Telecom Regulatory Authority of India

Consultation Paper

on

Internet Telephony (VoIP)

New Delhi

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Telecom Regulatory Authority of India
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Stakeholders are requested to furnish their comments to the Advisor (Broadband & Policy Analysis), TRAI by 21/07/2016 and counter comments by 04/08/2016. Comments and counter comments would be posted on TRAI's website www.trai.gov.in. The comments/counter comments in electronic form may be sent by e-mail to broadbandtrai@gmail.com.

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CHAPTER-I

Introduction

1.1   Since the 1960's when digital voice communication first emerged, the Public Switched Telephone Network (PSTN) has been supported worldwide as the primary means of voice communication. The PSTN is a connection-oriented, circuit-switched network in which a dedicated channel (or *circuit*) is established for the duration of a communication. Originally transmitting only analog signals, the PSTN ultimately switched to digital communication, which offered solutions to the attenuation, noise and interference problems inherent in the analog system. The modern PSTN uses Pulse Code Modulation (PCM) to convert all analog signals into digital transmissions at the originating network and reverses the processes in the receiving network.

1.2   Although highly rated for reliability and Quality of Service (QoS), PSTN Networks have two significant disadvantages:

   (a) Expensive bandwidth, which results in high telephone bills for individuals and businesses alike.
   (b) Inefficient use of networking channels, which results from dedicating an entire channel for each conversation.

1.3   Packet Switched Networks offer solutions to such problems and are increasingly being used as alternatives to the traditional circuit switched telephone service. IP Telephony provides alternative means of originating, transmitting, and terminating voice and data transmissions that would otherwise be carried by the public switched telephone network (PSTN).

1.4   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 Voice Over IP (VoIP). 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 faster Internet connections, widespread take-up in broadband and the emergence of new technologies.

1.5   The terms "IP Telephony", "VoIP", Internet Telephony and other variants often generates confusion as there are many different definitions used by various organizations. Some use them interchangeably while others give them distinct definitions. Further confusion is caused by using the terms to refer to both the IP-based technologies and the services that are enabled by these technologies.

1.6   Initially, there were two major categories for voice transmission over IP networks based on type of IP network used. When voice is transmitted over public Internet, it is termed as Internet Telephony. Similarly when voice is transmitted over managed IP networks, it is termed as Voice over IP (VoIP). Internet Telephony can be deemed to be a subset of Voice over IP, in the sense that, when voice is carried over a IP network it can be termed as Voice over IP. And if the IP network in this case is the public Internet then it can be called Internet telephony. The primary difference between voice services on managed and unmanaged IP Networks is in quality of speech. However this difference is getting narrower with technological advancement, new coding techniques and availability of higher bandwidth broadband connections.

1.7   The high costs of maintaining legacy networks alongside the requirement to upgrade to intelligent networks with inherent monitoring and adaptive capabilities are the key reason for growing adoption of IP based Network. Consumer VoIP applications can run over a range of

devices, offering flexibility towards seamless communications. For some operators, IP-based transmission is the first step in implementing NGN strategy, although true NGN is a broader concept that involves specific QoS guarantees and generalized mobility not offered by most types of VoIP.

1.8 Still, some existing operators may be reluctant to introduce VoIP, because they already offer voice services over the PSTN/PLMN. Perhaps understandably, they do not wish to cannibalize their higher-margin services offerings. However, the reality is that convergence, in the form of VoIP services, is redefining markets and blurring boundaries between networks and content.

1.9 The ICT sector is developing rapidly. Technological advances are making new services, and new modes of service delivery, possible. In future, Internet will be the primary medium through which converging voice and data services will flow. As a result, market structure, business models, and commercial arrangements for interconnection amongst operators are changing. Internet telephony, or Voice over the Internet Protocol (VoIP) enable users to make real time voice calls, transmitted over the Internet (rather than using traditional circuit switched telephone networks). VoIP enables network operators, service providers, and consumers to make significant savings, by reducing the underlying costs of a telephone call. VoIP uses network resources much more efficiently than conventional telephone service, reducing the costs of providing a call (albeit with the loss of some call quality and service features), and, creating opportunities for regulatory arbitrage that enable TSPs and consumers to reduce or avoid call charges. The volume of VoIP traffic is growing rapidly and the potential exists for packet switched, Internet Protocol networks to become the primary medium for most voice and data services.

1.10   Voice over Internet Protocol (VoIP) is an example of an innovative and disruptive technology. VoIP demonstrates that the basic premise of traditional voice telephony – the network and voice services must be owned and operated by the same firm – is no longer relevant. VoIP is disrupting the pre-existing business plans of traditional telephone service providers and is being introduced by service providers outside the traditional community. For instance, Google launched its Google Voice service in March 2009. Rather than own or operate any part of the underlying network, Google simply offers an application that gives users one phone number for all of their phones, provides free long distance within the United States and low international calling rates.

1.11   Convergence is primarily driven by increasing processing power, high capacity memory storage devices, reduced price, lesser power requirement and miniaturization of the devices. High-speed data transfer is now possible which is necessary for delivering innovative and advanced multimedia applications. Recent trends indicate that Telecom operators are adopting converged platforms to deliver multimedia rich applications containing voice, video and data.

1.12   Presence of unified IP based backbone and the benefits associated with the converged telecom access scenario has enabled the service providers world over to launch more and more converged services such as Internet Telephony, IPTV, Mobile TV etc. The separation of service provisioning and its management from the underlying network infrastructure in packet based networks is further increasing the acceptability of IP based Networks. It is now possible to separate provision of service contents, configuration and modification of service attributes regardless of the network catering such service. There has been enough evidence to suggest that in future IP networks will play much important role and

may ultimately encourage migration of conventional networks towards Next Generation Networks or an All IP Network.

1.13   The acceptability of IP based networks globally has facilitated growth of Broadband. However, this growth is highly dependent on availability of innovative IP based services and their affordability. Telecom service providers across the world are realizing benefits of carrying the TDM traffic over IP based Network in their backbone and access networks. Internet Telephony is considered to be one of the front-runner IP based converged service which is transmission of voice over IP based Network.

1.14   The existing licensing framework has been effective and has contributed to growth of telecom sector. However fast technological development, convergence of networks, services and end-devices is blurring the boundaries of scope of services among different licenses. Rapid changes are taking place worldwide with respect to business models, service delivery platforms and regulatory frameworks to meet the challenges posed by the convergence.

1.15   This Consultation Paper is divided into five chapters. The first chapter introduces the background in which this consultation is being initiated. Chapter - II deals with VoIP technology; Chapter - III presents current Regulatory and Licensing Framework for Internet Telephony; and Chapter - IV deals with Regulatory Issues and their implications. Chapter - V lists the issues for consultation.

CHAPTER II

VoIP TECHNOLOGY

2.1   The Internet is often characterized as being a packet-switched network. 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

2.2   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.

