonized half-channel loss plan standard.

TIA TR-45.5, "Wideband Spread Spectrum Digital Technology". Completed definitions of two IP-based packet data services for IMT-2000, including TIA/EIA/IS-707-A, "Data Service Options for Wideband Spread Spectrum Systems" and TIA/EIA/IS-707-B (soon to be published), "Data Services Option Standard for Wideband Spread Spectrum Systems".

TIA TR-45.5: Involving TIA/EIA/IS-cdma2000-A, "cdma2000 Series", service option 33, the underlying Radio Link Protocol (RLP), has been extended to support all rates and frame sizes defined in the cdma2000 physical layer with support also added for all physical layer frame sizes of the MAC layer. Involving TIA/EIA/IS-856, "cdma2000 High Rate Packet Data Air Interface Specification", the cdma2000 technology has been enhanced with 1xEV-DO, which is specifically designed to maximize a cdma2000 system's IP-based packet data throughput and capacity.

ETSI Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Project: Addresses voice communication and related voiceband communication (such as facsimile) between users, helping to ensure that users connected to IP-based networks can communicate with users in Switched-Circuit Networks (SCN) such as PSTN/ISDN and GSM, and vice versa. Related work involves service interoperability, architecture, call control, numbering/addressing, verification and mobility aspects (<http://www.etsi.org/tiphon/>).

IETF Media Gateway Control (MEGACO) WG: Addresses architectures and requirements for controlling Media Gateways from external control elements such as a Media Gateway Controller. A media gateway is a network element that provides conversion between the information carried on telephone circuits and data packets carried over the Internet or over other IP networks.

3 Quality of Service issues

ITU-T SG 12, "End-to-End Transmission Performance of Networks and Terminals". Lead ITU-T SG on QoS.

ITU-T SG 12 Q.10 [General planning]: "Transmission Planning for Voiceband, Data and Multimedia Services".

ITU-T SG 12 Q.12 [IP and Mixed PSTN/IP Planning]: "Transmission Performance Considerations for Voiceband Services Carried on Networks that Use Internet Protocol (IP)".

ITU-T SG 12 Q.13 [MM Focused QoS Planning]: "Multimedia QoS/Performance Requirements".

ITU-T SG 12 Q.14 [IP and Mixed PSTN/IP Planning]: "Effects of Interworking between Multiple IP Domains on the Transmission Performance of VoIP and Voiceband Services".

ITU-T SG 12 Q.15 [General QoS Coordination Aspects]: "QoS and Performance Coordination".

ITU-T SG 13: WP 4, "Network Performance and Resource Management". Q.4, "Broadband and IP related Resource Management"; Q.6, "Performance of IP-based Networks and the Emerging GII"; Q.9, "Call Processing Performance".

ITU-T SG 13 Q.4, "Broadband and IP Related Resource Management". Draft Recommendation Y.iptc, "Traffic Control and Congestion Control in IP-based Networks". Describes traffic control and congestion control mechanisms needed to support services with QoS, where the QoS is negotiated between a user and the network. Specified IP transfer capabilities including, for each IPTC, the service model, the associated traffic patterns, and conformance definition.

ITU-T SG 13 Q.6, "Performance of IP-Based Networks and the Emerging GII". Recommendation Y.1540 (ex. I.380, "Internet Protocol Data Communication Service – IP Packet Transfer and Availability Performance Parameters"), "IP Packet Transfer and Availability Performance Parameters". Defines parameters that may be used in specifying and assessing the speed, accuracy, dependability and availability of IP packet transfer in international IP data communication service.

ITU-T SG 13 Q.6, "Performance of IP-Based Networks and the Emerging GII". Draft Recommendation Y.1541, "IP Performance Objectives and Allocations". Specifies provisional objectives for Internet Protocol network performance parameters and Quality of Service (QoS) Classes.

ITU-T SG 13 Q.9, "Call Processing Performance". Draft Recommendation Y.1530, "Call Processing Performance for Voice Service In Hybrid IP Networks". Defines performance parameters and objectives and QoS Classes for point-to-point call processing in voice service for hybrid IP networks.

ITU-T SG 16 Q.F, "Quality of Service and End-to-end Performance in Multimedia Systems".

ITU-T IP-related Y.1500 Series Recommendations.

Committee T1 Technical Subcommittee T1A1, "Performance and Signal Processing". Develops and recommends standards and technical reports (TR) related to the description of performance and the processing of speech, audio, data, image and video signals, and their multimedia integration.

Committee T1 Technical Subcommittee T1A1: ANS T1.520, "IP Packet Transfer and Availability Performance Parameters". Adopts ITU-T Recommendation I.380, "Internet Protocol Data Communication Service – IP Packet Transfer and Availability Performance Parameters".

Committee T1 Technical Subcommittee T1A1: ANS T1.522, "Quality of Service for Business Multimedia Conferencing". Defines QoS requirements for multimedia conferencing over IP-based networks.

Committee T1 Technical Subcommittee T1A1: T1 TR 70, "Reliability/Availability Framework for IP-Based Networks and Services".

ETSI/TIPHON WG 5, "Quality of Service Aspects".

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TIA TR-34.1: Proposed WG on "IP QoS Architectures for Satellite Networks".

TIA TR-34.1: With regard to Satellite VoIP, an "IP over Satellite" WG was created to examine performance issues.

TIA TR-45.2: (PN-4934) to be published as TIA/EIA/IS-872, "IP Core Network-Legacy MS Support Domain" and (PN-4935) to be published as TIA/EIA/IS-873, "IP Core Network-Multimedia Domain".

3a Issues of interworking with IP and PSTN

Committee T1 Technical Subcommittee T1A1: T1 TR 55, "Reliability and Survivability Aspects of the Interactions Between the Internet and Public Telecommunications Networks".

Committee T1 Technical Subcommittee T1A1: T1 TR 56, "Performance Guidelines for Voiceband Services over Hybrid Internet/PSTN Connections".

TIA TIA/EIA-810-A, "Telecommunications – Telephone Terminal Equipment – Transmission Requirements for Narrowband Voice over IP and Voice over PCM Digital Wireline Telephones", [TR-41] was published 12/2000 and establishes handset, headset and handsfree telephone audio performance requirements for digital wireline telephones regardless of protocol or digital format. This standard only addresses conventional narrowband performance, where narrowband is defined as the frequency range between 300 and 3400 Hz. Wideband telephony, in the frequency range between 150 and 6800 Hz, is an enhancement that is likely to be offered by VoIP telephones. The performance requirements of wideband telephony will be addressed in future TIA/EIA standards.

