Consultation Paper

For

Fully Opening up of VoIP/Internet

Telephony Services

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PREFACE

The development of telecommunication infrastructure and services has constituted an engine of economic growth and social development for many decades and will continue to do so. The telecommunications environment, however, has been for the past decades, in a state of change. One of the major factors of this change is the explosive growth of Internet and Internet Protocol (IP) based applications. The present world scenario indicates that Internet has become a ubiquitous means of communication, and the total volume of packet-based network traffic has quickly surpassed traditional voice (circuit-switched) network traffic. It is becoming possible to deliver carrier grade voice services using Internet or IP-based network technologies. Further, these technologies have the potential to reduce cost, support innovation and improve access to communications services thereby contributing in reducing the digital divide.

About ten ISPs have obtained tariff approval for Internet telephony. These ISPs are providing Internet Telephony for calling Public Switched Telephone Network (PSTN) / Public Land Mobile Network (PLMN) abroad. This has resulted competition in the International Long Distance (ILD) service and ILD tariff rates are decreasing. This has benefited the subscribers for calling abroad. Similar benefits are expected from VoIP for national long distance service. However, this will require examination of certain regulatory issues like interconnection, numbering, lawful interception, emergency number dialing, Quality of Service (QoS) etc.

Nepal Telecommunication Authority (NTA), therefore, has issued this consultation paper and **request concerned stakeholders, experts and any interested party to send their comments/suggestions or inputs** either in electronic format or in written form on the various issues raised in consultation paper **by November 5, 2009.** The comments and inputs provided by the stakeholders will enable the Authority in examining present licensing conditions and suitable addressing the issues. The consultation paper is available on NTA's website (www.nta.gov.np). In case of any clarification or information, please write to ntra@nta.gov.np or contact Mr. Bijay Kumar Roy, Assistant Director, NTA (email: bkroy@nta.gov.np, tel: 977-1-4101030).

Mr. Bhesh Raj Kanel
Chairman, NTA

Executive Summary

The development of telecommunication infrastructure and services has constituted an engine of economic growth and social development for many decades and will continue to do so. The telecommunications environment, however, has been for the past decades in a state of change. One of the major factors of this change is the rapid growth of Internet and Internet Protocol (IP) based applications. Internet has become a ubiquitous means of communication in the present world and the total volume of packet-based network traffic has quickly surpassed traditional voice (circuit-switched) network traffic.

VoIP is prominent, cheaper and most revenue generating application for Internet or IP-based network. Therefore, to benefit from this technological innovation, NTA already opened up VoIP, two years before, for ILD Operators vide its decision dated 30th May 2007 and Internet Telephony for the ISPs vide its decision dated 24th August, 2007 through license amendment. To date NTA has approved Internet Telephony tariffs of ten Internet Service Providers (ISPs). ISPs are mainly providing Internet Telephony to call abroad. This has resulted in competition in ILD sector and international calling rates are continuously decreasing. However, as it is not allowed to terminate IP telephony calls to PSTN/PLMN networks, people are deprived of this technological benefit for making IP telephony calls to PSTN/PLMN within the country. This consultation paper, therefore, is prepared to seek the views of stakeholders to fully open up the VoIP/Internet Telephony for the benefit of Nepalese people.

Chapter 1 briefly introduces about Internet Protocol (IP), IP-based application, ITU definition of VoIP, difference between PSTN and IP-based network. **Chapter 2** deals with the technical aspects of VoIP/Internet Telephony where PC-to-PC or IP terminal-to-IP terminal (Pure IP voice), PC-to-Phone, Phone-to-PC, and Phone-to-Phone communication over IP network, Numbering, ENUM and QoS are described in detail. **Chapter 3** illustrates the present regulatory framework for VoIP and Internet Telephony in Nepal.

In Chapter 4, regulatory issues related to VoIP/Internet Telephony such as level playing field, Interconnection, Numbering, QoS, Emergency calling and positioning, Privacy, Security and Consumer Protection, Interoperability and Standardization, Universal Service and Service for disabled users are discussed.

Chapter 5, briefly, describes international best practices on VoIP regulation.

The last chapter, **Chapter 6** enumerates the pertinent **issues for consultation.** NTA requests comments and suggestions from stakeholders on these issues with justifications.

Chapter 1

Introduction

The use of Internet Protocol (IP)-based networks, including the Internet, continues to grow around the world due to the multitude of applications it supports and particularly due to VoIP. The IP-networks and services have evolved from Internet to 21st century IP-based network for providing carrier grade services. IP-based networks are capable of providing real-time services such as voice and video telephony as well as non real-time services such as email and are driven by pervasive use of Internet, faster Internet connections, widespread take-up in broadband and the emergence of new technologies.

The telecommunications scenario has been changing from separate networks for different services to a single network (which is IP-based) for different kinds of services. Internet Protocol has been the enabling technology for convergence of network infrastructure, service platform and user terminals or end devices. Voice over Internet Protocol (VoIP) is one of the prominent services or applications in IP platform.

VoIP has raised a number of issues of adjustment to the new environment by telephone operators and service providers, by policymakers and regulators, and by users. It has raised far greater concerns than data, pictures, music or video over IP because public voice services, particularly the long distance and international services, have been the largest source of revenue of established PSTN operators. Internet, primarily designed for data, is viewed as a disruptive technology. VoIP provided by ISPs over the Internet has made distance dependent billing irrelevant for voice calls resulting in the erosion of the established revenue base for many traditional telephone operators, who must now seek to establish new business models and new pricing structures for their services. NGN/IMS are also very near and holds a great promise for providing a plethora of IP-based services at reduced cost to the end-user.

The following technical definition has been adopted by **ITU-T Study Group 13** in **Recommendation Y.101** on Global Information Infrastructure terminology for defining the **Internet**: *"A collection of interconnected networks using the Internet Protocol which allows them to function as a single, large virtual network."*

The Internet is often characterized as being a packet-switched network. And using this type of characterization, there can fundamentally be three types of networks: circuit-switched (of which telephony is an example); connection-oriented packet-switched (of which X.25 and X.75 are examples) and connectionless packet-switched (of which Signaling System 7 and IP-based networks are examples). The IP-based network technologies are designed in a way that enables radically different environment for service development, innovation and competition, both when it comes to infrastructure platforms or service development platforms.

The connectionless packet switched nature of the IP-based networks possesses some of the important characteristics enumerated as follows:

- IP technology is based on a distributed network architecture, where routing and intelligence are distributed in the network.
- The service provision is disintegrated from infrastructure operation and the terminals attached at the edges of the network can create and offer services.
- The service development platforms have mainly been open.

These characteristics of the IP technology create good conditions for development and competition.

IP-based networks can be public or private. The Internet is in fact a complex collection of public and private networks, in which portions of the private networks are partially accessible to the public (for example, to access a private group's website, or to send e-mail to such a group). A public network is one that can be accessed by any user, while a private network is a network that can only be accessed by some restricted group of people, typically employees of a particular private company.

PSTN (public-switched telephone network) is a circuit-switched network, which is optimized for voice communication. Because of the deployed technology and the way PSTN have historically been evolved, a centralized structure has been implemented to offer telephone services. Two separate networks viz. transport and control/signaling networks are deployed in parallel in order to establish a network connection and to provide service to the two end points. Consequently, service creation and provision requires access to both the control/signaling and the transport networks and which necessitate for huge investment for the newcomers to offer services.

Thus, the differences in service offerings and capabilities between the Public Switched Telephone Network (PSTN) and IP-based networks arise from network architecture and the use of circuit-switched gateways or IP packets for the carriage of voice services. Some of the key differences between PSTN and IP-based networks for carriage of data and voice are summarized in Table-1.

Table-1, Comparison of PSTN and IP-based Network

Particulars	PSTN	IP-based Network
Transmission/carriage	Dedicated Link	Packets: Best effort/managed routing
Signaling	SS7	SIP, H323
Inter-carrier contractual relations	Transit and Termination	(Free) Peering Transit
Main Pricing basis	Distance	Quality + Volume
Main Billing Factor	Time (Minutes of usage)	Volume / capacity
Routing	Fixed way	Variable
Charging principles	Service (including connectivity)	Content and connectivity separately
Quality	Fixed Quality	QoS classes
NW Topology	Several Network levels	One Network Level
Number of POIs.	Larger Number	Less number

VoIP is also referred to as and broadly includes Voice over Broadband (VoB), Voice over Digital Subscriber Line (DSL), Voice over Internet (VoI), Voice over Wireless Local Area Network and Internet telephony. In its **2001 Report on IP Telephony**, the International Telecommunication Union (ITU) distinguished the term **"IP telephony"** as referring to voice over IP-based networks *irrespective of ownership,* in contrast to VoIP service, that refers more usually to the provision of voice services over networks competing with incumbent operators. These technologies all describe the transfer of voice (and associated services) in digital form in discrete data packets using Internet Protocol (IP) over some or the entire communication route (in contrast to the traditional circuit-switched protocols of the Public Switched Telephone Network or PSTN). These technologies all involve the digitalization, conversion and compression of recorded voice signals into data packets that are transmitted over an IP network (Internet or private network), to be reassembled and converted back at the other end of the network into voice communication.