2.3   Traditional telecommunication operators are now moving beyond the public switched telephone network (PSTN) into IP-based, full-service networks, which are generally known as Next Generation Networks (NGNs). TSP can use these NGNs to deliver a package of voice, data and video offerings, all using the same core network hardware. Following the PSTN/PLMN model, many operators want to control the entire network value chain – in other words, they want to build end-to-end networks, including trunking and access elements. This means that many NGNs are deployed with control and service-layer functions that resemble the

closed systems of PSTN/PLMN operations. These types of networks can be referred to as the closed network model.

2.4   Meanwhile, Telecom Service providers (TSPs) who are not having full fledged networks or not having own large subscriber base may also want to compete head-on with existing TSPs by offering their own packages of voice (often VoIP), video and data. This model however, more closely complements and resembles the open Internet, with the ―intelligence and control of the network decentralized and powered by intelligent terminal equipment (i.e. computers, handsets or set-top boxes). This model can be termed as the open network model.

2.5   Currently we are at an evolutionary stage that features both models: The operator-managed, closed network model, which is successor of the legacy, public-switched telephone network (PSTN); and The open network model. For regulators this raises several questions. Can these different types of networks coexist? Can they interconnect? How will they evolve? The answers to these questions are important because of the value that can be unlocked through interconnection and the resulting ubiquity of information and content.

2.6   The **IP Multimedia Subsystem** or **IP Multimedia Core Network Subsystem** (**IMS**) is an architectural framework originally developed by 3GPP to support convergence and new services in the network. To ease the integration with the Internet, IMS uses IETF protocols wherever possible, e.g., SIP (Session Initiation Protocol). According to the 3GPP, it aids the access of multimedia and voice applications from wireless and wire-line terminals, i.e., to create a form of fixed-mobile convergence (FMC). This is done by having a horizontal control layer that isolates the access network from the service layer. From a logical architecture perspective, services need not have their own control functions, as the control layer is a common horizontal layer.

2.7 The consumer can connect to IMS in various ways, most of which use the standard IP. IMS terminals (such as mobile phones and computers) can register directly on IMS, even when they are roaming in another network or country (the visited network). The only requirement is that they can use IP and run SIP user agents. Fixed access (e.g., Digital Subscriber Line (DSL), cable modems, Ethernet), mobile access (e.g. WCDMA, CDMA2000, GSM, GPRS) and wireless access (e.g., WLAN, WiMAX) are all supported. Other phone systems like plain old telephone service (POTS—the old analogue telephones), H.323 and non IMS-compatible systems, are supported through gateways.

2.8 Underlying technology i.e. Voice over IP (VoIP) is a group of technologies used for the delivery of voice and multimedia sessions over IP (Internet Protocol) networks. In VoIP, the signaling that controls the session (e.g. a voice call) is distinct from the audio stream that carries the voice content. Hence, VoIP protocols are classified either as signaling or media protocols.

2.9 Examples of signaling protocols include
  (a) Session Initiation Protocol (SIP) – a widely used application layer protocol for creating, modifying and terminating sessions with one or more participants. SIP typically makes use of the Session Description Protocol (SDP) to negotiate media parameters for a call.
  (b) H.225.0 – part of the H.323 stack (a family of VoIP protocols standardized by ITU-T); used to establish, control and end a call.

2.10 An example of a media protocol is the Real-time Transport Protocol (RTP), which is used by nearly every VoIP stack today. While RTP is used to transport the actual voice and video data, its sister protocol, the RTP Control Protocol (RTCP), provides feedback on the quality of media distribution in a call.

2.11 Following diagram describes the SIP system architecture, call flows in a VoIP system, and VoIP-to-PSTN bridging.
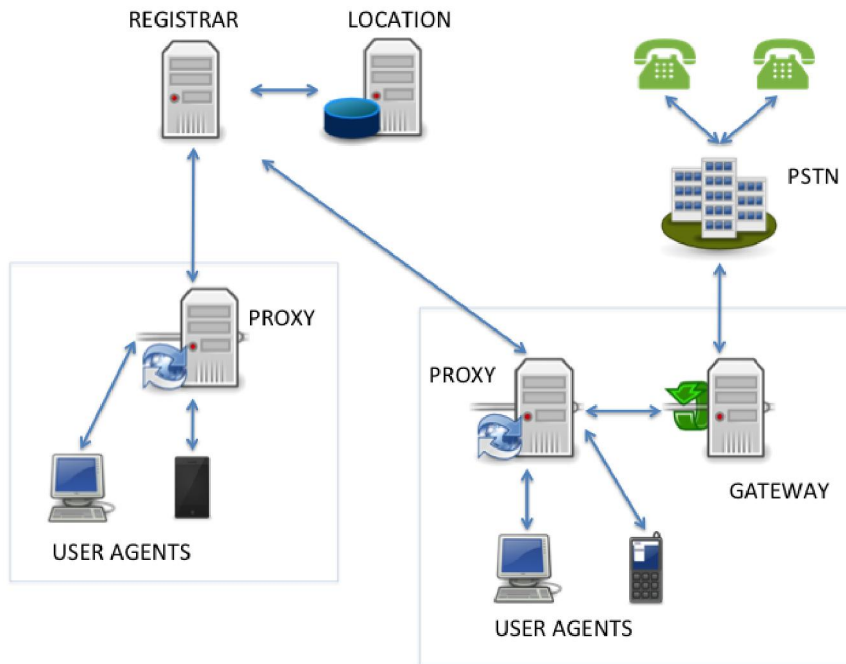
SIP System Architecture

Figure 2.1   SIP System Architecture

2.12 The main elements involved in a SIP system are:

- **User Agents** – A user agent (UA) is an endpoint that originates or receives calls on a SIP network. Examples include a SIP phone, a PC or a smart phone with a SIP app installed, or a SIP-to-PSTN gateway. SIP user agents are usually known as SIP clients.

- **Proxy Servers** – a proxy server routes SIP requests and responses on behalf of user agents. Its job is to ensure that a SIP message is sent to another entity closer to the target user. Proxies can also enforce policies, such as to determine whether a user is allowed to

make a call. In practice, every SIP user agent needs at least one proxy server (also known as a "home proxy") that acts on its behalf. The home proxy is either manually configured in a user agent or discovered through DHCP.

- **Registrars** – A user agent sends a registration request to a SIP registrar when its available to receive calls on a SIP network. A registrar binds one or more IP addresses[1] to the SIP URI[2] of the registering agent and stores this binding in a location server.

- **Location Servers** – A location server stores all the aforementioned bindings. Typically a location server is co-located with a registrar server. Location servers are queried by SIP proxies in order to locate the recipient of a SIP call.