TIA TIA/EIA/IS – 811, "Telecommunications – Telephone Terminal Equipment – Performance and Interoperability Requirements for Voice-over-IP (VoIP) Feature Telephones", [TR-41] was published July 2000 and specifies requirements for the interoperability, acoustic performance, supplementary services support, safety, electromagnetic compatibility, and environmental performance of VoIP Feature Telephones. This standard applies specifically to VoIP Feature Telephones that are connected to Ethernet networks using category-rated (per ISO/IEC 11801) unshielded twisted pair wiring. This standard harmonizes, wherever possible, with IEEE 802.3 standards for Ethernet electrical/physical interface.

TIA TSB-116, "Telecommunications – IP Telephony Equipment – Voice Quality Recommendations for IP Telephony", [TR-41] was published 3/2001. The objectives of this TSB are to provide end-to-end voice quality guidelines for IP telephony (e.g., North American) and to provide an E-Model tutorial for IP scenarios. IP telephony introduces several impairments, some of which are familiar and some new. The E-Model (ITU-T Rec. G.107) is a tool that can estimate the end-to-end voice quality, taking the IP telephony parameters and impairments into account. This TSB first describes how the E-Model handles IP telephony impairments and then it provides general design recommendations for the best possible voice quality performance irrespective of cost, available technology or customer requirements. These recommendations include specific IP scenarios to provide an E-Model tutorial for analyzing real networks.

TIA TSB-122-A, "Telecommunications – IP Telephony Equipment – Voice Gateway Loss and Level Plan Guidelines"; including associated ANSI/EIA/TIA-470-B (analog telephones) and ANSI/TIA/EIA-810-A (digital telephones).

TIA TR-41.4: [PN-3-4462-URV] "Performance and Interoperability Requirements for Voice-over-IP (VoIP) Telephone Terminals". Project to revise TIA/EIA/IS-811 and upgrade it to ANS status. Target ballot 5/2002.

TIA TR-41.4: [PN-3-4826] "IP telephony Voice Gateway Transmission Requirements". Extends TIA TSB-122-A and allows for interworking with European IP voice gateways. ETSI/STQ and TIA are developing an equivalent harmonized half-channel loss plan standard.

TIA TR-34.1 WG, "Satellite IP (VoIP) over Satellite". Examines performance issues.

TIA TR-45.4, "Radio to Switching Technology". Specifications developed use mobile IP in their R-P (Radio-Packet) interfaces (e.g., A-interface) to the PDSN (Packet Data Switching Network). TIA/EIA/IS-2001, "Interoperability Specifications (IOS) for cdma2000 Access Network Interfaces", published 12/2000, supports the features corresponding to phase 0 of the 1X air interface TIA/EIA IS-2000, "cdma2000 Series", including an open packet data interface. In addition, a project is moving forward that would be an evolution of TIA/EIA/IS-2001 (supporting most Release A features), including simple PDSN selection, concurrent voice and data, short data bursts, inter-generation handoffs and packet data improvements. Another project scheduled to be a revision of TIA/EIA/IS-2001-A, features content for an evolution of the IOS V4.x series from 3GPP2 (completion scheduled for mid-2001). Specific features to be supported in IOS V4.2 include 1x EV/DO (may be independent of V4.2), end-to-end QoS and QoS negotiation, RAN OAM&P, support for common channel only devices and 128 kbit/s ISDN inter-working. Also note that IOS V5.0 will be developed in 3GPP2 and is planned to contain the ALL-IP phase 1.

ETSI TIPHON: Communication between users connected to IP-based networks and Switched Circuit.

IETF PSTN/Internet Interfaces (PINT) WG: Addresses connection arrangements through which Internet applications can request and enrich PSTN telephony services. An example of such services is a Web-based Yellow Pages service with the ability to initiate PSTN calls between customers and suppliers. Other work includes Service Support Transfer Protocol (SSTP), service management protocols and architectures, and protocols needed to support services in which a user of the Internet requests initiation of a telephone (i.e., PSTN-carried) call to a PSTN terminal (i.e., telephone, FAX machine).

IETF Services in the PSTN/IN Requesting Internet Services (SPIRITS) WG: Addresses how services supported by IP network entities can be started from IN (Intelligent Network) requests, as well as the protocol arrangements through which PSTN can request actions to be carried out in the IP network in response to events (IN Triggers) occurring within the PSTN/IN. SPIRITS concerns architecture and protocols for secure transport of IN trigger information (requests for actions, as well as plain event notifications, including parameters) from PSTN/IN to the IP network, and optional responses from the IP network back to the PSTN/IN.

IETF Signalling Transport (SIGTRAN) WG: Addresses the transport of packet-based PSTN signalling over IP Networks, taking into account functional and performance requirements of the PSTN signalling. For interworking with PSTN, IP networks will need to transport signalling such as Q.931 or SS7 ISUP messages between IP nodes such as a Signalling Gateway and Media Gateway Controller or Media Gateway.

IETF Media Gateway Control (MEGACO) WG: Addresses architectures and requirements for controlling Media Gateways from external control elements such as a Media Gateway Controller.

3b Interworking between techno islands:

i) Standards (IETF) like MPLS, DiffServ (Differentiated services – best effort in IP networks, QoS Forum.org) RSVP

ITU-T SG 11, "Signalling requirements and protocols", completed (7/2001) the second set of BICC protocols, which allows network operators to offer the complete set of PSTN/ISDN services, including all supplementary services, over a variety of data transport networks. In addition, BICC Capability Set 2 (CS 2) provides the network signalling protocols for third-generation wireless networks, enabling a transcoder-free call to be established ultimately leading to reduced cost while maintaining all the call's features.

IETF MPLS (MPLS) WG: Responsible for standardizing a base technology for using label switching and for the implementation of label-switched paths over various link-level technologies, such as Packet-over-Sonet, Frame Relay, ATM, and LAN technologies (e.g. all forms of Ethernet, Token Ring, etc.). This includes procedures and protocols for the distribution of labels between routers, encapsulations and multicast considerations.

IETF Differentiated Services (DiffServ) WG: Activity involves differentiated classes of service for Internet traffic, to support various types of applications and specific business requirements. Also involves traffic conditioners and Per-Domain Behaviours.