Chapter 2

Technical Aspects of VoIP/Internet Telephony

Voice over Internet Protocol (VoIP) refers to the ***transmission of voice over IP networks***. VoIP can be delivered over the Internet and managed IP networks. The Internet is a 'best effort' medium, where QoS can't be ensured whereas in the managed IP networks, it is possible to maintain specified levels of QoS. Furthermore, gateways can be used to interconnect VoIP to the PSTN or mobile networks. This results in following **four main deployment scenarios for VoIP:**

- **Pure IP (PC-to-PC or IP terminal to IP terminal):** Here, the VoIP services are implemented using Computers (or other IP terminals) as end devices. In some deployments, the regular PSTN terminal is connected to an IP converter. The communication takes place over the Internet or over managed IP networks. Skype is an example of VoIP over the Internet.
- **IP to PSTN/PLMN (one way):** In this scenario, there is a gateway between the IP and the PSTN or mobile networks, such that one can call from the IP terminal to legacy PSTN or mobile networks. Skype-out is an example of this service type.
- **IP to PSTN/PLMN (two way)**: In this scenario, it is possible to call from an IP terminal to the PSTN/mobile AND from a PSTN/mobile terminal to an IP terminal. Skype-in is an example of this service type.
- **Phone to Phone:** In this scenario, there are gateways at the both ends which packetizes/depacketizes and performs protocol conversions for communication.

2.1 PC-to-PC Voice or IP-terminal to IP-terminal Voice (Pure IP) Communication

PC-to-PC Voice was the first generation ***IP Telephony service***, fuelled by the availability from 1994 onwards of low cost software that could, in many cases, be downloaded for free from the Internet. Users on both ends of a PC-to-PC 'call' require a personal computer (PC) equipped with audio capabilities, the same software, and an Internet connection. Calls generally have to be pre-arranged because there is no way of 'ringing' the other user, if he or she is not online at the desired time. PC-to-PC calling is well suited to Internet applications such as "chat rooms," where two or more parties can initially contact each other using text and data, and then choose to switch to voice. ***No gateway with the PSTN is required because such calls are never switched by the PSTN,*** **and the** ***principal underlying means of transmission is almost always the public Internet***. Due to ***sound quality*** limitations and the awkwardness of its use, PC-to-PC Voice likely has very ***limited impact on traditional voice services.***

The ISP's role in such scenario is limited to provide access to the Internet. The ISP network is transparent to such voice application used by the subscribers. Today PC equivalent devices like personal digital assistants (PDA) or advanced mobile handsets are available, which can also run such Internet telephony software. This type of voice communication is permitted under present ISP license.

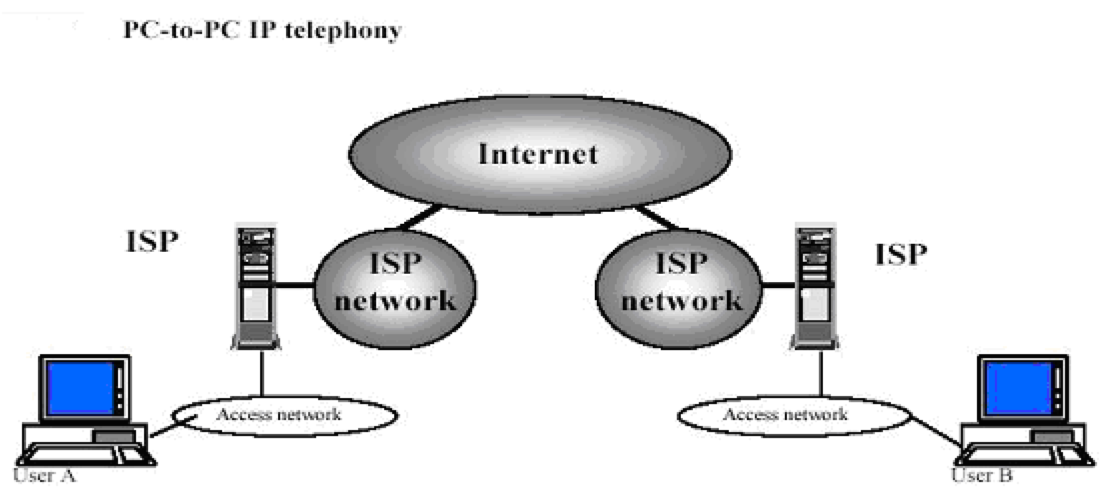

Figure 1: PC-to-PC Voice Communication

2.2 PC-to-Phone or IP terminals to PSTN/PLMN terminals

PC-to-Phone Voice became possible around 1996. From the ***service*** provider’s point of view, PC-to-Phone calling is more complex than simple PC-to-PC calling because ***calls need to be billed and routing arrangements negotiated, including interconnect payments in the distant location where applicable.*** For that reason, many ***service*** providers restrict the ***service*** to a limited range of countries in which they offer ***service***, or concentrate on the potentially large niche market of PC-to-Phone (or Fax) where ***quality***-of-***service*** requirements are not so demanding (because duplex communication is not a requirement).

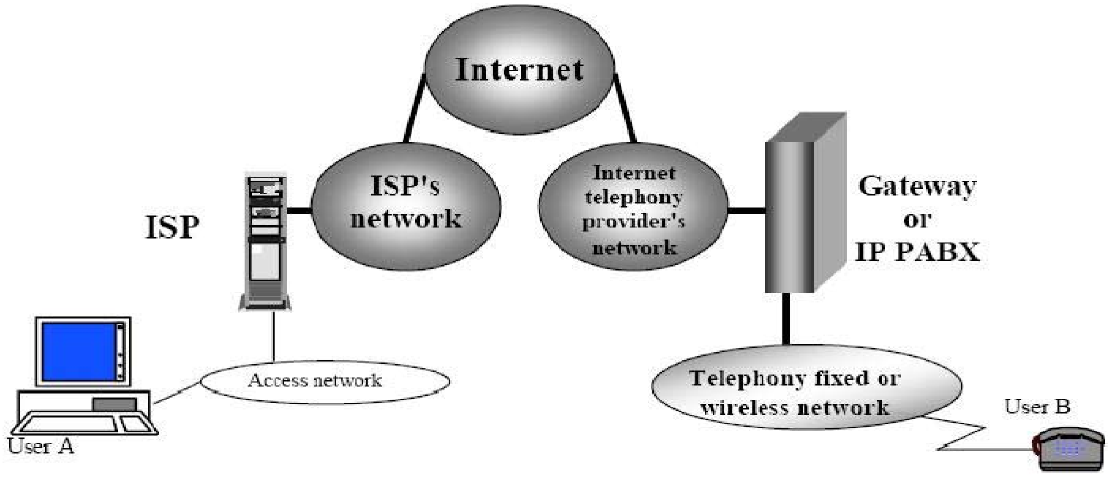

Figure 2: PC-to-Phone Communications

PC-to-Phone communication can be simply understood as a hybrid of PC-to-PC and phone-to-phone communications. The process mimics that on the originator's side in the PC-to-PC model, until the digital voice signal is traveling over the Internet in packet form. Then on the destination side, the process mimics that of the phone-to-phone model, with the signal traveling through an Internet Telephony Service Provider's POP (Gateway) and the local PSTN network before reaching the destination. At the end, a User A connects his PC to the Internet provided by an Internet Service Provider (ISP), and with service from an **Internet Telephony Service Provider operating Internet Telephony Gateway**, can speak through his computer or IP phone to any phone in the world, for ***the price of a local phone call plus a nominal Internet Telephony Service Provider (ITSP) service charge.***

Thus, voice communication from PC to Phone, needs a gateway on the receiving side to convert the IP packets back to the telephone signals (Router to PBX or PSTN). ***The calling party needs to register (using pre-paid calling card or by other means) at Internet Telephony Service Provider server and call processing and authentication is performed to verify a valid user, before a communication is established.***

ITSP are presently permitted to provide one-way PC-to-Phone service for International long distance outgoing calls. An end user is allowed to make PC–to-Phone Internet Telephony calls only on PSTN/PLMN abroad.