- **SIP Gateways** – SIP gateways allow SIP users to communicate with users on a different voice network (e.g. H.323 or PSTN). They do this by translating SIP messages to that of the other network and vice-versa.
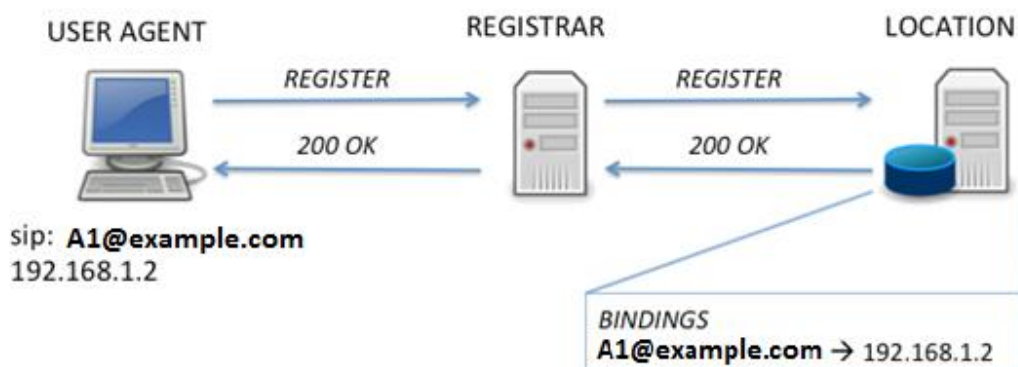
SIP Registration

Figure 2.2 SIP Registration

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[1] In many instances an IP address is insufficient to reach a SIP user agent (because of network elements like NATs); registrars will bind additional info in these instances.
[2] A SIP URI looks similar to an email address (e.g. `sip:A1@example.com`)

2.13 To register itself, a SIP user agent sends a REGISTER request to the registrar, which binds the user's IP address to the SIP URI, and stores this binding in the location server.

SIP Calls

2.14 In a SIP call, SIP messages are relayed through one or more proxy servers, which make use of location servers to locate the recipient. The media stream bypasses proxy server(s) altogether – wherever possible user agents will directly send media traffic to each other. The most common SIP arrangement is illustrated in the following figure 2.3, and is known as a **SIP trapezoid**.
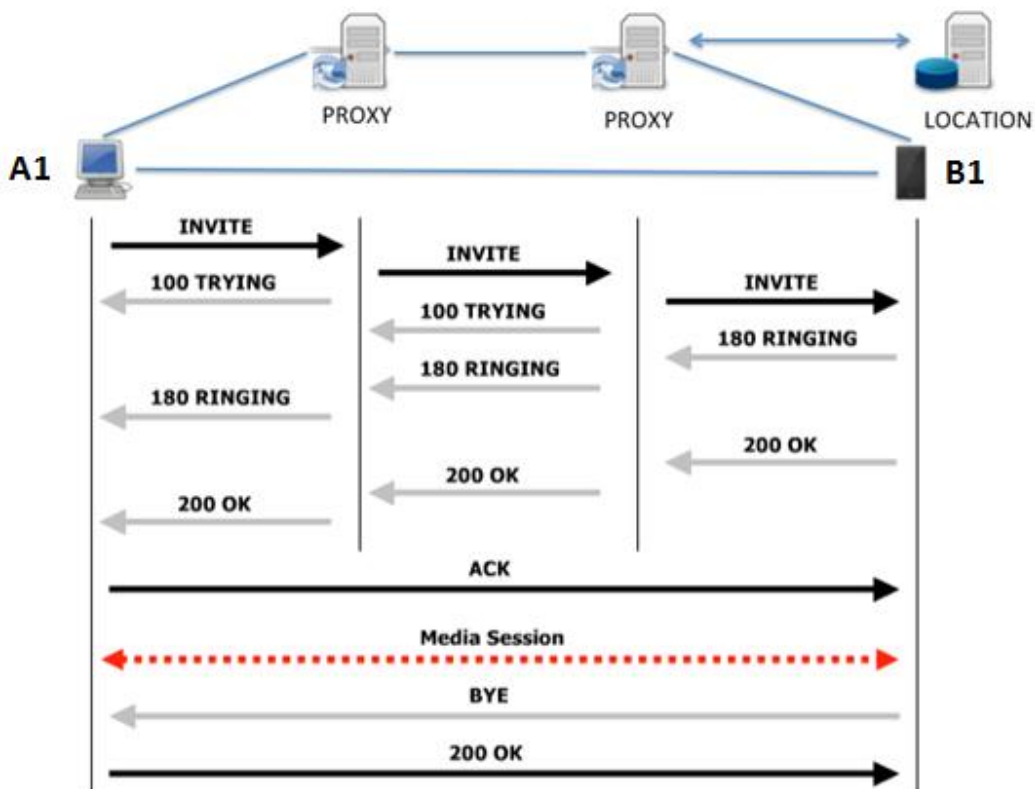

Figure 2.3 SIP Calls

2.15 In this arrangement, A1 (sip:A1@example.com) and B1 (sip:B1@example.net) are two SIP users. A1 initiates a call with B1 using her SIP user agent. Also shown are A1 and B1's home proxies. The sequence of messages sent in a SIP call is as follows:

  i.   An INVITE goes out from A1's user agent to her proxy server, which responds back with 100 TRYING to indicate that it has received the INVITE and is trying to locate B1.

  ii. A1's proxy locates B1's proxy (possibly by performing a particular type of DNS lookup), and forwards it the request.

  iii. B1's proxy, on receiving the INVITE, responds back with 100 TRYING. Meanwhile, it looks up B1 in the location server, and forwards the request to its UA.

  iv. B1's user agent starts ringing and relays back a 180 RINGING response to A1 as soon as it receives the INVITE.

  v.   Once B1 picks up the call a 200 OK is sent to A1.

  vi. An acknowledgement (ACK) of the 200 OK response is sent by A1's user agent to B1. If the proxy configuration permits it, the ACK might be sent directly to B1's UA, bypassing the proxies.

  vii. At this point, the RTP media stream is established between A1 and B1's user agents. This is a peer-to-peer (P2P) stream[3] and the proxies are not involved in its path. Parameters for the RTP are negotiated via SDP messages encapsulated inside the INVITE request and its responses.

  viii. As soon as either of B1 hangs up the call a BYE is sent from its UA to A1's UA.

  ix. A1's UA responds with a 200 OK to the BYE request and the call ends.

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[3] In certain NAT configurations two hosts will not be able to setup a P2P connection, in this case they will need to relay the media stream through a TURN server.

SIP to PSTN Bridging

2.16 In the PSTN world, ISDN User Part (ISUP) is used to relay call signaling information between switches, whereas Time-division multiplexing (TDM) channels are used to transmit voice signals.