Resource Reservation Setup Protocol (RSVP) Concluded WG: Primary purpose was to evolve the RSVP specification and to introduce it into the Internet standards track.

ii) Real-time voice and video (RTP and RTCP) protocols

IETF Transport Area "Audio/Video Transport (AVT)" WG.

Industry Note: Audio and video coding standards at the RTP level are defined by means of payloads published by the IETF as RFCs. Some ITU and other SDO-based descriptions exist, however, as supplement to or in exact duplication of the IETF definitions (e.g. when the RFC number does not exist at the time of the approval of a Recommendation, see H.225.0 Annex F). The IETF provides under the AVT Working Group the RFCs after discussion of the specific proposals from the protocol aspects and does not distinguish *a priori* between proprietary and standard-based proposals. Example of SDOs contributing to that work include, besides ITU-T SG 16, 3GPP, 3GPP2, and MPEG.

TIA TSB-116, "Telecommunications – IP Telephony Equipment – Voice Quality Recommendations for IP Telephony." Involves E-Model (ITU-T Recommendation G.107) for end-to-end voice quality estimation.

IETF Session Initiation Protocol (SIP) WG: Chartered to continue the development of SIP, currently specified as proposed standard RFC 2543. SIP is a text-based protocol, similar to HTTP and SMTP, for initiating interactive communication sessions between users. Such sessions include voice, video, chat, interactive games, and virtual reality. The WG will concentrate on the specification of SIP and its extensions and will not explore the use of SIP for specific environments or applications.

Multiparty Multimedia Session Control (MMUSIC) WG: Develops Internet standards track protocols to support Internet teleconferencing sessions. MMUSIC's focus is on supporting the loosely controlled conferences that are pervasive on the Mbone today. However, the WG also will ensure that its protocols are general enough to be used in managing tightly controlled sessions. Related work areas include multimedia conferencing (e.g., AVT, RSVP) and the interoperability of this WG's output with other applicable standards (e.g., SIP/SAP/SDP with ITU H.323 and H.332).

4 Security issues

ITU-T SG 7 WP 3 Q.13, "Security services, mechanisms and protocols". SG 7 is Lead SG for Communications Systems Security.

ITU-T SG 16 Q.G, "Security of Multimedia Systems and Services".

TIA TR-45.2: Involved in IP transport-related activities, which has led to discussions with TR-45, "Mobile & Personal Communications Public 800 Standards" AHAG (Ad-Hoc Authentication Group), regarding security issues relative to IP-based transport.

IETF Security Area WGs (i.e., ipsec and ipsra).

4a Reliability

TIA TR-41.4: [PN-3-4462-URV] "Performance and Interoperability Requirements for Voice-over-IP (VoIP) Telephone Terminals". Project to revise TIA/EIA/IS-811 and upgrade it to ANS status.

4b Vulnerability

TIA TR-41.4: [PN-3-4462-URV] "Performance and Interoperability Requirements for Voice-over-IP (VoIP) Telephone Terminals". Project to revise TIA/EIA/IS-811 and upgrade it to ANS status.

5 Interoperability issues: Coexistence and evolution/migration strategies (technology availability...)

ITU-T SG 9 IPCablecom Project (<http://www.itu.int/itudoc/itu-t/com9/ipcable/index.html>).

ITU-T SG 13 Q.5, "Network Interworking including Multi-Service Networks".

ITU-T SG 13 Q.11, "Mechanisms to Allow IP-Based Services Using MPLS to Operate in Public Networks".

ITU-T SG 13 Q.14, "Access Architecture Principles and Features at the Lower Layers for IP-Based and Other Systems".

ITU-T SG 16 Q.D, "Interoperability of Multimedia Systems and Services".

ITU-T SG 16 Q.G, "Security of Multimedia Systems and Services".

ITU-T SG 16 Q.3, "Infrastructure and interoperability for Multimedia over packet networks".

ITU-T IP-related Y.1400 Series Recommendations.

TIA TR-45.2: Plans to standardize an IP-based transport mechanism in a future revision of TIA/EIA-41, "Cellular Radiotelecommunication Intersystem Operations" and is currently working on a project (PN-4762) to be published as TIA/EIA/IS-879, "Internet Protocol Based Data Transfer Services", regarding IP-based data transfer services for TIA/EIA-41. Planned approval in 3Q01.

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TIA TR-45.2: (PN-4934) to be published as TIA/EIA/IS-872, "IP Core Network-Legacy MS Support Domain" and (PN-4935) to be published as TIA/EIA/IS-873, "IP Core Network-Multimedia Domain".

TIA TR-45.3, "Time Division Digital Technology". An update of the TDMA Universal Wireless Communications (UWC-136) technology has been completed and will be published as TIA/EIA-136-Rev C, "TDMA Third Generation [3G] Wireless". TR-45.3 worked with ETSI to design the UWC-136 Radio Transmission Technology (RTT) data section, General Packet Radio Service (GPRS) backbone for IP traffic, based on ETSI's GSM specifications.

ETSI TIPHON: Communication between users connected to IP-based networks and Switched Circuit.

IETF SPIRITS WG: PSTN/IN to the IP network and optional responses from the IP network back to the PSTN/IN.

5a Traffic volumes (influence of IP traffic on fixed and mobile PSTN traffic), traffic engineering and dimensioning of IP networks

ITU-T SG 2 WP 1 Q.2, "Routing and interworking plans for fixed and mobile networks".

ITU-T SG 2 WP 3, "Traffic Engineering".

IETF Internet Traffic Engineering (TEWG) WG.

5b Pending items at ITU-T (especially ITU-T IP Project)

ITU-T SG 4 Q.10, "Framework for unified management of integrated circuit-switched and packet-based networks (with initial emphasis on IP-based networks)".

ITU-T SG 13 IP Project Description Version 5 (<http://www.itu.int/ITU-T/studygroups/com13/ip/documents/IPProjectLastVersion5.doc>).