2.3 IP to PSTN/PLMN (two ways)

This is similar as PC-to-Phone communication as described above. However, in this case, the calls can be established from IP device (IP-Phone) or PC to PSTN/PLMN and vice versa.

2.4 Phone-to-Phone (or Fax-to-Fax) Communication

Phone-to-Phone VoIP service been commercially available since around 1997, and is the baseline for future development of IP Telephony. The reason is simple – people like to use a ***telephone*** to make phone calls. This third generation ***service*** requires operators to have originating and terminating gateways to enable transmission of voice over IP Network. This has required the operators to **enter into termination agreements** all over the world, both with independent ISPs as well as with PTOs.

Phone-to-Phone VoIP ***services*** most closely approximate the traditional ***telephone*** experience and can provide very good or very poor *quality*, depending on the nature of the network or networks over which packets are carried. While the Internet can be used as the underlying means of transmission for Phone-to-Phone calls, it is much more likely for these ***services*** to rely on closed, **managed IP networks** and **formal billing relationships among gateways and carriers**. In that respect, **Phone-to-Phone VoIP services** actually have very little to do with the Internet, but rather operate nearly in parallel to the global PSTN and its settlement rate system.

To the user, the fact that a particular call travels for part of its journey via the Internet or another IP network is irrelevant as long as the price is low and the *quality* is acceptable. For the Operator, the main motivation is to reduce costs, particularly on the ***international*** leg of a call. Fax-to-Fax ***services*** work in substantially the same way as Phone-to-Phone voice.

While using Phone-to-Phone VoIP service, the originating caller first dials his local VoIP Operator using the phone set and is connected via the local PSTN network. The VoIP operator has a gateway (called a ‘point of presence” or “POP”) that is essentially a combination of a CODEC, Computer, and Switch etc. The POP first asks the caller to enter his User ID and passcode for authentication. As the caller enters the information, the CODEC translates it into a digital signal, allowing the computer portion of the POP to verify the caller’s identity. Once verified, the POP asks for the destination phone number. Again, the CODEC converts this into a digital signal that can be understood by the computer. The computer determines which other POP in the VoIP Operator’s network of POPs is closest to the destination. Once determined, packets of data are switched to that POP, using the TCP/IP protocol, instructing the POP to establish a connection with the destination phone number. The destination POP switches the call over the local loop to the PSTN network and ultimately to the final destination, establishing the connection.

Thus, Phone-to-Phone technology allows consumers to place phone calls over the IP-based network or Internet using just their **ordinary home phones**: **no additional hardware, software, or Internet connectivity is required.** With phone-to-phone telephony, there is voice communication between individuals anywhere in the world, over the Internet ***for the price of a local call plus the VoIP Operator’s charge for providing POPs and service.*** A variety of calling card services to talk over long distances from anywhere, including different countries, are also available. Many of these services offer low rates and provide reliable billing mechanisms.

In terms of quality of service, voice over managed IP networks/backbones shows better performance than over Internet. That is why corporate intranets are increasingly being used to carry packetized voice. Companies with IP based private data network utilizing it to connect remote offices and thus saving on long distance telephone charges. This savings will be especially dramatic for traffic to international facilities.

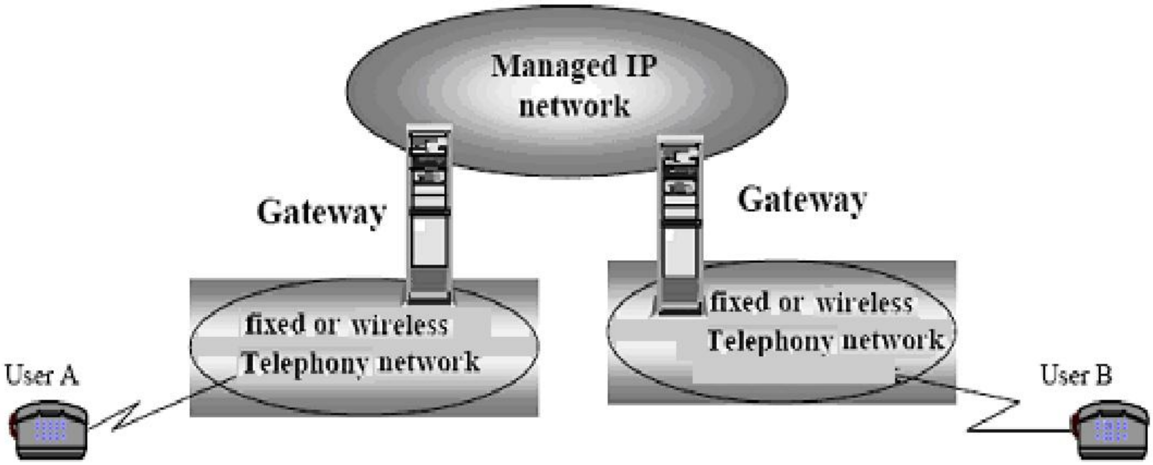

Figure 3: Phone-to-Phone Communications

Phone-to-Phone communication over the Intranet is also possible by employing a gateway attached to local PBX. The user calls an internal telephone (PBX) extension connected to the gateway. The user is asked to enter an access code and the phone number for the call. The gateway then compresses the voice, packetizes the compressed voice data on the host, and completes the call through a packet data network to another gateway at another company location.

2.5 Numbering

In the deployment of PC-to-PC and PC-to-Phone; call can be originated to PSTN/PLMN without requiring any E.164 number assigned to the calling party. The caller can call the PSTN/PLMN subscribers using their existing number if the two networks are properly interconnected. However, if the caller has to receive a call on its IP device or its adopter, it will require E.164 number allocation.

VoIP services will co-exist with traditional public telephony for many years before the transition to all VoIP is completed. The rate of growth of VoIP will depend on its access to the national E.164 number plans. Any regulatory obstacles in accessing numbers can impede or slow down VoIP development. One model is to assign a ***new number series for VoIP services,*** although this would create confusion among consumers. The best model would be to ***assign numbers similar to the current PSTN numbers and to require number portability***, so people are not forced to change their phone numbers when they want to change to a competitor offering VoIP services.

2.6 Types of Numbering in VoIP

There are varieties of numbering methods used in VoIP telecommunications and they are described below briefly.

2.6.1 Geographic numbering

- For the provision of public telephony in fixed networks;
- Addressing location-fixed network termination point;
- Geographic number associated to physical network termination point;
- In case of emergency calls the location of the caller can be derived from the (static) subscriber directory; there are many cases of emergency the caller is not able to provide reliable location information (children, old people, caller highly upset/confused/wounded etc.).
- Dynamic move of location of geo number forbidden

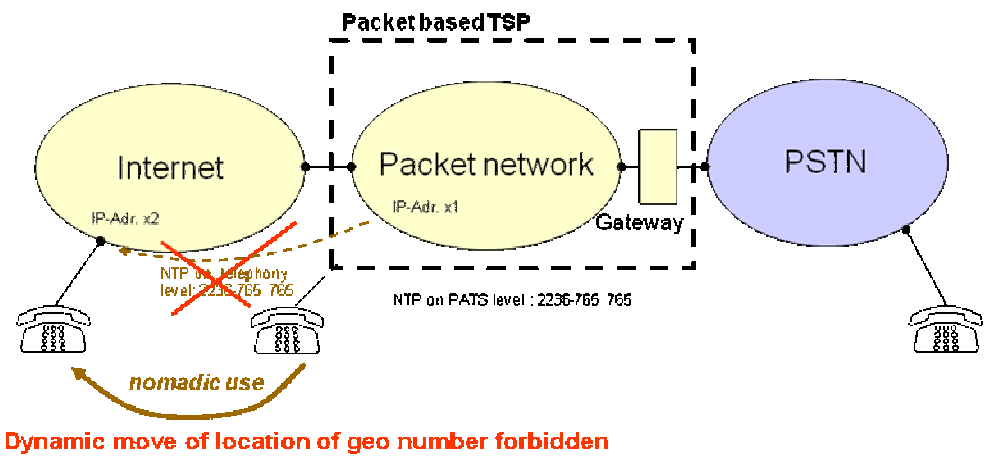

Figure 4: A Glance of Geographical Numbering

2.6.2 Location independent fixed network numbers

- Addressing of subscribers in connection with public telephone services, which enable the subscriber to hold its number independently of the location (“nomadic” services).
- Similar as in case of mobile (radio) services subscriber access on telephony level (NTP) “follows” subscriber to actual usage location.