2.17 Bridging calls from SIP to a PSTN network requires the use of a SIP-to-PSTN gateway. The gateway acts as a SIP user agent and performs two functions:

1. Translates SIP/SDP messages to ISUP messages
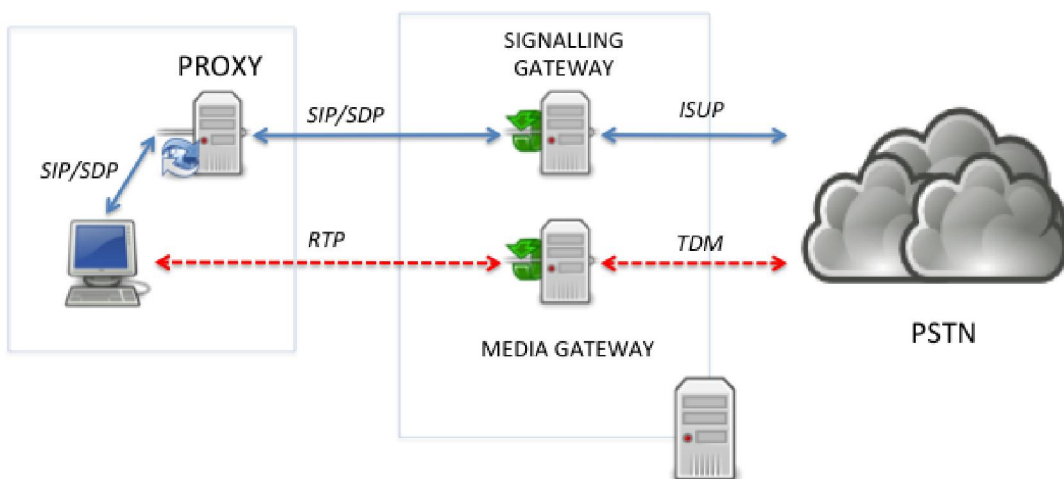2. Translates an RTP media stream to a TDM channel

Figure 2.4 SIP to PSTN Bridging

2.18 In order to make calls from a SIP device to a PSTN number, there needs to be a way to translate a SIP URI to a PSTN phone number. Usually this is done by replacing the user part of a SIP URI with the E.164 representation of a phone number (e.g. sip:+91987654321@example.com).

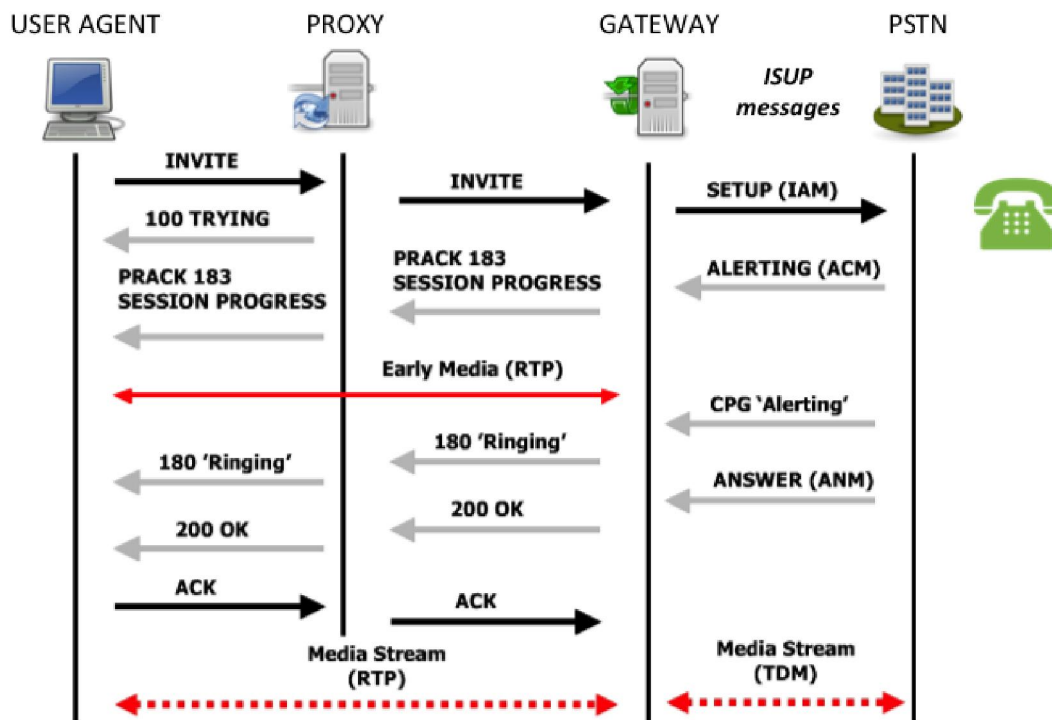2.19 The call flow for a SIP to PSTN call looks as below:

Figure 2.5 SIP to PSTN Call Flow

2.20   To allow the calls in the reverse direction i.e. PSTN to SIP, the service
provider allocates a PSTN number for the SIP user. PSTN calls made to
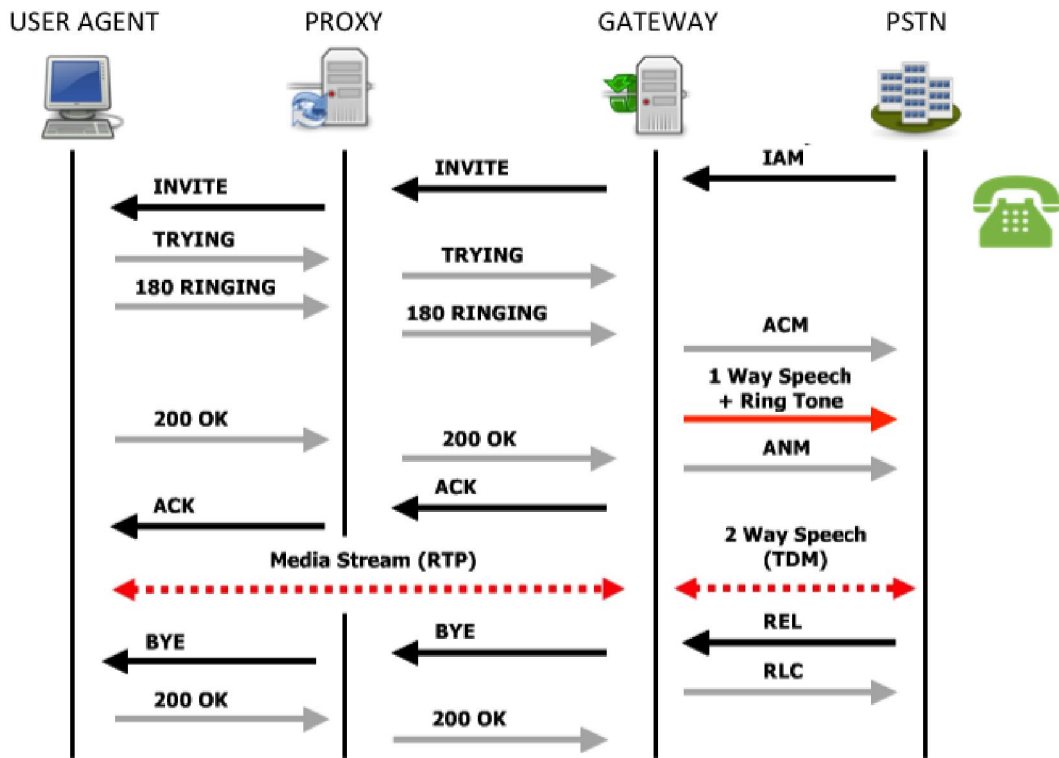this number are routed to the gateway and subsequently to the user's
device.