Scope of ITU-T IP Project: 1) IP Applications Including Multi media: Q.3/13, SG 10, SG 16. 2) Service Interworking: Qs 2/13, 5/13. 3) IP and non-IP Access: Qs 12/13, 14/13, SGs 15 and 16. 4) Signalling Support: SG 11. 5) Network Capabilities: Qs 1, 2, and 3/13, SG 9. 6) Interworking: Q5/13, SG16 and SSG. 7) IP Performance: Q 6/13, SGs 2 and 12. 8) IP Transport: Qs 10/13, 16/13, SG 15. 9) Management: SG 4, 6/SSG, 4/15, 3/16. 10) Naming, Numbering, Addressing and Routing: SG 2. 11) Overall Network Architecture and Framework: Qs 1, 11, 14 and 16/13. 12) Operational Issues: SGs 2, 3, and 7.

ITU-T SG 13 List of Questions Under Study (<http://www.itu.int/ITU-T/studygroups/com13/questions.html>).

ITU-T SG 16 List of Questions under Study (<http://www.itu.int/ITU-T/studygroups/com16/questions.html>).

Organization of the IP-related ITU-T Recommendations Y.1000-series.

IP-based Networks Contribution originating from SG 13, to the Secretary-General's Report to the ITU Council (<http://www.itu.int/ITU-T/studygroups/com13/ip/documents/ip.doc>).

IETF WGs/ITU-T SGs Mapping (<http://www.itu.int/ITU-T/studygroups/com13/ip/ietf-wg.html>).

5c Examples of coexistence and migration:

i) IP platforms

TIA/EIA/IS – 811, "Telecommunications – Telephone Terminal Equipment – Performance and Interoperability Requirements for Voice-over-IP (VoIP) Feature Telephones". Harmonizes, wherever possible, with IEEE 802.3 standards for Ethernet electrical/physical interface.

TIA TR-41.4: [PN-3-4462-URV] "Performance and Interoperability Requirements for Voice-over-IP (VoIP) Telephone Terminals." Project to revise TIA/EIA/IS-811 and upgrade it to ANS status.

ETSI TIPHON: Communication between users connected to IP-based networks and Switched Circuit.

ii) Gateways

TIA TSB-122-A, "Telecommunications – IP telephony Equipment – Voice Gateway Loss and Level Plan Guidelines". Covers transmission (loss and level) requirements for voice gateways. This TSB consists of two different telephone types: analog telephones compatible with the parameters specified in ANSI/EIA/TIA-470-B and digital telephones compatible with ANSI/TIA/EIA-810-A.

TIA TR-41.4: [PN-3-4826] "IP telephony Voice Gateway Transmission Requirements". Extends TIA TSB-122-A and allows for interworking with European IP voice gateways. ETSI/STQ and TIA are developing an equivalent harmonized half-channel loss plan standard.

IETF IP telephony (IPTEL) WG: "Framework for a Gateway Location Protocol (RFC 2871)".

IETF Signalling Transport (SIGTRAN) WG: PSTN-IP networks transport signalling and messages between IP nodes such as a Signalling Gateway and Media Gateway Controller or Media Gateway.

IETF Media Gateway Control (MEGACO) WG: Addresses architectures and requirements for controlling Media Gateways from external control elements such as a Media Gateway Controller. Output includes Megaco/H.248 R2 Package, Megaco MIB, Megaco/H.248 NAS Package, Media Gateway Control Protocol Architecture and Requirements (RFC 2805), Megaco Protocol (with errata folded in) (RFC 3015) and Megaco IP Phone Media Gateway Application Profile (RFC 3054).

5d Protocol issues (towards IPV6, BICC...)

ITU-T SG 11: Completed (7/2001) BICC CS 2, which allows network operators to offer the complete set of PSTN/ISDN services, including all supplementary services, over a variety of data transport networks. Additionally, BICC CS 2 provides the network signalling protocols for third-generation wireless networks, enabling a transcoder-free call to be established ultimately leading to reduced cost while maintaining all the call's features. Also see BICC CS 1 and Q-Series Recommendations.

TIA/EIA/IS – 811, "Telecommunications – Telephone Terminal Equipment – Performance and Interoperability Requirements for Voice-over-IP (VoIP) Feature Telephones". Harmonizes, wherever possible, with IEEE 802.3 standards for Ethernet electrical/physical interface.

TIA TR-41.4: [PN-3-4462-URV] "Performance and Interoperability Requirements for Voice-over-IP (VoIP) Telephone Terminals". Project to revise TIA/EIA/IS-811 and upgrade it to ANS status.

IETF IPTEL, PINT, SIP and MMUSIC WG-related activities.

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5e Inner network (inter PSTN) issues

Includes several relevant ITU-T SGs previously mentioned, involving many aspects of inter-networking with the PSTN/ISDN.

TIA/EIA-810-A, "Telecommunications – Telephone Terminal Equipment – Transmission Requirements for Narrowband Voice over IP and Voice over PCM Digital Wireline Telephones". Performance requirements of wideband telephony will be addressed in a future TIA/EIA standard.

TIA/EIA/IS – 811, "Telecommunications – Telephone Terminal Equipment – Performance and Interoperability Requirements for Voice-over-IP (VoIP) Feature Telephones". Harmonizes, wherever possible, with IEEE 802.3 standards for Ethernet electrical/physical interface.

TIA TR-41.3, "Analog and Digital Wireline Terminals". [PN-3-4705] "Transmission Requirements for Wideband Digital Wireline Telephones" will produce a standard similar to TIA/EIA-810-A, defining requirements for the transmission performance of wideband telephones in the frequency range from 150 to 6800 Hz. It is expected the primary application for this standard will be in IP telephony. Expected completion in 2002.

TIA TR-41.4: [PN-3-4726] "Location Identification and Callback Procedures for IP Terminals" is a project that deals with the equivalent of E911 for IP terminals. Balloting is expected after 2/2002.

TIA TR-41.4: [PN-3-4462-URV] "Performance and Interoperability Requirements for Voice-over-IP (VoIP) Telephone Terminals". Project to revise TIA/EIA/IS-811 and upgrade it to ANS status.

5f Intra network (between PSTN/IP networks) e.g. interoperability of Info. Sys.

ITU-T SG 4 Q.10, "Framework for Unified Management of Integrated Circuit-Switched and Packet-Based Networks (with Initial Emphasis on IP-based Networks)".

5g Operations and maintenance of IP network

ITU-T SG 4 Q.10, "Framework for Unified Management of Integrated Circuit-Switched and Packet-Based Networks (with Initial Emphasis on IP-based Networks)".

ITU-T SG 13 Q.3, "OAM and Network Management in IP-based and other Networks".

ITU-T IP-related Y.1700 Series Recommendations.