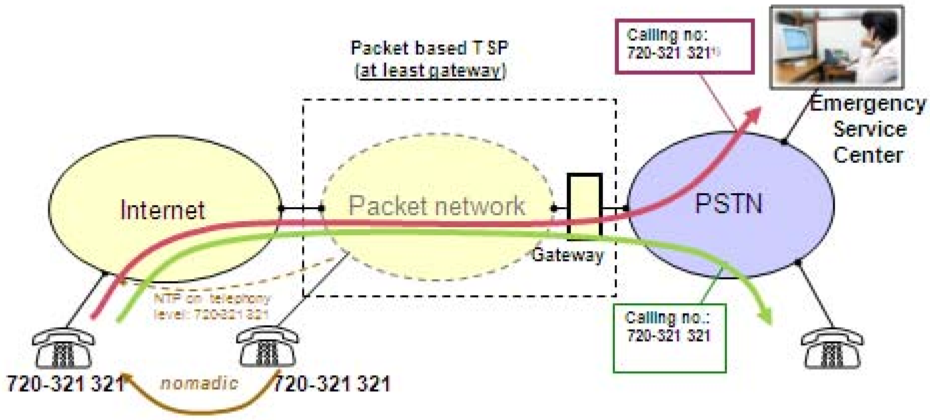

Figure 5: A Glance of Location Independent Numbering

2.6.3 Joint usage of geographic and location independent fixed network numbers

- Subscription of additional number required,
- Subscriber has usage right for geo number

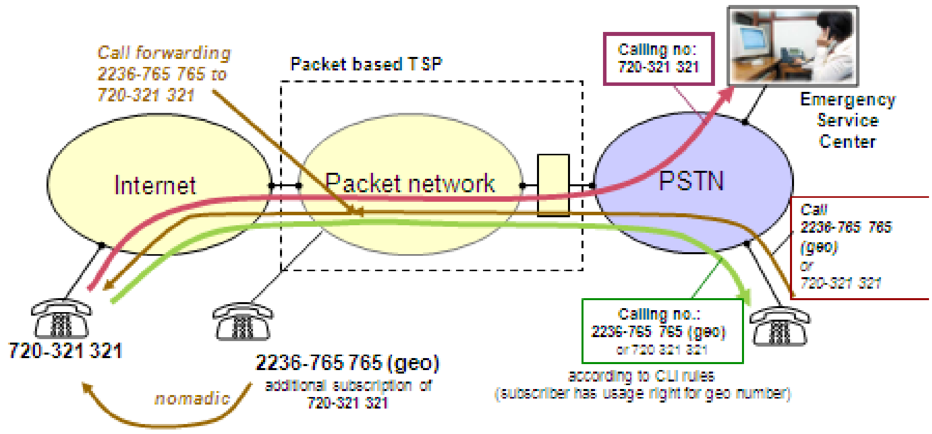

Figure 6: A Glance of Joint Numbering

2.7 ENUM

Telephone number mapping is the process of unifying the telephone number system of the public switched telephone network with the Internet addressing and identification name spaces. Telephone numbers are systematically organized in the **E.164 standard,** while the Internet uses the **Domain Name System** for linking domain names to **IP addresses** and other resource information. Telephone number mapping systems provide facilities to determine applicable Internet communications servers responsible for servicing a given telephone number by simple lookups in the Domain Name System.

The most prominent facility for telephone number mapping is the **ENUM** (or Enum, from E.164 NUmber Mapping) standard. It uses special DNS record types to translate a telephone number into a Uniform Resource Identifier or IP address that can be used in Internet communications.

Being able to dial telephone calls the way customers have come to expect is considered crucial for the convergence of classic telephone service (PSTN) and VoIP, and for the development of new IP multimedia services. The problem of a single universal personal identifier for multiple communication services can be solved with different approaches. One simple approach is the **Electronic Number Mapping System ENUM** (also known as Telephone Number Mapping), developed by the IETF, using existing E.164 telephone numbers, protocols and infrastructure to indirectly access different services available under a single personal identifier. ENUM also permits connecting the IP world to the telephone system in a seamless manner.

For an ENUM subscriber to be able to activate and use the ENUM service it needs to obtain three elements from a Registrar:

- A personal **Uniform Resource Identifier (URI)** to be used on the IP part of the network, as explained below
- One E.164 regular personal **telephone number** associated with the personal URI, to be used on the PSTN part of the network
- **Authority** to write his call forwarding/termination preferences in the Naming Authority Pointer Record **(NAPTR)** accessible via the personal URI

This works as follows: (1) the Registrar provides the Subscriber (or Registrant) with a **domain name**, **the URI,** that will be used to access a DNS server to fetch a NAPTR record, (2) a personal E.164 telephone number (the ENUM number). The URI domain name of (1) is bi-univocally associated (one-to-one mapped) to the subscriber E.164 ENUM number of (2). Finally (3) the NAPTR corresponding to the subscriber URI contains the subscriber call forwarding/termination preferences. NAPTR record accessible via the personal URI

Therefore, if a calling party being at the PSTN network dials a called party ENUM number by touch typing the E.164 called party number, **the number will be translated at the ENUM gateway** into the corresponding URI. This URI will be used to look-up and fetch the NAPTR record obtaining the called party wishes about how the call should be forwarded or terminated (either on IP or on PSTN terminations) – the so-called access information – which the registrant (the called party) has specified by writing his/her choice at the 'NAPTR record', "**Naming Authority Pointer Resource Records**" as defined in RFC 2915, such as e-mail addresses, a fax number, a personal website, a VoIP number, mobile telephone numbers, voice mail systems, IP-telephony addresses, web pages, GPS coordinates, call diversions or instant messaging. Alternately, when the calling party is at the IP side, the User Agent (UA) piece of software of the dialer will allow to dial a E.164 number, but the dialer UA will convert it into a URI, to be used to look-up at the ENUM gateway DNS and fetch the NAPTR record obtaining the called party wishes about how the call should be forwarded or terminated (again, either on IP or on PSTN terminations).

One potential source of confusion, when talking about ENUM, is the variety of ENUM implementations in place today. Quite often, people speaking of ENUM are really referring to only one of the following:

- **Public ENUM:** The original vision of ENUM as a global, public directory-like database, with subscriber opt-in capabilities and delegation at the country code level in the **e164.arpa** domain. This is also referred to as User ENUM.
- **Private ENUM:** A carrier, VoIP operator or ISP may use ENUM techniques within its own networks, in the same way DNS is used internally to networks.
- **Carrier ENUM:** Groups of carriers or communication service providers agree to share subscriber information via ENUM in private peering relationships. The carriers themselves control subscriber information, not the individuals. Carrier ENUM is also referred to as Infrastructure ENUM, and is being the subject of new IETF recommendations to support VoIP peering.

2.8 Quality of Service (QoS)

With POTS, there are detailed recommendations on Quality of Service from the ITU and in many national regulations. In managed VoIP services it is possible to provide measurable QoS, but this is more difficult in best effort services. Another important issue is the willingness of facility based operators to offer access to QoS provision to non-facility based operators. For example, a major debate in Europe and other regions is the lack of QoS provision in the wholesale Bit stream access products offered by the PSTN incumbents. Regulators will need to take the lead in ensuring consumer protection with respect to QoS.

Quality of Service of an IP network used for telephony is also an important issue. The packet mode of data transmission used by IP networks may introduce degradation in speech quality due to following factors:

- **Packet Loss**: Possible disappearance of packets during the communication. Highly stable media like optical fiber reduces packet loss to virtually zero.
- **Delay**: This refers to transit time, including the time taken to reassemble the packets upon arrival and compensate for fluctuations in transit times (this overall transit time must be lower than 400 ms.). Such delays are network dependent and are taken care in network designing.
- **Jitter:** Variation in the packet arrival delay. Synchronization of network is very important to reduce such jitter.
- **Echo:** This refers to the delay between the transmission of a signal and receipt of the same signal as an echo. Effective echo cancellation can be used in well-planned networks.

Since packet networks are known to suffer from some impairments vis-à-vis circuit switched networks with reference to real-time services it becomes imperative to measure the extent of such impairments like packer loss, mean packet delay and packet jitter etc. These impairments can result from vocoder activity; signal processing, protocols conversion (from CCS7 to SIP or H.323 etc.) and packetization among other factors. Such impairments are aimed at benchmarking against QoS.

Service user perceives quality and in so far as he/she concerned, qualitative measures of quality, such as Mean Opinion Score (MOS) are enough. However additionally from service provider’s and Regulator’s perspective objective, measurable parameters are required to ensure that the qualitative requirements are being met.