Figure 2.6 PSTN to SIP Call Flow

CHAPTER III

CURRENT REGULATORY AND LICENSING FRAMEWORK

3.1   Internet services in India were first launched in 1995 by erstwhile VSNL then a Government owned PSU. However at that time Internet telephony in any form was not permitted. Later in November 1998, the Government issued new guidelines for Internet services and ISP licenses to private operators. Even at this stage Internet telephony was not envisaged as a service.

3.2   In the New Telecom Policy 1999 (NTP 1999) announced by the Government in March 1999, various steps were taken to support the Internet services however even at this stage Internet telephony was not allowed.

3.3   Later, Department of Telecom announced the guidelines for opening of Internet telephony w.e.f. 1st April 2002 with restricted use of Internet Telephony. Existing ISPs were permitted to offer Internet telephony services only after signing the amended ISP license called Internet Telephony Service Provider (ITSP) license. Internet telephony was permitted only in limited way, as there were restrictions on the type of the technology and devices, which could be used. ITSPs were not permitted to have connectivity with PSTN/PLMN. Initially provisioning of Internet telephony service did not envisage any financial implications (no additional entry fee or license fee). DoT imposed a license fee of 6% of AGR earned from Internet telephony by ITSPs with effect from 1st January 2006.

3.4   In March 2006, Unified Access Service Providers (UASPs) were permitted to provide Internet telephony. In August 2007, all ISPs were permitted to

provide Internet telephony and separate category of Internet Telephony Service Providers (ITSPs) was done away with. License fee of 6% of AGR was imposed on all ISPs except on the revenue earned from provisioning of pure Internet access services.

3.5   The present regulatory framework permits Unified Access Service Licensee (UASL), Cellular Mobile Telecom Service (CMTS) licensees and Unified Licensee to provide voice services within country. They have been permitted to provide unrestricted Internet Telephony. The relevant clauses of UASL and CMTS licenses are reproduced below:

Clause 2.2 (a)(i) of UASL

"... *Access Service Provider can also provide Internet Telephony, Internet Services and Broadband Services. If required, access service provider can use the network of NLD/ILD service licensee.*"

Clause 2.1 (a) of CMTS License

"... *The Licensee can also provide Internet Telephony, Internet Services and Broadband Services. If required, the Licensee can use the network of NLD/ILD service licensee ...*".

clause 2.1 (a) (i) of UL

"......The Licensee can also provide Internet Telephony, Internet Services including IPTV, Broadband Services and triple play i.e voice, video and data. While providing Internet Telephony service, the Licensee may interconnect Internet Telephony network with PSTN/PLMN/GMPCS network....."

3.6   Internet telephony in the above license has been defined as *""Internet Telephony" Means "Transfer of message(S) including voice signal(S)* ***through public Network".***

3.7 Internet Telephony has been also permitted to Internet Service Providers (ISPs) in restricted manner under ISP licensing conditions issued by Government in October 2007. As per ISPs licensing provisions, there is no restriction on PC-to-PC Internet Telephony calls. PC or adapter can be used to call PSTN/PLMN abroad; however Internet Telephony calls from such devices to PSTN/PLMN in India are not permitted under ISP license. ISPs are also not allowed to have interconnection with PSTN/PLMN networks.

3.8 The scope of services as stated under Clause 2.2(ii) of Part II in ISP License for provision of Internet Services is reproduced below:

"Internet telephony means a service to process and carry voice signals **offered through Public Internet** *by the use of Personal Computers (PC) or IP based Customer Premises Equipment (CPE) connecting the following:*

a) *PC to PC; within or outside India*

b) *PC / a device / Adapter conforming to standard of any international agencies like- ITU or IETF etc. in India to PSTN/PLMN abroad.*

c) *Any device / Adapter conforming to standards of International agencies like ITU, IETF etc. connected to ISP node with static IP address to similar device / Adapter; within or outside India.*

Explanation: Internet Telephony is a different service in its scope, nature and kind from real time voice service as offered by other licensed operators like Basic Service Operators (BSO), Cellular Mobile Service Operators (CMSO), Unified Access Service Operators (UASO)."

Addressing under Clause 2.2 (iv):

"Addressing scheme for Internet Telephony shall only conform to IP addressing Scheme of Internet Assigned Numbers Authority (IANA) exclusive of National Numbering Scheme / plan applicable to subscribers of Basic / Cellular Telephone service. Translation of E.164 number / private number to IP address allotted to any device and vice versa, by the licensee to show compliance with IANA numbering scheme is not permitted.

Interconnection under Clause2.2 (v):

"The Licensee is not permitted to have PSTN/PLMN connectivity. Voice communication to and from a telephone connected to PSTN/PLMN and following E.164 numbering is prohibited in India".

3.9 In year 2007/08, when unrestricted Internet Telephony for ISPs were deliberated, the main argument given by TSPs was that they have paid huge entry fee and have made heavy investments to create infrastructure. Opening up of unrestricted Internet telephony to ISPs will impact their business model to a great extent as they apprehend reduction of voice traffic on their network. They argued that as access providers are subjected to higher regulatory levies, huge upfront entry fee and have sunk-in investments on infrastructure development, their overheads will be higher as compared to ISPs. As per them it will disturb level playing field among different licensees. They also argued that infrastructural developments can be impacted due to reduced margins if ISPs start unrestricted Internet telephony. Access providers were of strong opinion that in case ISPs want to offer unrestricted Internet telephony then ISPs should also pay the same entry fees and levies as paid by access service providers.

3.10 After due consultation process and detailed deliberation, TRAI on 18.08.2008 recommended to the Government that ISPs may be

permitted to provide Internet telephony calls to PSTN/PLMN and vice-versa within country and necessary amendments may be made in the license provisions. However, Government did not accept these recommendations of TRAI.

3.11 Since then, there have been significant changes in licensing framework of the country. Now allocation of Spectrum has been delinked with the grant of License. Unified license has been introduced with entry fee of Rs 15 crore for whole country. Therefore any ISP or new service provider who is willing to provide unrestricted Internet Telephony can obtain Unified License with authorization for Access services. Further, some existing access licensee are also planning to start Internet Telephony service. Unrestricted internet telephony to Unified Licensee only with authorization of access services will also ensure that only serious players would provide Internet Telephony. Therefore it is for the consideration of stakeholders that whether there is still need for permitting unrestricted telephony to Internet service providers (ISP) or they may be facilitated to migrate to Unified License with authorization of Access services if they wish to provide unrestricted Internet Telephony.

Question 1:
What should be the additional entry fee, Performance Bank Guarantee (PBG) and Financial Bank Guarantee (FBG) for Internet Service providers if they are also allowed to provide unrestricted Internet Telephony?

CHAPTER IV

REGULATORY ISSUES AND IMPLICATIONS

4.1   With the advancement of technology, Internet Telephony has now become similar to conventional telephony and these providers compete directly with the existing PLMN/PSTN TSPs. Therefore it eventually has to be decided what aspects of conventional telephony regulation should apply to Internet Telephony service. To encourage Internet Telephony services in the country, issues such as allocation of telephone numbers, Interconnection, Interconnection Usage charges and access to Emergency service need to be addressed urgently.