Committee T1 Technical Subcommittee T1M1, "Internetwork Operations, Administration, Maintenance & Provisioning (IOAM&P)". Develops internetwork operations, administration, maintenance and provisioning standards, and technical reports related to network interfaces. Also involved in aspects of internetwork operations such as network management; circuit and facility installation, line-up, restoration, routine maintenance, fault location and repair; contact points for internetwork operations; and service evaluation.

IETF: Operations and Management Area WGs.

5h Interconnect charging systems (interoperator)

ITU-T IP-related Y.1800 Series Recommendations.

6 Applications and services

ITU-T SG 16 Q.C, "Multimedia applications and services".
 ITU-T SG 16 Q.H, "Accessibility to Multimedia Systems and Services".
 ITU-T IP-related Y.1100 Series Recommendations.

7 Coding

International Organization for Standardization/International Electrotechnical Commission Joint Technical Committee 1 (ISO/IEC JTC1) / SC29, "Coding of Audio, Picture, Multimedia and Hypermedia Information". Involves standardization of the coded representation, of audio, picture, multimedia and hypermedia information, and sets of compression and control functions for use with such information [e.g., MPEG in WG 11 at <http://www.itscj.ipsj.or.jp/sc29/29w42911.htm>]. Note there is mutual interest between ITU-T SG16 and ISO/IEC JTC 1/SC 29 (e.g., WG 11) involving IP MPEG-4 video/audio over MPEG-4 Systems streams, and future video coding standardization.

Industry Note: For Internet telephony per se, MPEG/Audio may not be relevant, since their main thrust is broadcasting/streaming-type of applications, which is different from conversational applications such as IP telephony. Relevant players are ITU-T, ETSI & ARIB, 3GPP, TIA, 3GPP2.

TIA/EIA-810-A, "Telecommunications – Telephone Terminal Equipment – Transmission Requirements for Narrowband Voice over IP and Voice over PCM Digital Wireline Telephones". Performance requirements of wideband telephony will be addressed in a future TIA/EIA standards.

TIA TR-41.3: [PN-3-4705] "Transmission Requirements for Wideband Digital Wireline Telephones".

TIA TR-41.4: [PN-3-4462-URV] "Performance and Interoperability Requirements for Voice-over-IP (VoIP) Telephone Terminals". Project to revise TIA/EIA/IS-811 and upgrade it to ANS status.

IETF Multiparty Multimedia Session Control (MMUSIC) WG: Coding activities related to Internet teleconferencing sessions and controlled conferences.

7a Coding technologies used to compress/optimize bandwidths (ISO, IETF, SG9 and SDO...)

ITU-T G-series, "Transmission Systems and Media, Digital Systems and Networks" Recommendations.

Industry Note: See G.711, G.723.1, G.726, G.729/G.729A; G.722, G.722.1, and (very new) G.722.2 in ITU; AMR and AMR-WB in 3GPP; and (not finalized) SMV in 3GPP2. In ITU-T, the work is performed in SG 16 WP 3 "Media Coding", with relevant Questions 6 (Video), 7 (Wideband Speech), 8 (4 kbit/s speech), 9 (Variable rate speech).

Committee T1 Technical Subcommittee T1A1: ANS T1.521, "A Packet Loss Concealment Technique for Use with ITU-T Recommendation G.711". Adopted in ITU-T Recommendation G.711, "Pulse Code Modulation (PCM) of Voice Frequencies".

Committee T1 Technical Subcommittee T1A1: ANS T1.521A, "Packet Loss Concealment for Use with ITU-T Recommendation G.711, Annex B". Defines an alternative PLC technique.

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7b Convergence of coding (e.g. MPEG)

ITU-T SG 16 and related G.720-Series Recommendations.

Industry Note: In 3G, AMR is the mandatory speech codec. *De facto* coders are G.723.1 and G.729/G.729A. However, more frequently we see demand for higher bandwidth, e.g. G.726 or G.711.

Industry Note: MPEG may not be relevant as a convergence codec for IP telephony since it is not tailored for conversational applications. There is a dominance of MPEG/Audio for streaming applications. However, non-MPEG *de facto* (non-standard) schemes abound, e.g. Real Audio, Windows Media Player, QuickTime.

8 Accessibility

8a Access types

TIA/EIA/IS – 811, "Telecommunications – Telephone Terminal Equipment – Performance and Interoperability Requirements for Voice-over-IP (VoIP) Feature Telephones". Harmonizes, wherever possible, with IEEE 802.3 standards for Ethernet electrical/physical interface.

TIA TR-41.4: [PN-3-4462-URV] "Performance and Interoperability Requirements for Voice-over-IP (VoIP) Telephone Terminals". Project to revise TIA/EIA/IS-811 and upgrade it to ANS status.

IETF SPIRITS WG: Addresses access from PSTN/IN to the IP network, and optional responses from the IP network back to the PSTN/IN.

9 Numbering and addressing plans in IP networks for interworking with PSTN

IETF Internet-Draft "A Framework for ENUM Neutrality"

(<http://search.ietf.org/internet-drafts/draft-rutkowski-enum-neutrality-00.txt>).

ETSI TIPHON Project: Related work involving numbering/addressing and mobility aspects.

Committee T1 Technical Committees and TIA Engineering Committees involved in technically-related numbering/addressing issues.

[Potentially] Committee T1 Technical Subcommittee T1A1: T1 TR 55, "Reliability and Survivability Aspects of the Interactions Between the Internet and Public Telecommunications Networks".

9a ITU-T and the ENUM Initiative (with IETF)

IETF Telephone Number Mapping (ENUM) WG (<http://www.ietf.org/html.charters/enum-charter.html>).

Annex L – Opinion A of WTPF-01

The general implications of IP Telephony for the ITU Membership with respect to:

- a) the telecommunications policies and regulations of ITU Member States;
- b) the implications of IP Telephony for developing countries, particularly with respect to policies and regulatory frameworks, as well as technical and economic aspects;
- c) the impact of IP Telephony on the operations of Sector Members, notably in terms of the financial challenges and commercial opportunities it presents.