The voice quality can be measured by subjective methods such as Mean Opinion Score (MOS) mandated in ITU-T Recommendation P.800 and parametric estimation (Objective) methods like PSQM (ITU-T Q.861), PSQM+, PESQ (ITU-T P.862) and PAMS (British Telecom). The subjective methods are time consuming and expensive to use while parametric estimation can be done quickly and inexpensively on the voice codecs. Recently ITU-T has created the “E-Model” for estimating the voice quality in packet networks. The application of the “E-Model” results in the transmission Rating or the R-Factor. The resultant value of the R-factor is known as the R-value. The Table-2 below shows the relationship between R-value and MOS. From table below for “Toll Quality” voice, the R-value must be 80, giving MOS of 4 or higher, while for “below Toll Quality” 70 or higher i.e. MOS higher than 3.5.

Table-2, R-Value and MoS for measuring voice quality			
R-Value (Lower limit)	**MOS (Lower limit)**	**User Satisfaction**	**Quality**
90	4.34	Very Satisfied	Toll
80	4.03	Satisfied	Toll
70	3.60	Some Users dissatisfied	Below-Toll
60	3.10	Many Users dissatisfied	Unacceptable (use only in very special conditions)
50	2.58	Nearly all users dissatisfied	Unacceptable

The Advantage Factor: The E-Model provides for downward adjustment in the R-value in situations where the user is being provided certain "advantage", for instance, in building mobility (5 points), vehicular mobility (10 points) and providing access to hard-to-reach locations via multi-hop satellite connections (20 points). This means that when G.711 codec is used in a connection with two satellite hops, service shall be Toll Quality even with end-to-end delay of 400ms, and below toll quality with delay around 500 ms. Similar adjustments can be made for calls involving cellular networks.

Chapter 3

The Present Regulatory Framework for VoIP and Internet Telephony in Nepal

NTA vide its decision dated 2063.04.04 ***(20th July 2006)*** defined **IP Telephony** as ***"the transmission of voice signals over packet switched IP-based networks"*** and has categorized it into two groups:

(i) VoIP (Voice communication over managed IP based network) and
(ii) Internet Telephony (Voice communication over the public Internet)

3.1 VoIP

The NTA vide its decision dated 2064.02.16 ***(30th May 2007)*** permitted the licensees who have **International Long Distance (ILD)** service license, to provide VoIP. The scope of VoIP is reproduced below:

3.1.1 Scope of VoIP

The ILD operators after getting their license amended are allowed to establish VoIP Gateway for providing following services to its customers:

(i) Phone-to-Phone communication using the managed IP-based packet switched network.

(ii) The operators have to provide and announce certain access code for VoIP so that customers have the choice to access a gateway of any VoIP service providers and get service of their desired quality.

No additional license fee was charged and the quality of service parameters for VoIP was defined and is reproduced below in Table 3:

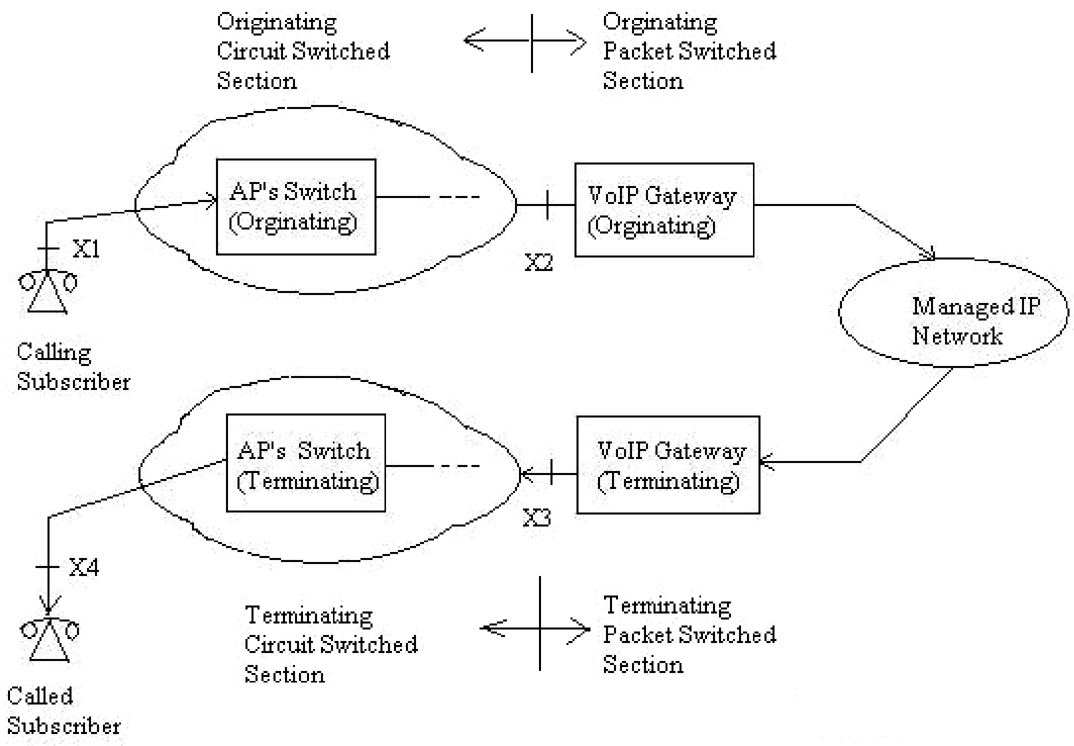

Figure 7: Reference VoIP Network

Table 3, Quality of Service Specifications for VoIP		
Parameters	**Toll Quality**	**Below Toll Quality**
1. MOS (Mean Opinion Score) **Or** **R-Value**	**> = 4 (without satellite link)** **and** **> = 3.5 (with satellite link)** **Or,** **> = 80 (without satellite link)** **and** **> = 70 (with satellite link)**	**< 4 (without satellite link)** **and** **< 3.5 (with satellite link)** **Or,** **< 80 (without satellite link)** **and** **< 70 (with satellite link)**
2. One-way End-to-End Delay	**< = 150 ms (without Satellite Link)** **< = 400 ms (with Satellite Link)**	**> 150 ms (without satellite link)** **> 400 ms (with Satellite Link)**
3. Grade of Service (GoS)	**< 2 %**	**< 2%**
4. Packet Loss	**< 0.1 %**	**< 2 %**
5. Packet Jitter	**< 5 ms**	**< 10 ms**
6. CCS7 signaling delay	**To confirm with Q.709**	**To confirm with Q.709**

3.1.2 The Issue with VoIP

However, the Operators having ILD service license, have not implemented and announced such service publicly; the consumers are still deprived of getting benefit from such technological advancements and NTA is facing challenges from VoIP call bypass resulting in the loss in government revenue.

3.2 Internet Telephony

The NTA vide its decision dated 2064.05.07 (***24th August 2007***), amended the license of Internet Service Providers (ISPs) to provide Internet Telephony. The scope of Internet Telephony is described below:

3.2.1 Scope of Internet Telephony

The ISPs after getting their license amended are allowed for providing following services to its customers:

- (i) PC-to-PC voice communication using public Internet
- (ii) PC in Nepal –to- Phone abroad voice communication using public Internet
- (iii) H.323 terminal/SIP terminal-to-H.323 terminal/SIP terminal voice communication using public Internet within Nepal and Abroad
- (iv) The Customer Premise Equipments or End Terminals will be named using IANA assigned IP address scheme; but not the ITU's E.164 numbering scheme.

However, the ISPs providing Internet Telephony services are not allowed to terminate any packetized voice calls (VoIP calls) to PSTN or PLMN terminals.

No additional license fee was charged and QoS was left to the market forces for competition. "Conditions to be abided by the licensee to operate Internet Telephony services" was also developed.

3.2.2 The Issue with Internet Telephony

Even though a number of ISPs are providing Internet Telephony, they are ***mainly focused on providing voice calls to abroad*** rather than within the country. They are providing Internet Telephony through the pre-paid calling cards and using **softphones** and **IP phones** to some extent. The people have benefited from such service as they can make calls to abroad in very cheaper rates (about NRs. 1.25/minute for US, UK etc. and NRs. 2.50/minute for India) which is not possible otherwise using PSTN or Mobile. This has created pressure to decrease international long distance tariff rates of PSTN/PLMN operators and competition has started in ILD sector.

However, people living in rural areas having access to Internet are not yet able to enjoy such a cheaper rate service to make calls to PSTN or PLMN within the country due to regulatory restrictions that *ISPs providing Internet Telephony services are not allowed to terminate IP calls to PSTN/PLMN terminals.* Also, due to this restriction, there is the scope of grey market/VoIP call bypass causing loss in government revenue.

NTA, therefore, has come up with this consultation paper to seek the views of the concerned stakeholders.