Interconnection

4.2   Interconnection is the most important aspect of the telecom network. Interconnection framework since beginning has been designed to cater for circuit switched networks and regulations are framed to ensure that licensees such as access providers, NLDO and ILDOs interconnect with each other as per National routing plan of the country. Internet is global phenomenon and there is no boundary such as service area or country in internet domain. Therefore applying same rules of conventional telephony for Internet Telephony may not be desirable as it allow a TSP to pass advantage of cost effectiveness of VoIP technology to the consumers.

4.3   Recently BSNL has proposed to introduce Fixed Mobile Telephony (FMT) value added services for its customers. BSNL informed that FMT service will be an extension of their fixed line service using IMS based NGN core switch and IP based access network. Their Subscribers will be assigned a SDCA based number from the number series allocated to BSNL for their fixed line service. Subscriber roaming anywhere in the country or

even abroad can avail this service using an App installed on any device, including its mobile phone, once a subscriber has registered in any SDCA of the country wherein service is being offered by BSNL. FMT service is a voice call using the IP access and NGN switch of BSNL landline for the call routing. Internet access is required to access this service. FMT service essentially needs access to reach BSNL's NGN equipment for registering SIP subscriber for making voice call which means FMT call is not possible without internet.

4.4 A subscriber of such a service will be able to make or receive calls as long as he has access to Internet. Schematic diagram of a call is shown in following diagram:
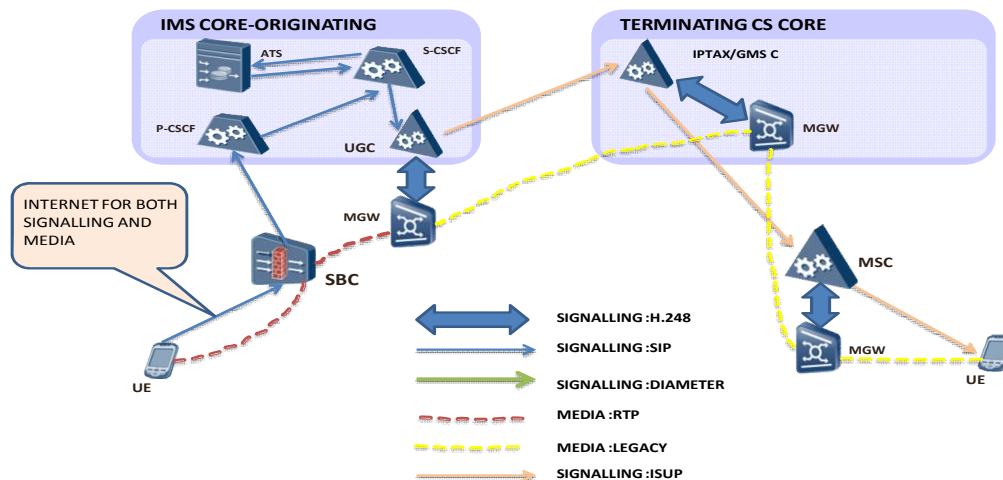

Figure 4.1 Schematic Diagram of Fixed Mobile Telephony Calls

4.5 Other TSPs are also either deploying IMS based network or are planning to migrate to IMS based network. As mentioned earlier, the consumer can connect to IMS in various ways, most of which use the standard IP.

IMS terminals (such as mobile phones and computers) can register directly on IMS, even when they are roaming in another network or country (the visited network). The only requirement is that they can use IP and run SIP user agents. Authentication or routing of call can be done either through application or through SIM in case of Mobile.

4.6 Subscriber may be anywhere in India or even abroad, when he makes the call and the call is routed on public internet upto one of the node of IMS core or to the SIP server as the case may be and finally routed to destination as per national routing plan. Therefore when call is on public internet, it is not being routed through NLD/ILD though it may be traveling through access service areas to reach the node of IMS core or SIP server. In this case inter-service area call is travelling through public internet to reach node of IMS core or SIP server without NLDO.

4.7 In case of Internet Telephony, voice services are simply software applications riding over the internet. Converging technologies and markets make conventional approaches to interconnection charging unsustainable. Many technology forecasters predict that in future voice telephony will migrate completely from circuit-switched telephony to packet switched Technology. Once this happens, Internet interconnection and pricing models may replace the current arrangements. However, in the interim, Internet Telephony network operators will need to interconnect with existing network operators' PSTN/PLMN network.

Transit of Calls

4.8 Unified License also provides that Licensee may also enter into mutual agreements with other Unified Licensee for carrying its intra-Circle Long Distance traffic. Relevant clause is reproduce below:

"2.2 Licensee may carry intra-circle long distance traffic on its network. However, subject to technical feasibility, the subscriber of the intra-circle long distance calls, shall be given choice to use the network of another Licensee in the same service area, wherever possible. The Licensee may also enter into mutual agreements with other UL Licensee (with authorization for access service)/ other Access service licensee/National Long Distance Licensee for carrying its intra-Circle Long Distance traffic."

4.9 Further, Unified License with authorization for NLD services provides that the Licensee may also carry <u>intra-circle switched traffic</u> where such carriage is with mutual agreement with originating access service provider. Relevant clause is reproduce below:

"2.1 (a) The NLD Service Licensee shall have the right to carry inter-circle switched bearer telecommunication traffic over its national long distance network. The Licensee may also carry intra-circle switched traffic where such carriage is with mutual agreement with originating access service provider."

4.10 These clauses provides flexibility to a TSP to transit traffic of other TSP within service area thus avoiding need for large number of interconnection points to start service. A small Internet telephony service provider may connect to only one TSP and this TSP can transit/carry traffic to other TSP as well. Presently, transit charge is in the form of ceiling, ITSP can negotiate transit charge with any TSP who is willing to transit its traffic to other TSPs. As per IUC regulations, transit charge should be less than Re.0.15 (Fifteen paise only) per minute and, can be decided by the concerned service providers through mutual commercial arrangement.

Question 2:
Point of Interconnection for Circuit switched Network for various types of calls is well defined. Should same be continued for Internet

Telephony calls or is there a need to change Point of Interconnection for Internet Telephony calls?

Question 3:
Whether accessing the telecom services of a TSP by the subscriber through public Internet can be construed as extension of fixed line or mobile services of the TSP? Please provide full justification in support of your answer.

Question 4:
Whether present ceiling of transit charge needs to be reviewed? In case it is to be reviewed, please provide cost details and method to calculate transit charge.