The third World Telecommunication Policy Forum (Geneva, 2001),

considering

that, pursuant to the basic provisions of the ITU Constitution, that each member State has the sovereign right to develop policies related to telecommunications to meet its needs and objectives, and that the purposes of the Union include:

- to maintain and extend international cooperation among all members of the Union for the improvement and rational use of telecommunications of all kinds;
- to promote the development of technical facilities and their most efficient operation with a view to improving the efficiency of telecommunication services, increasing their usefulness and making them, so far as possible, generally available to the public;
- to promote the extension of the benefits of the new telecommunication technologies to all the world's inhabitants;
- to facilitate the worldwide standardization of telecommunications, with a satisfactory quality of service;
- to foster collaboration among Member States and Sector Members with a view to the establishment of rates at levels as low as possible consistent with an efficient service and taking into account the necessity for maintaining independent financial administration of telecommunications on a sound basis,

recognizing (broader economic implications for a country)

- a) that the deployment of IP-based networks and applications has the potential to benefit users, industries, and the economy at large, because it fosters technical and market innovation, and diversity and growth in the economy; not forgetting the potential of upgrading of existing communication networks through use of other new technologies;
- b) that these new enhanced communication capabilities may be essential for the development of other service sectors, and for the production and distribution of goods in the global economy as a whole;
- c) that IP-based applications are likely to become more readily available at cost-effective prices, for the benefit of all users and industries, particularly when supplied under competitive market conditions in which multiple, alternative sources or means are available to address user and industry needs;
- d) that IP Telephony and other IP-based applications could be viewed as a significant opportunity for all countries to respond to the convergence of information and communication technologies and evolve their networks in order to expand the availability and use of a broader range of modern communication capabilities,

noting (implications for operators)

- a) the continued development of the Internet and IP-based networks as a significant medium for communications and commerce;

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- b) that the flexibility of IP technologies will contribute to an integration of voice and data networks, thereby allowing suppliers to take advantage of synergies and possible cost reductions, which will enable the provision of new innovative services and applications for the benefit of all citizens;
- c) that networks that can support IP Telephony are being designed with a variety of core network and access technologies and capabilities, including wireless technologies;
- d) that mobile wireless systems are expected to migrate towards an IP-based architecture in order to deliver integrated voice, data and multimedia services, as well as access to the Internet;
- e) that Sector Members face both challenges and opportunities during this transition to a market-driven industry,

conscious (implications for government policies and regulation)

- a) that Member States pursue policies that seek to:
 - i) attract capital investment so as to fund infrastructure that serves users and society as a whole;
 - ii) stimulate innovation in order that applications and products that meet the needs of people can be offered in the marketplace;
 - iii) exploit the synergies between capital investment and innovation so as to promote sustainable economic development that can attract further investment and create the environment necessary to promote more innovation;
- b) that Member States have national public policy goals in the telecommunication sector, including universal access and service, competitive markets, technology innovation and transfer of technical know-how, and the development of human resources; in addition public interest goals (for example, access to emergency services, security and privacy);
- c) that IP Telephony has created some negative impacts on voice revenues generated by a number of communication operators, particularly in some developing countries; there could also be a revenue gain for other communication operators and service providers;
- d) that the dynamic growth of IP-based networks, applications and services has been due to a combination of private and public sector investment and innovation, and an effective competitive environment;
- e) that initiatives and policies dealing with IP-based networks would benefit from input from users (consumers and business organizations),

is of the view

- a) that IP Telephony applications are best supplied in a market in which consumers have choices among multiple, alternative sources or means to address their needs, because only then will citizens, businesses and the overall economy reap the benefits of innovation and cost effectiveness;
- b) that government regulation should aim to foster an effective competitive environment and that regulation may be appropriate where there is market failure or when public interests cannot be adequately met by industry (e.g. universal access and service); for some countries, there may be other reasons for regulators to intervene, for example to ensure the rebalancing of tariffs;
- c) that Member States should examine the implications of applying existing regulatory regimes to IP-based services and applications,

invites

- 1 Member States and Sector Members to consider the possibility of the introduction and deployment of IP technologies and IP applications, including the exchange of information;

- 2 all Member States to review their current regulatory frameworks with a view to:
- i) encouraging investment, spurring innovation and advancing development;
 - ii) achieving public policy goals in the context of a converged communication services environment;
 - iii) considering the possibility of opening their communication services market with respect to IP Telephony by adopting a competition-oriented approach in order to achieve clearly defined public policy goals, taking into account, among other things, the concept of technology neutrality for fully-substitutable services.

Annex M – Opinion B of WTPF-01

Actions to assist Member States and Sector Members in adapting to the changes in the telecommunication environment due to the emergence of IP Telephony, including analysing the current situation (e.g. by case studies) and formulating possible cooperative actions involving Member States and Sector Members to facilitate adaptation to the new environment

The third World Telecommunication Policy Forum (Geneva, 2001),

considering

that, for several years, technology analysts have observed a tendency for telephony services and other forms of communications to converge and that, in recent years, IP appears to have emerged as one of the possible unifying platforms,

noting

- a) that some global telecommunications operators have announced that they are migrating their traffic to IP-based platforms;
- b) that the introduction of competition within the telecommunication market benefits the consumer;
- c) that ITU, in its service definitions, has been careful not to imply or specify any particular implementation technology,

conscious

- a) that increasing access to the Internet is a policy goal in some Member States;
- b) that emerging IP technologies offer opportunities for the development of new multimedia applications, including voice;
- c) that deployment of such technologies may offer particular benefits to small and medium-sized enterprises,

encourages Member States

to share experiences in developing new methodologies and approaches that recognize the market conditions of advanced technologies, such as IP Telephony, including, but not limited to:

- i) approaches towards making any sector-specific regulation technology-neutral;
- ii) the application of domestic competition laws as part of a pro-competitive policy designed to establish a level playing field;
- iii) establishing sustainable bases to generate financing for universal access/service,

invites the Secretary-General and the Directors of the Bureaux

1 to promote understanding of the potential benefits of IP-based technologies and IP applications and, within existing budgetary resources, to assist Member States and Sector Members, particularly in developing countries:

- i) by updating previous IP Telephony case studies and carrying out further country case studies, as required;

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- ii) by carrying out cost studies and establishing a process to assist members in performing cost-benefit analyses in order to plan for investment in converged telecommunication networks on IP platforms, on request;
- iii) by helping to attract investment and promoting the use of international lending, such as attractive, flexible, low-interest, long-term loan arrangements, and donor organization resources,

2 in the pursuit of the above, to conduct regional workshops in partnership with Member States and Sector Members, and concerned regional telecommunication organizations, complementing existing ITU activities, on the following basis:

- The workshops should provide forums on:
 - i) how telecommunication infrastructure build-out and the evolution of existing networks can be facilitated by deployment of IP-based technologies;
 - ii) technologies that can support IP Telephony;
 - iii) how to create an environment that will attract investment in infrastructure development;
 - iv) issues such as cost structures, pricing mechanisms, interconnection, numbering, quality of service and market considerations, etc.
- ITU may call upon voluntary contributions from Member States, Sector Members and other industry participants to support such activities.
- The workshops should bring together regulators, government officials responsible for economic development and telecommunication issues, incumbent and new entrant network operators, Internet Service Providers (ISPs), equipment suppliers, consumers and consumer organizations.