Chapter 4

Regulatory issues

4.1 Level Playing Field

The basic telephone service providers or mobile operators or WLL operators carry voice calls under the well-defined regulatory environment. On the other hand, ISPs providing Internet Telephony services have evolved under the "light touch regulation". As a result, the licensing and other regulatory conditions are radically different for both these segments. For basic telephony service or mobile telephony service, tendering is needed whereas for Internet Telephony service or ILD service, one can get the license immediately upon the submission of an application and satisfying certain conditions. There is vast difference in the license fee (and renewable fee) of the basic telephone service provider and the VoIP/Internet Telephony service provider. It is usually argued that non-uniform regulatory burden may disturb level playing field between basic telephone service provider/WLL/Cellular Mobile Service Provider/Rural Telecom Service Provider and Internet Telephony Service Providers. The voice telephony service providers are always in a view that similar license fee and regulatory burden should apply to any other operators if they are permitted to provide voice telephony within the country using any methodology.

The world scenario indicates that in most of the countries ISPs are under light touch regulation and regulatory levies imposed on them are low; still they have been permitted to provide Internet telephony. Many of the regulators have mandated ISPs to clearly inform their subscribers about quality of service before providing such services. It is also argued that opening up of Internet telephony will have positive impact on development of telecom services, Gross domestic Product (GDP) of the country and greatly benefit many who are not able to afford the same till now, apart from increasing competition and reducing prevailing price. Many also feel that Internet telephony is an information service and therefore it is an exclusive domain of ISPs. There has been extensive debate globally on the classification of the Internet Telephony service.

In view of the above discussions, therefore, it needs to be discussed whether ISPs be permitted Internet telephony to make calls to PSTN/PLMN within the country or a separate license category named ***"IP Telephony Service Provider (ITSP)"*** be created to provide VoIP and Internet Telephony calls to PSTN/PLMN within the country. ***NTA seeks views of stakeholders in this regard.***

Table 4, Comparison of Regulatory Levies among Various Licenses

S.N.	Type of Licenses	Name of Operator	Licensing Procedure	License Fee (NRs.)	Renewable Fee (NRs.)	License Period (Years)	Royalty (NRs.)	RTDF contribution (NRs.)	Other Obligations
1	**Basic Telephone Service Provider License**	Nepal Doorsanchar Company Limited (NDCL)	As per clause 23 (3) of Telecom Act	15,58,50,000	14,02,65,000	10	*4% of their annual gross income*	*2% of their annual gross income*	
		Nepal Satellite Telecom Pvt. Limited	By bidding process As per Clause 23 (2) of Telecom Act	25,00,000	22,50,000	5	*4% their annual gross income*	*2% their annual gross income*	*Service to be started from rural areas of Mid Western Region*
2	**Basic Telephone Service based on WLL (Wireless Local Loop)**	United Telecom Limited (UTL)	By bidding process As per Clause 23 (2) of Telecom Act	10,00,00,000	9,00,00,000	10	*4% of their annual gross income & committed Royalty whichever is higher*	*2% of their annual gross income*	*15% of their annual investment in rural sector*
3	**Cellular Mobile Operator**	Nepal Doorsanchar Company Limited (NDCL)	By the directive issued by GoN as per clause 20 of Telecom Act	21,00,00,000	20,00,00,00,000	10	*4% of annual gross income & committed Royalty whichever is higher*	*2% of their annual gross income*	*15% of their annual investment in rural sector*
		Spice Nepal Pvt. Ltd. (SNPL)	By bidding process As per Clause 23 (2) of Telecom Act	21,00,00,000	20,00,00,00,000	10	*4% of their annual gross income*	*2% of their annual gross income*	*15% of their annual investment in rural sector*

4	**Rural Telecom Service Provider**	STM Telecom Sanchar Pvt. Ltd.	By bidding process As per Clause 23 (2) of Telecom Act	1,00,000	Not exceeding 4% (four percent) of the Licensee's gross annual revenues in the fiscal year immediately preceding the start of the relevant renewal period	10	*4% of their annual gross income*	*2% of their annual gross income (exempted for first 5 years of operation)*	*Service to be started from 534 VDCs of Eastern Development Region*
		Smart Telecom Pvt. Ltd.	By bidding process As per clause 23 (2) of Telecom Act	1,00,000	Not exceeding 4% (four percent) of the Licensee's gross annual revenues in the fiscal year immediately preceding the start of the relevant renewal period	5	*4% of their annual gross income*	*2% of their annual gross income*	*Service to be started from rural areas except Eastern Development Region (EDR)*

5	**Internet Service Provider (ISP)**	This license is issued to 38 licensees,	Upon Application	3,00,000.00	2,27,000.00	5	*4% of their annual gross income*	*2% of their annual gross income*	
6	**Rural Internet Service Provider (RISP)**	This license is issued to 1 licensee,	Upon Application	100	90	5	*4% of their annual gross income*	*2% of their annual gross income*	
7	**ILD Service Provider**	Open as per clause 23 (2) of Telecom Act	An applicant should have obtained either GSM cellular Mobile license OR Basic Telecommunication License from NTA & must have operated at least 100,000 telephone lines.	6,25,00,000	5,62,50,000	5	*4% of their annual gross income*	*2% of their annual gross income*	

4.2 Interconnection

Interconnection to the legacy PSTN networks is essential for the success of VoIP services. This interconnection is implemented by using gateways and contractual agreements between VoIP/Internet Telephony providers and PSTN operators. These interconnection arrangements should be monitored by regulators. ***If fair and non-discriminatory conditions for interconnection are not established in a timely fashion in the marketplace, regulators should intervene*** following traditional interconnection principles, as reasonable interconnection is a precondition for the successful development of VoIP.

The present interconnection guidelines can be downloaded from NTA website (www.nta.gov.np). Keeping in view the number of ISPs providing Internet telephony, it may be difficult to provide interconnection to each ISPs and therefore, new category of license may be created to ensure interconnection with PSTN/PLMN operators. However, it is to be noted that interconnection can only be provided at appropriate locations. ***NTA requests comments and suggestions from stakeholders in this regard.***

4.3 Numbering

Numbering is the Public User Identity by which a subscriber is identified in a Network. In PSTN and PLMN this is a telephone User Resource Identifier (TEL URI) in E.164 format such as +977 01 410 1030 which is allocated to different service providers under provisions of **National Numbering Plan** in accordance with their licensing conditions. E.164 numbering generally identifies the geographical location of the subscriber and the service provider providing such services.

Generally subscriber prefers to have E.164 format of numbering primarily due to ease of use and familiarity. Moreover, billions of the devices currently available on different networks use only numeric keypads. Hence E.164 numbers can easily be dialed using such devices.

Fully Opening of the VoIP/Internet telephony to call PSTN/PLMN may increase requirement of numbering blocks, as the Internet telephony subscriber will also receive incoming calls. As per the licensing conditions, allocation of E.164 numbers is not permitted to ISPs but success of VoIP/Internet telephony is greatly linked with the ease with which a subscriber can dial a call and receive a call. Any non-familiar method to allocate addresses is likely to restrict the potential and popularity of VoIP/Internet telephony. More over allocation of dedicated public IP address to individual device may also be difficult due to shortage of IPv4 address space. Dialing an IP address requires special type of telephone commonly known as IP Phones or dialers using PC or equivalent devices. These options are costly and may neutralize the price advantage of VoIP/Internet telephony.

Assuming that certain number blocks can be identified to ***IP Telephony Service Provider*** **(ITSP)** for VoIP/Internet telephony, there can be different combinations possible to allocate E.164 numbering in existing 10 digit frameworks. One possible allocation could be: ***(Operator Code: 2 Digits) + (ITSP Operator Code: 2 Digit) + (Subscriber Number: 6 Digits)***

As already explained, the Telephone Number Mapping (ENUM) is another globally adopted methodology for addressing the end devices in case of Internet Telephony. ENUM permits additional means for identifying users, enriching the user identification information, creating private number plans, introducing special billing arrangements (e.g. reverse billing, split billing, etc.) makes it suitable for Internet telephony-based solutions. The main argument against ENUM approach has been that it should not be seen as solution for existing crunch for E.164. ENUM will require allocation of at least one E.164 number to each entity irrespective of services being offered to such entity. ENUM facilitate use of existing number for an entity to provide multiple services including Internet telephony but does not permit a service provider to use existing E.164 number allocated to an entity by some other service provider. ENUM does not resolve problems associated with number portability also.

VoIP services will co-exist with traditional public telephony for many years before the transition to all VoIP is completed. The rate of growth of VoIP will depend on its access to the national E.164 number plans. ***Any regulatory obstacles in accessing numbers can impede or slow down VoIP development.*** One model is to assign a new number series for VoIP services, although this would create confusion among consumers. The best model would be to assign numbers similar to the current PSTN numbers and to require number portability, so people are not forced to change their phone numbers when they want to change to a competitor offering VoIP services.