Interconnection Usage Charges

4.11 The present framework prescribes Interconnect Usage Charges (IUC) among service providers for various types of calls. This facilitates settlement of the interconnection charges smoothly and curbs the possibility of the disputes. As per the present IUC framework Rs 0.14/- per minute is the termination charge of the domestic calls on wireless network, if calls are originating from domestic wireless network. For rest of domestic calls termination charge has been set to zero. For international calls, termination charges has been prescribed as 53 paisa per minute. IUC regulation has prescribed 35 paisa per minute as ceiling for the carriage charges. The IUC framework has been very effective in the past as it succeeded in overall regulating interconnection charges yet leaving lot of scope to service providers for bringing new tariff packages and effective competition among the service providers. This time tested IUC framework can easily be applied to the Internet

Telephony calls except that additionally termination charges for calls originating/terminating as Internet Telephony calls has to prescribed.

4.12 The most important issue with Internet Telephony calls is that it is very difficult especially by the terminating operator, to identify the originating network (if same number is used for Internet Telephony and PSTN/PLMN) or country of the call. Difference in termination charge between Internet Telephony and PSTN/PLMN will lead to the possibility of arbitrage and the impact on the market can be substantial. Further, even when a PSTN operator is able to detect Internet Telephony traffic, it may not be able to differentiate between domestic and international Internet Telephony calls.

4.13 Internet Telephony providers require access to the PSTN to terminate calls to recipients who do not subscribe to the Internet Telephony provider's service. Such interconnection typically occurs between a Internet Telephony operator's gateway and the PSTN operator's Tandem Switch closest to the call.

4.14 Internet Telephony has significant implications for interconnection charging. To have sustainable charging regime, there may be a need to have uniform charge to avoid regulatory asymmetries that treat similar services differently based on the technology used to provide the services As more services are delivered as packets over digital networks, minutes of use are no longer an important cost driver. Changes in technology in telecommunications network is rapidly changing the cost structures of telecom network and per-minute pricing may become an inefficient cost recovery mechanism.

4.15 Cost Drivers for VoIP Per-minute cost recovery has a number of weaknesses in a VoIP world. Call duration has no meaningful relationship to the costs of a VoIP call. As VoIP traffic increases,

interconnection charges based on bandwidth used would better reflect underlying cost drivers, and would be more consistent with economic efficiency. One way could be that where VoIP operators provide a service that is functionally equivalent to conventional telephony, treating Internet Telephony providers in the same way as conventional service providers will remove arbitrage opportunities. Generally, VoIP operators do not receive any compensation from PSTN operators for terminating calls that originate on the PSTN. As more traffic migrates to VoIP, a new approach to interconnection pricing may be needed. Any new approach to interconnection pricing should: encourage efficient competition and the efficient use of, and investment in, telecommunications networks, treat technologies and competitors neutrally, allow innovation and minimize regulatory intervention and enforcement, consistent with the general trend toward less regulation wherever possible.

4.16 Termination charge issue gets further complicated as there is different termination charges between wire line and wireless network. Basically, at present termination charge is @14 paise per minute for domestic calls between wireless to wireless and for the rest of domestic calls it is zero.

4.17 Internet telephony call may terminate or originate either from wire-line or wireless but basic important difference is that voice call should be accessed through public Internet. The Internet telephony is different when compared to present PSTN/PLMN. It requires minimum threshold speed Internet connection for good speech quality. The incoming calls shall be feasible only when broadband is connected and functioning well. Hence, when call is terminating on Internet Telephony subscriber, subscriber is already paying for data charges and Internet Telephony service provider is simply providing voice service through software. Therefore this cannot be truly called as calling party pay (CPP) regime as

called subscriber is also paying for terminating the call in the form of data charges.

4.18 There are certain challenges that the existing regime imposes on Internet telephony calls. These are described in the following paragraphs

4.19 Mobility: An Internet telephony subscriber as per the license uses the public Internet to make a call from his Internet telephony terminal, which then traverses over the public Internet to the SIP gateway or IMS node of the licensee. Since the call travels over the public Internet, an Internet delivery subscriber may actually make the call from anywhere. Hence a subscriber of say the Mumbai circle, could be sitting in Bangalore and still make and receive Internet telephony calls since the last mile would traverse over the public Internet. National long-distance calls: when an Internet telephony subscriber makes a long-distance call from his Licensed Service Area (LSA) to another LSA, the existing interconnection regime manages this scenario without any issues. For instance if a Mumbai circle subscriber sitting in Mumbai is making a call to a Delhi subscriber, the call will travel between Mumbai and Delhi using an NLDO as is required under the license. However if a Mumbai subscriber were to make a Internet telephony call from Bangalore, to Delhi, the call would travel from Bangalore to Mumbai over the public Internet and then from Mumbai to Delhi via the NLDO

4.20 International calls: when an Internet telephony subscriber makes or receives an international call, the existing interconnection regime manages this scenario without any issues since these calls would travel over an ILDO. However if an Internet telephony subscriber, is located outside the country, for example in the United Kingdom, and chooses to make a call to someone in New Delhi, in this case call could travel from the United Kingdom to Mumbai over the public Internet and only then over the PSTN networks, thus potentially bypassing an ILDO for carrying

inbound traffic. This scenario is not prevented by any Internet telephony providers worldwide and in fact service providers such as Verizon and T-Mobile actually provide a hybrid service allowing their subscribers to use either Wi-Fi or roaming networks to make calls when they are traveling. This is to the benefit of consumers and as such represents advantages derived from using Internet telephony. In view of the above, Stakeholders' views are solicited on the following:

Question 5:
What should be the termination charge when call is terminating into Internet telephony network?

Question 6:
What should be the termination charge for the calls originated from Internet Telephony Network and terminated into the wire-line and wireless Network?

Question 7:
How to ensure that users of International Internet Telephony calls pay applicable International termination charges?

Question 8:
Should an Internet telephony subscriber be able to initiate or receive calls from outside the SDCA, or service area, or the country through the public Internet thus providing limited or full mobility to such subscriber?

Question 9:
Should the last mile for an Internet telephony subscriber be the public Internet irrespective of where the subscriber is currently located as long as the PSTN leg abides by all the interconnection rules and regulations concerning NLDO and ILDO?

Numbering

4.21 Numbers always play a central role in telecommunications and their importance is well recognized. A well designed numbering for any service ensures structured growth of any service. UL/UASL/CMTS allow Licensee to provide unrestricted Internet Telephony but it is not clear that whether TSP can use same numbering resource or it will be given separate numbering resource for providing Internet Telephony. Relevant clause of Unified License with regard to numbering of Internet Telephony is as follows:

"2.5 IP Address assigned to a subscriber for Internet Telephony shall conform to IP addressing Scheme of Internet Assigned Numbers Authority (IANA) only. Translation of E.164 number / private number to IP address and vice versa by the licensee for this purpose shall be as per directions/instructions issued by the Licensor. "

4.22 It is worth noting that Internet telephony can be offered without allocation of number resources from E.164 numbering plan. However, it is not possible to call an Internet telephony subscriber from an existing PSTN/PLMN network without allocation of a number, which can be recognized, by the traditional fixed and mobile telecom network. This will greatly restrict the scope and popularity of the Internet telephony services.

4.23 Identification of such Internet telephony numbers from other PSTN/PLMN numbers may be desirable. Considering distinct service features of Internet telephony, a separate series of numbers may be required for Internet telephony services irrespective of the license under which such services are being provided. Since Internet telephony supports CLI, it is desirable that Internet telephony service providers for the benefit of subscribers also provide calling line identification.