Annex N – Opinion C of WTPF-01

Actions to assist Member States and Sector Members in meeting the human resource development challenges presented by new telecommunications technologies such as IP Telephony, in particular skill shortages and the need for education, and technology transfer

The third ITU World Telecommunication Policy Forum (Geneva, 2001),

considering

a) that the purposes of the Union include promoting the extension of the benefits of new telecommunication technologies, encouraging continued participation by the private sector in telecommunication development, offering technical assistance in the field of telecommunications, and promoting the mobilization of the material, human and financial resources needed for implementation of telecommunication systems;

b) that Council-2000 requested that action be taken to assist Member States and Sector Members in meeting the human resources development challenges presented by new telecommunication technologies such as voice over IP,

recognizing

a) that economic, social, technical and regulatory environments are changing in the context of ongoing telecommunication and information technology developments;

b) that transition to an IP environment places new demands on the management and operation of Member States and Sector Members as well as other private entities;

c) that these new challenges need to be addressed by ITU, and in particular ITU-D/BDT, which plays a crucial role in assisting countries that are developing and building institutional, physical and organizational telecommunication capacity,

further recognizing

a) that the speed with which countries can extend the benefits of telecommunication technologies will depend on their ability to deploy skilled personnel who are able to meet the operational and policy challenges stemming from the new environment;

b) that, in today's environment of globalization, shortages of skilled personnel and the absence of comprehensive human resources policies are impeding the transition to a new IP environment in all countries;

c) that technology transfer from Sector Members and Member States may help reduce the knowledge gap, although the problem is exacerbated by the "brain-drain",

encourages Member States and Sector Members to take into account

that training and education of a broad range of people benefits individuals, the communications system, communities and the economy as a whole, in particular, training and education for staff members of small and medium-sized communication enterprises, and keeping gender perspectives in consideration;

invites ITU, particularly ITU-D

to facilitate the sharing of knowledge and views, by Member States and Sector Members, of global trends in all new technologies, in particular IP applications, including the activities of other standardization bodies, infrastructure development, IP-based services and applications, and regulatory activities and policies,

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invites ITU-D

- 1 to encourage Member States and Sector Members to create integrated human resources transition plans for new technologies, business operations and regulatory and policy activities;
- 2 to assist Member States and Sector Members in evaluating and identifying new and changing human resources requirements in order to meet the challenges of an evolving communications environment;
- 3 to draw upon existing BDT research and skilled personnel to identify:
 - i) HRD/HRM and training issues related to network evolution;
 - ii) HRD/HRM and training issues related to new technologies, including IP;
 - iii) skills leading to the creation of a business environment that will attract infrastructure investment,

invites ITU-T and ITU-R

to collaborate with and assist the Director of BDT in creating the technical training components of ITU-D workshops, seminars, training projects, and modules,

invites Sector Members

to suggest, initiate, and/or participate in ITU-D human resources development and infrastructure investment programmes,

invites Sector Members and Member States

to work with educational institutions, NGOs and other organizations in order to draw upon resources, studies and expertise and to collaborate in assisting countries with training, resource retention and other HRD/HRM issues,

invites the Secretary-General and the three Directors

- 1 to disseminate widely information on the urgent need for the United Nations and Member States to review and develop policies that lead to:
 - i) recognition of the widespread opportunities for people skilled in information and telecommunication technologies;
 - ii) widespread training and education of all levels of their citizens, taking account of human factor aspects, in all domains of information and telecommunications technologies;
- 2 to encourage Associates and in particular the academic community to become actively involved in knowledge sharing and skills development;
- 3 to develop virtual capabilities for global knowledge-sharing, training and skills development;
- 4 to coordinate with regional telecommunication entities in identifying and developing programmes to advance regional skills bases.

Annex O – Opinion D of WTPF-01

Essential studies by ITU to facilitate the introduction of "IP telephony", including inter-operability considerations and implications, when implementing "IP telephony" in association with the existing circuit-switched national and international telecommunication networks in developing countries

The third World Telecommunication Policy Forum (Geneva, 2001),

considering

- a) that important technical, economic and regulatory decisions are essential, by developing countries in order to introduce "IP telephony";
- b) that the ITU Sectors should undertake further studies to facilitate the introduction of "IP telephony" by analysing inter-operability and interconnection issues with existing circuit-switched telecommunication networks, especially in the developing countries in order for them to take the proper decisions in due course,

invites the three ITU Sectors

each, for matters within its competence, and based on contributions from ITU Member States and Sector Members, to initiate new studies or pursue current ones, and to present any findings as soon as possible, to facilitate the introduction of "IP telephony" on a global basis, particularly with respect to:

1. in ITU-R,

compatibility and inter-operability of radio access between IP networks and PSTNs,

2. in ITU-T,

- a) a clear working definition of "IP telephony" and "Internet telephony";
- b) whether, and to what extent:
 - i) to require compatibility of "IP telephony" with the existing international telephone service and not place additional requirements on existing circuit-switched international networks;
 - ii) compatibility should also include, but not be limited to, aspects of performance metrics, and other aspects as detailed in the relevant ITU-T Recommendations, in particular those on quality of service;
- c) whether, and to what extent:
 - i) to consider the possibilities of how "IP telephony" can be a part of national PSTNs;
 - ii) aspects of traffic identification and measurement need to be considered for interworking of IP-based networks and PSTN/ISDN;
- d) identifying the cost elements of international "IP connectivity" with respect to the introduction of "IP telephony",

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3. In ITU-D,

taking account of the need of developing countries to plan a strategy of migration for their networks into IP-based networks, establishing a group of experts especially from developing countries to carry out the following tasks:

- a) to prepare, as soon as possible, a checklist of Factors which developing countries may use in the process of accelerating the introduction of IP networks, thus facilitating the introduction of "IP telephony";
- b) to advise and assist in response to the concerns and needs of the developing countries on the technical, socio-economic and policy impacts on the introduction of "IP telephony";
- c) to prepare a report to the forthcoming World Telecommunication Development Conference, in order for this Conference to take the necessary action.