In view of above discussion, the stakeholders are requested to provide suggestions on allocating the numbering to the IP Telephony Service Providers.

4.4 Quality of Service (QoS)

Quality of speech in any communication service is an important performance parameter. Subscribers are habituated to the PSTN/ PLMN voice quality and expect similar quality from VoIP/Internet telephony also irrespective of the technology being used to provide such services. Ensuring good voice quality will therefore be necessary for VoIP/Internet Telephony service provider. Though VoIP/Internet telephony standards do not prescribe minimum Internet access speed for good quality of service, it is generally perceived that broadband connection will be required to provide good speech quality. ITU-T Recommendation G.114 (5) defines maximum one-way latency as 150 ms for good VoIP/Internet telephony voice quality. This puts a restriction on round trip delay, packet loss, and speed of Internet access.

NTA has defined certain parameters for VoIP in International long distance segment like end to end delay not exceeding 150 ms, Jitter not exceeding 5 ms, packet loss not exceeding 0.1% and R-value greater than 80 which is already presented in Table 3 above. At that point of time, the QoS for Internet telephony was left to the market and Internet telephony within country on PSTN/ PLMN was not considered.

Therefore, now the issue of consideration is whether there is a need to define QoS parameters for VoIP/Internet telephony within country for terminating IP calls to PSTN/PLMN or it should be left to service providers. Both the models are prevailing world over. In some countries all VoIP/ITSP have to match QoS parameters as defined for PSTN/ PLMN whereas in some other countries no specific QoS have been defined.

In the light of above discussion, ***comments of the stakeholders are invited in this issue as well.***

4.5 Emergency calling and positioning

The possibility to perform emergency calls and to route the call to the nearest authority (fire department, police, hospitals etc.) has been defined as a core element of Publicly Available Telephony Services (PATS) in Europe. Similar requirements are part of regulation in other countries. Location information is also more frequently becoming a requirement for both fixed and mobile telephony. **In VoIP/Internet Telephony, it is possible to maintain positioning and routing information for emergency calls.** ***However, this requires use of VoIP services from fixed locations.*** However, one of the promising characteristics of VoIP services is nomadic use. In nomadic use, at the current level of technological development, the position information cannot be connected to the emergency call. Though this type of system is claimed to be designed and have been put into service in some of the EU Member states, it serves the purpose to some extent only. This is a challenge both to the market players and to the regulatory framework. This issue is being hotly debated world over. Regulators should take a lead in facilitating the resolution of these important emergency issues in future networks.

Some of the countries have gone ahead with the Internet Telephony services with clear instructions to the subscribers that VoIP/Internet Telephony service can not provide access to emergency services.

It can be noted that emergency service limitations are not unique to VoIP services. Subscribers to mobile wireless services are, by definition, not limited to a single location. These subscribers require the development of special practices or technologies to direct emergency services as closely as possible to the subscriber's actual location. ***In view of above discussion, stakeholders are therefore, requested to provide their suggestions/comments in this regard.***

4.6 Privacy, Security and Consumer Protection

In regular telephony services the security and consumer protection standards have been defined and are generally found adequate.

With our increasing dependence on computer networks, the importance of network security, including appropriate provisions for law enforcement concerns and privacy, needs to be addressed. The explosive growth in the use of computers has increased the dependence of organizations and individuals on the information stored and communicated using these systems. This has led to a heightened awareness of the need to protect data and resources, provide law enforcement officials with effective tools to combat cybercrime, develop a global culture of cyber security, and find effective means to combat spam.

Some developed countries have provisions designed to facilitate tracking and eavesdropping by law enforcement authorities, legal frameworks to combat spam, as well as provisions to protect the identities of users of communication services and the content of those communications. ***In many such countries, privacy and security provisions are very general and apply to any medium, not just to telecommunications.***

A major policy question is whether, and if so to what extent and how, provisions related to security and privacy should apply to IP-based networks or IP-based applications such as VoIP, taking into account the traditional differences in the treatment of public and private networks. In particular, to what extent, if any, should there be provisions for IP-based networks to ensure the identification and traceability of packet-originators and/or recipients.

The use of advance encoding and encryption techniques by VoIP/Internet telephony providers also poses challenges for lawful monitoring. It is important to recognize vital requirement for law enforcement agencies to monitor and intercept Internet (IP) based voice traffic; hence ITSPs providing Internet telephony within country may have to ensure suitable encryption in coordination with concerned security agencies so that effective monitoring of all IP packets can be ensured. However, this will require installation of Lawful Interception equipment at each VoIP/Internet Telephony Service Providers interested to provide VoIP/Internet telephony service within country and may add some Capex for ITSPs starting this service. Monitoring of International calls can be done at International Internet gateways. Similarly, it will be possible to monitor the calls to PSTN/PLMN at terminating exchange however effectiveness of such monitoring will be of greater concern.

In view of above discussion, NTA seeks suggestions from the stakeholders that the lawful interception be made mandatory?

4.7 Interoperability & Standardization

The ITU-T Recommendation H.323 is an umbrella standard for specifying for an IP-based multimedia conferencing system. It refers to a couple of other standards, which specify signaling protocols, media coding and call control services. H.323 uses an evolutionary approach to VoIP, which offers a high degree of interoperability to legacy 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.

Various other protocols have been defined between Media Gateway (MG) to MG, Media Gateway Controller (MGC) to MG, MGC to Signaling Gateway (SG), and MG to SG. It is suspected that devices using different protocols may not interoperate. Standardization as stipulated by IETF, ETSI, ITU will be required to ensure interoperability to provide VoIP-based enhanced services in order to ensure rapid development in the public interest.

Therefore, it is important to establish interoperability between these standards. These developments should be monitored by regulators, and if the market players do not find adequate solutions for interoperability, regulatory measures may be necessary. However, at this level of technological development, it may be left to the vendors/manufacturers and standardization organization to ensure the interoperability of different systems. ***The views of the stakeholders are requested in this regard.***

4.8 Universal Service

In rural areas the new wireless technologies will play an important role, where a combination of VoIP services and wireless infrastructures can enable a more efficient development of all communications services, including basic voice services. ***A particular problem related to use of VoIP for the provision of universal service is in-line powering of terminals.*** Traditional telephony service is designed with back-up power, and so it continues to work in case of electricity power failure. The current VoIP services/terminals are dependent on a functioning power supply. ***A requirement for in-line powering of terminals could put an enormous burden on the VoIP operators and slow development of service to un-served rural areas.*** Regulators should facilitate VoIP growth as a driver for network extension. For the longer term, the requirements for emergency communication standards and services can best be addressed by considering all the new technologies and services in the NGN, and most particularly VoIP and mobile.

In the light of above discussion, NTA seeks views of stakeholders that should the requirement for in-line powering of terminals be made mandatory.

4.9 Services for Disabled Users

Perhaps the best example of disabled user access is the use of *Message Relay Service* allowing hearing-impaired subscribers to communicate with others using operator intervention. A hearing person who wishes to communicate with a hearing-impaired person dials a toll-free number to be connected to an operator who contacts the hearing-impaired user and relays the communication using a *teletypewriter device.* Conversely, a hearing-impaired person uses a teletypewriter and contact through the intervening operator to communicate with a hearing person.

Technical interfaces between teletypewriter devices and IP-based technologies are still under development, meaning that conventional *Message Relay Service* is not feasible for at least some VoIP service configurations at the present time. It has been suggested that VoIP/Internet Telephony service providers might be able to provide a functionally equivalent means of text to speech and speech to text intervention using web-based facilities (or by using an instant messaging application). However, another view favours a Message Relay Service-based solution given the familiarity of teletypewriter devices in the hearing-impaired community. For the proliferation of VoIP/Internet Telephony, at this time, it may be left to the service providers and the regulator closely monitor the development of technology and impose conditions at suitable time. ***Stakeholders are kindly requested to provide suggestions/comments in this regard.***

Chapter 5

International Best Practices on VoIP Regulation

We find a variety of different definitions in use worldwide for VoIP. The table below summarizes some of the key categories of definitions of VoIP. This difference is because of the specific market conditions and economic status of each country. It has been a key part of policy-makers' and regulators' work to establish a relevant definition of VoIP as it applies to their specific market. Regulatory definitions of VoIP and VoIP providers have important implications, not only for regulation, but also for the development of the wider market, innovation and competition.