4.24 On the other hand, arguably, Internet telephony is merely a technology mechanism and medium. It per se has no impact or relevance on numbering. Convergence may actually be beneficial to consumers. For instance in the United States, no distinction is made between mobile numbers, fixed line numbers or Internet telephony numbers. Even in countries such as United Kingdom, Germany and others Internet telephony is merely a technology and does not have a special numbering block. This has tremendous advantages. For instance one of the biggest applications of this is the Google Fi service, wherein Google provides a single number to a subscriber and based on whether subscriber is at home, in a basement, out on the streets, his cell phone automatically selects the best mobile or wireless network to connect a call. This ensures much higher call quality and ubiquitous service. If Internet delivery numbers were different from fixed line and cellular numbers this type of a service would not be feasible since a mobile handset would not be able to seamlessly transition between networks without having to drop and reinitiate a new call with a new phone number.

4.25 In India, Number blocks are allocated separately for fixed line which is SDCA based and for Mobile which is at country level. One option could be that TSP can use same number resources and have similar restriction for Internet Telephony service with regard to mobility as it for normal voice services. This will also be consumer friendly as he can be reached or can make call with same identity irrespective of whether he is making Internet Telephony calls (if access to internet is available or it is cheaper) or normal call by same number. However, it will be possible when there is same termination charge for Internet Telephony calls.

4.26 Other way could be to allocate separate series for Internet Telephony service and all spare codes which are not being used can be allocated for Internet Telephony calls. At present these numbers cannot be used for

mobile services. If these numbers are allocated to mobile, it will have conflict with local fixed line number. If we add '0' in dialing pattern from Internet Telephony calls to/from other calls (Fixed line/Mobile), it will not have any conflict and this numbering resource which is otherwise idle can be used for Internet Telephony service. In view of the above, stakeholders are requested to comments on following:

Question 10:

What should be the framework for allocation of numbering resource for Internet Telephony services?

Question 11:

Whether Number portability should be allowed for Internet Telephony numbers ? If yes, what should be the framework ?

Access to Emergency Services

4.27 The facility to call nearest authority like police, fire station, hospital, etc has been termed as access to Emergency Service. Accurate identification of geographical location of subscriber is a must for availing emergency services. The concept of emergency number calling has changed with introduction of the mobile services. It is envisaged that accurate location of the caller will also be available to the authority (Hospital, Police, Fire-station) handling emergency situation along with emergency number calls.

4.28 Different telecom networks adopt different technologies to facilitate emergency number calling. In case of usage of Internet telephony services from a fixed location, it is possible to map the position information and route emergency calls to appropriate agency. However, one of the promising features of Internet telephony services is the

nomadic use. In the nomadic use it may be difficult to accurately map position information while originating the emergency call.

4.29 There are various technical options available to make emergency number call even using Internet telephony. One of the simplest options is to route emergency services call to appropriate geographically decentralized emergency service centres and provide them with the appropriate location information. A soft switch in such cases can effectively handle emergency number calls and provide sufficient location information, though such information may not very accurately point to subscriber's geographical location.

4.30 The prevailing International scenario to facilitate emergency number calling is different in different countries. Some of the countries have gone ahead with Internet telephony services without mandating emergency number calling facility. They have emphasized the issue of transparency and desired that Internet Telephony service provider shall inform their subscribers that Internet telephony service will not support emergency numbers calling.

4.31 In India, when subscriber calls from fixed line, the call goes to nearest police/fire station which has been mapped to corresponding location. For mobile, TSPs provide the information of SDCA to BSNL/MTNL along with CLI of calling party and call is routed by BSNL/MTNL to nearest Police station in that very SDCA.

Question 12:

Is it possible to provide location information to the police station when the subscriber is making Internet Telephony call to Emergency number? If yes, how?

Question 13:

In case it is not possible to provide Emergency services through Internet Telephony, whether informing limitation of Internet Telephony calls in advance to the consumers will be sufficient ?

Quality of Service

4.32 Quality of speech in any communication service is an important consideration. Subscribers are accustomed to the PSTN/ PLMN voice quality and expect similar quality from Internet telephony also irrespective of the technology used to provide such services. Ensuring good voice quality will therefore be necessary for ISPs providing Internet telephony. Though 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 speech quality.

4.33 The issue of consideration is whether there is a need to define QoS parameters for Internet telephony or it should be left to service providers. Both the models are prevailing world over. In some countries all Internet service providers have to match QoS parameters as defined for PSTN/ PLMN whereas in some other countries no specific QoS have been defined. Service providers are required to appraise the subscribers

about QoS before they subscribe to such services. Comments of the stakeholders are invited in this regard.

Question 14:

Is there a need to prescribe QoS parameters for Internet telephony at present? If yes, what parameter has to be prescribed? Please give your suggestions with justifications.

CHAPTER- V

Issues for Consultation

Q1: What should be the additional entry fee, Performance Bank Guarantee (PBG) and Financial Bank Guarantee (FBG) for Internet Service providers if they are also allowed to provide unrestricted Internet Telephony?

Q2: Point of Interconnection for Circuit switched Network for various types of calls is well defined. Should same be continued for Internet Telephony calls or is there a need to change Point of Interconnection for Internet Telephony calls?

Q3: Whether accessing of telecom services of the TSP by the subscriber through public Internet (internet access of any other TSP) can be construed as extension of fixed line or mobile services of the TSP? Please provide full justification in support of your answer.

Q4: Whether present ceiling of transit charge needs to be reviewed or it can be continued at the same level? In case it is to be reviewed, please provide cost details and method to calculate transit charge.

Q5: What should be the termination charge when call is terminating into Internet telephony network?

Q6: What should be the termination charge for the calls originated from Internet Telephony Network and terminated into the wire-line and wireless Network?

Q7: How to ensure that users of International Internet Telephony calls pay applicable International termination charges?

Q8: Should an Internet telephony subscriber be able to initiate or receive calls from outside the SDCA, or service area, or the country through the public Internet thus providing limited or full mobility to such subscriber?

Q9: Should the last mile for an Internet telephony subscriber be the public Internet irrespective of where the subscriber is currently located as long as the PSTN leg abides by all the interconnection rules and regulations concerning NLDO and ILDO?

Q10: What should be the framework for allocation of numbering resource for Internet Telephony services?

Q11: Whether Number portability should be allowed for Internet Telephony numbers ? If yes, what should be the framework?

Q12: Is it possible to provide location information to the police station when the subscriber is making Internet Telephony call to Emergency number? If yes, how?

Q13: In case it is not possible to provide Emergency services through Internet Telephony, whether informing limitation of Internet Telephony calls in advance to the consumers will be sufficient?

Q14: Is there a need to prescribe QoS parameters for Internet telephony at present? If yes, what parameter has to be prescribed? Please give your suggestions with justifications.

Q15: Any other issue related to the matter of Consultation.