Annex P – Interconnection in the European Union (EU)

Interconnection and access in the new EU regulatory framework for electronic communications services

P.1 The Directive on access to, and interconnection of, electronic communications networks which will enter into force in 2002, harmonizes the way in which Member States regulate the market between suppliers of communications networks and services in the Community.

P.2 The Directive lays down a framework of rules that are technologically neutral, but which may be applied to specific product or service markets in particular geographical areas, to address identified market problems between suppliers of access and interconnection. The Directive provides legal certainty for market players by establishing clear criteria on their rights and obligations and for regulatory intervention. It imposes limitations on what obligations concerning access and interconnection can be imposed in which circumstances, whilst at the same time allowing for sufficient flexibility to allow regulatory authorities to deal effectively with new market problems that hinder effective competition.

P.3 The following definitions of access and interconnection will apply in the new EU regulatory framework (Directive on access to, and interconnection of, electronic communications networks and associated facilities):

- a) "access" means the making available of facilities and/or services, to another undertaking, under defined conditions, on either an exclusive or non-exclusive basis, for the purpose of providing electronic communications services. It covers *inter alia*: access to network elements and associated facilities and services, which may involve the connection of equipment, by wire fixed or wireless non-fixed means, including in particular access to the local loop and to facilities and services necessary to provide services over the local loop; access to physical infrastructure including buildings, ducts and masts; access to software systems including operational support systems; access to number translation or systems offering equivalent functionality; access to fixed and mobile networks, in particular for national and international roaming; access to conditional access systems for digital television services and to electronic programme guides. Interconnection is a specific type of access implemented between public network operators. Access in the Directive does not refer to access by end users.
- b) "interconnection" means the physical and logical linking of public electronic communications networks used by the same or a different undertaking in order to allow the users of a network provided by one undertaking to communicate with users of a network provided by the same or another undertaking, or to access services provided by another undertaking. Services may be provided by the parties involved or other parties who have access to the network.

P.4 In an open and competitive market, there should be no restrictions that prevent undertakings from negotiating access and interconnection arrangements between themselves, in particular on cross-border agreements. In the context of open markets, undertakings which receive request for access or interconnection should in principle conclude such agreements on a commercial basis, negotiated in good faith by the two parties concerned without any intervention by regulatory authorities. However, since end-to-end connectivity and interoperability of services for end users is a primary regulatory policy objective in the European Union, national regulatory authorities are empowered to intervene in a justified and proportionate manner to ensure this objective. Regulatory authorities in the European

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Union will otherwise only intervene when there is market failure in a particular market (e.g., no commercial agreements are reached). They may then impose proportionate access obligations, in accordance with the problem found, with regard to the undertakings determined to have significant market powers in a specific market. To this end regulatory authorities in the European Union will regularly undertake a market analysis review of relevant markets. They will withdraw any obligations imposed when a market analysis determines that markets have become effectively competitive.

P.5 In this context an Internet service provider (ISP) may obtain the right of access to a local network access provider in order to connect end users to the Internet backbone network of services and thus to offer global Internet connectivity, if this network access provider is deemed to possess significant market power for this particular access market²⁹.

P.6 Similarly, regulatory interconnection obligations may apply to ISPs who are providers of Internet backbone network services if these ISPs are also deemed to possess significant market power in the market or markets for such services³⁰.

P.7 The regulatory obligations could include transparency, non-discrimination, accounting separation, access, and price control including cost orientation. This range of possible obligations is to be established as a set of maximum obligations that can be applied to undertakings in accordance with the particular situation on the specific market.

P.8 Transparency of terms and conditions for access and interconnection with undertakings determined to have significant market power in a particular market serve to speed up negotiation, avoid disputes, and give confidence to market players that a service is not being provided on discriminatory terms. Openness and transparency of technical interfaces can be particularly important in ensuring interoperability.

P.9 The principle of non-discrimination ensures that undertakings with market power in a relevant market do not distort competition, in particular if they are vertically integrated undertakings that supply services to competitors with whom they compete on downstream markets. Accounting separation allows internal price transfers to be rendered visible, and it allows national regulatory authorities to check compliance with obligations for non-discrimination where applicable.

P.10 Mandating access to network infrastructure of an operator with market power in a relevant market can be justified as a means of increasing competition, but national regulatory authorities need to balance the rights of an infrastructure owner to exploit its infrastructure for its own benefit, and the rights of other service providers to access facilities that are essential for the provision of competing services. The imposition by national regulatory authorities of mandated access that increases competition in the short term should not reduce incentives for competitors to invest in alternative facilities that will secure more competition in the longer term.

P.11 Price control may be necessary when market analysis in a particular market reveals inefficient competition. This could occur in situations where lack of effective competition allows operators with market power to maintain prices at an excessively high level, or apply a lasting price

²⁹ The market for local access IP connectivity still constitutes a bottleneck controlled by incumbents in most EU markets and therefore regulatory intervention and access obligations may be justified in these markets.

³⁰ However, it is very unlikely that a market for IP global connectivity (backbone) will be retained for *ex-ante* type regulation in the European Union, because this market is already regarded as effectively competitive. Local ISPs in the European Union market are therefore likely to have to continue to rely on commercial negotiations with the global Internet backbone network suppliers unless market bottlenecks or failure arise.

squeeze to the detriment of end users. The regulatory intervention may be relatively light, such as an obligation that prices for carrier selection are reasonable, or much heavier, such as an obligation that prices are cost oriented to provide full justification for those prices where competition is not sufficiently strong to prevent excessive pricing. Operators with significant market power should in particular avoid price squeezes whereby the difference between their retail prices and the interconnection prices charged to competitors who provide similar retail services is not adequate to ensure effective competition. National regulatory authorities shall ensure that any cost recovery mechanism or pricing methodology that is mandated serve to promote efficiency, sustainable competition and maximize consumer benefits. In this regard the regulatory authority may also take account of prices available in comparative competitive markets (via so-called benchmarking approaches).

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