Table 5, Main Categories of VoIP definitions[1]

Definition	**Examples**
Quality of service (*superceded*)	Japan, India
Equipment and terminals used, *and/or*	India, Japan, Jordan, Malaysia, Spain.
Network architecture, *and hence*	Israel, Saudi Arabia.
Functionality	Hong Kong
Numbering system	Japan, Taiwan (China).
Whole or part-provision of service over IP/PSTN	Israel, Jordan, ITU.
By service	Some countries distinguish between VoIP services in whether: • VoIP is viewed as a data or information service, as opposed to a voice or telecommunication service (e.g. Egypt, Jordan, the United States, Barbados); • VoIP as nomadic and non-nomadic services (Italy);and • Publicly Available Telephone Services (PATS) and Publicly Accessible Electronic Communications Service (PAECS) (EU).
Users/usage	Some countries make further distinctions according to users: • Public or closed group of end-users (e.g. Chile); and • Corporate or residential use (e.g. Australia, Tunisia).

[1] The status of voice over Internet Protocol (VoIP) Worldwide, 2006, ITU, The Future of Voice, 15-16 Jan 2007

5.1 Asia-Pacific

In Asia-Pacific, we find huge diversity of approaches from early and liberalized approaches to VoIP (e.g. Australia, Japan, Rep. of Korea, Malaysia and Singapore) to well developed licensing systems (e.g. Bangladesh and India) and outright bans (several of the Arab states, including Kuwait, Qatar and UAE). Where the Arab states permit VoIP, it has been mainly adopted by incumbents. For example, in Jordan, early concerns over the introduction of VoIP originated with concerns to preserve Jordan Telecom's exclusivity rights, as guaranteed under its License. No entity other than Jordan Telecom was permitted to offer voice service to the public using VoIP prior to 1 January 2005, including foreign originated calls terminating on Jordan's PSTN. There is work underway on adopting a common approach to VoIP as part of work on NGN by APEC (Asia Pacific Economic Cooperation).

5.2 North America

In the US and Canada, where VoIP applications are legal, different service models are developing – some VoIP providers are offering their services for free, bundled in with other service offerings. Other service providers charge for long-distance calls carried over VoIP, similar to traditional fixed-line telephone services. Other VoIP providers allow flat-rate calling regardless of distance, a business model that is gaining in popularity.

As defined by the 1996 Telecommunication Act that distinguishes telephone services from information services, the FCC in the United States does not consider VoIP as a traditional telephone service, but as a computer-based 'information service', that is relatively unregulated. USA has adopted a liberalized approach to VoIP, since it is considered an Internet application/Information Service. The FCC has sought to adopt a "light regulatory touch" approach. There are no licensing requirements, but a Universal Service contribution is required.

5.3 Europe

VoIP is not explicitly regulated in the EC framework, and European countries have tended to develop their own approach to VoIP in terms of regulation. This has been called by some a "laissez-faire" approach to VoIP regulation. In the Scandinavian countries, regulators have tended to adopt a light regulatory touch on the basis that "voice is voice", so Finland, Iceland, Norway and Sweden have referred back to the PSTN regulations. France and Ireland adopted an early and relatively liberalized approach to VoIP and actively advocated VoIP for open competition, greater choice and lower prices.

Ireland has focused on consumer protection issues, as illustrated by its publication of "Guidelines for VoIP service providers on the treatment of consumers" in 2005. Access to emergency services was a specific topic of concern for the UK. OFCOM developed an interim forbearance policy allowing VoIP providers to offer emergency services, without other regulatory requirements for PATS. This was to "diminish the disincentives" to provide access to emergency services. After consultation with the European Commission and European Regulators' Group on the New Regulatory framework, OFCOM ended its policy of interim forbearance policy and introduced a mandatory code of practice for consumer information for VoIP providers. Italy has adopted an original approach to VoIP legislation in terms of nomadic and non-nomadic services. Germany and Poland are still under consultation in relation to VoIP services.

These different approaches have been broadly observed by the European Commission. At the European level, there have been moves by the European Regulators' Group to formulate a common approach to regulation, with pro-competitive policies a particular concern of the European Commission. More recently, the EU Information Society and Media Commissioner has suggested that EU operators may be required to split out their infrastructure and services divisions in order to guarantee fair access and promote competition and investment.

5.4 Africa

VoIP was banned in many African countries. Many African governments continue to prohibit VoIP adoption except by monopoly incumbents, with the notable exceptions of Mauritius (the first country to explicitly liberalize VoIP and implement a licensing regime for VoIP services on the continent), Nigeria and South Africa. Regulatory statements include many references to technology neutrality and service specificity, but in practice, on the basis of the regulators' statements, VoIP is frequently only legal for those holding an international gateway license and whilst there are moves to extend these to mobile operators, in many countries, currently only incumbents hold international gateway licenses.

African incumbents' initially sought to exploit profit margins between falling costs in international minutes to relatively low prices, whilst continuing to sell them at higher PSTN prices. According to Balancing Act Africa, these price differences arose mainly for three reasons:

- The introduction of international competition (helped by the push to deregulation and commitments made under WTO GATS) that has pushed costs down.
- The shift to cheaper call rates through the use of data networks.
- Growing demand for international calls and the transformation of the international calling market from a low-volume, high-margin market to a higher-volume, lower-margin market (through greater demand from multinational corporations and migration).

This has led to a large, grey market in VoIP-based calling, with VoIP service providers exploiting ‘arbitrage’ opportunities. This phenomenon even resulted in declines in the annual international traffic volumes of some African incumbents according to Balancing Act Africa, as traffic went over to the grey market. Balancing Act Africa concludes that “African incumbents have been faced with a much starker choice than their developed world counterparts: to watch their traffic disappear into the grey market or devise a strategy for attracting it back”.

Chapter 6

Issues for Consultation

1. Should a separate category of License named **"IP Telephony Service Provider (ITSP)"** be created to make IP Telephony Calls to PSTN/PLMN network within the country and abroad that is allowing origination from PSTN/PLMN and termination into PSTN/PLMN using IP based networks? Or the **ISPs** providing Internet telephony are permitted to originate and terminate Internet Telephony calls to PSTN/PLMN within the country? If yes, what are the modifications required in the current regulatory framework? Provide your suggestions with justifications.

2. Should the number of Licenses for ITSP originating and terminating to PSTN/PLMN be limited or fully open for licensing such as ISP license? Provide inputs with justifications.

3. Whether allowing ITSP to provide IP Telephony to PSTN/ PLMN within country will raise issues of non-level playing field? If so, how can they be addressed within present regulatory regime? What should be the License fee and/or Eligibility criteria for ITSP licensee? Provide your suggestions with justifications.

4. ITSP would require interconnection with PSTN/PLMN network for IP Telephony calls to PSTN/PLMN. Could you please suggest Model/ Architecture/ Point of Interconnection between ITSP and PSTN/PLMN?

5. What will be the minimum number of Point of Interconnection required and how should they be located? Please suggest with justifications.

6. Could you please suggest any changes that would be required in the existing Interconnection Guidelines to enable growth of IP Telephony? Give your suggestions/comments with justification to provide affordable services to common masses?

7. What should be the numbering scheme for the ITSP keeping in view the limited E.164 number availability and likely migration towards Next Generation Networks? Please give your suggestions with justifications.

8. Should the regulator start charging a very small amount, say 1 NRs. per number per year, for allocating the numbering resource to ITSP so as to ensure maximum utilization of numbering resource and discourage reservation of the number space.

9. Is there a need to mandate QoS to ITSPs providing IP telephony interconnecting PSTN/PLMN within country? Please give your suggestions with justifications.

10. Should it be made mandatory for ITSP to provide the Emergency number dialing services? If so, Should option of implementing such emergency Number dialing scheme be left to ITSPs providing IP telephony? Please give your suggestions with justifications.

11. Is there any concern and limitation to facilitate lawful interception and monitoring while providing IP telephony within country? What will you suggest for effective monitoring of IP packets while encouraging VoIP/Internet telephony? Please give your suggestions with justifications.

12. Is there a need to regulate and mandate interoperability between IP networks and traditional TDM networks while permitting IP telephony calls to PSTN/PLMN within country through ITSPs? How standardization gap can be reduced to ensure seamless implementation of future services and applications? Please give your suggestions with justifications.

13. VoIP terminals may need separate powering of terminals and therefore the service may be interrupted at the time of power cut-offs. A requirement for in-line powering of terminals may put an enormous burden on VoIP providers and may slow down its development. Therefore, at this point of time, do you think that this requirement be relaxed or need to be imposed? Kindly suggest with justifications.

14. What provisions can be made for disabled users so that they can equally benefit from the VoIP/Internet Telephony? Please suggest with justifications.

END