Ortel communications
An ISO 9001 2008 COMPANY

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Telecom Regulatory Authority of India,
Mahanagar Door Sanchar Bhawan,
Jawahar Lal Nehru Marg,
New Delhi- 110002

Kind Attention : Shri Arvind Kumar, Advisor (Broadband & Policy Analysis)
Reference : Consultation Paper No. 13/2016.
Subject : Our comments in response to the consultation paper No 13/2016

Dear Sir,

At the outset we thank you for circulating a well thought out consultation paper on Internet Telephony (VOIP). It is long overdue for implementing convergent service including VOIP. We hope with this attempt we can be able to bring in a revolution in the Telecom Sector.

We are one of the convergent service provider providing last mile convergent services in the state of Odisha , Chhattisgarh, Madhya Pradesh, West Bengal, Andhra Pradesh and Telangana. We are also one of the oldest ISP license holder and providing Broadband, Leased Line, Public WiFi Hotspots, VOIP, Internet Telephony and Value Added Services.

Ortel has been a pioneer in launching Broadband Services using DOCSIS technology. Ortel launched VOIP services too. However, due to Regulatory hurdle the VOIP service could not be marketed as a full-fledged service. Nevertheless with this important initiative of TRAI, if seamless integration of VOIP network is made with traditional telecom services it can result in a revolution benefitting millions of subscribers and will give impetus to Digital India Mission.

With this background we have given our comments to the issues raised by you in your consultation paper is attached here with as Annexure.

Thanking you,

Yours faithfully,
For **Ortel Communications Ltd.**

Sd/-

LtCol. (Retd) Man Mohan Pattnaik
(CHIEF TECHNOLOGY OFFICER)

Enclosed as above.

Annexure

Consultation Paper on Internet Telephony (VoIP)

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?

Answer 1: In Ortel's opinion there should not be any additional entry fee, Performance Bank Guarantee(PBG) and Financial Bank Guarantee(FBG) for Internet Telephony Service Provider (ITSP)s due to the following reasons :-

a. Internet Telephony is a stream of data, and fully dependent on Internet Bandwidth. Without internet bandwidth the Internet Telephony can not be extended to customers for their benefit.
b. Internet Telephony does not require any type of National resources like spectrum or frequency as being used by mobile operator. Hence, Internet Telephony does not use any type of costly resources, for which a separate license is required.
c. The voice is fully initiated and extended on public internet and hence it does not require any special circuit like NLD or ILD to carry the signal separately.
d. At present all ITSPs are paying necessary license fees for providing broadband data and VOIP services to customers in its service area. Since Internet Telephony is a stream of this broadband data and already the ITSP license holder is paying license fees for both the services, there is no need to have any extra fees for Internet Telephony.
e. ITSP Service provider pays heavy cost for hiring internet data from TSPs. Since Internet Telephony (VOIP) is part of this stream of internet data for which ITSPs are already paying heavy cost to TSPs.

In view of the above, M/s Ortel is of the opinion that there should not be any additional entry fee, PBG or FPG for Internet Telephony service provided by ITSPs. In addition to this monthly rental for Internet Data to TSPs, firstly ITSPs are paying a handsome amount of money to ex-chequer for the ITSP license, secondly ITSPs are levied AGR on prorate basis yearly.

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?

Answer 2: The call routed for Internet Telephony is through public internet and not through the NLD/ ILD as required for circuit switched network. Therefore, in this case inter-service area call is travelling through public internet to reach node of IMS core or SIP server without NLDO. However, for connecting to the Intra Circle Switched network, ISPs need to be provided to interconnect to near by TSPs, who in turn would allow to interconnect to other TSPs with in the celling limit or as negotiated by the ISPs with the TSPs, below the ceiling limit.

Due to this appreciable differences between Internet Telephony and Circuit Switched telephony, Ortel would like to recommend not to have any interconnectivity charges for Internet Telephony.

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.

Answer 3 : This Internet Telephony service should be treated separately and not as an extension of fixed line or mobile service of TSP. This is because, Internet Telephony is part of the stream of Internet Data, for which ISPs are paying heavy monthly rental to TSP. While a call is initiated using Internet Telephony, the major part of the call is routed through the public Internet network and not in the circuit switched environment.

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.

Answer 4 : As discussed in Answer 3 for Question 3, Internet Telephony is altogether a separate service different from the normal fixed and mobile service. The charges levied for TSPs to interconnect with the other TSPs is different because they use maximum traffic in Circuit Switched Environment as a fresh network for the initiated call. Whereas in case of Internet Telephony, the call is generated as part of data stream in the public network for the maximum portion of the traffic and only in the last mile the traffic is handed over to the Circuit Switched Network through the gateway.

Keeping this difference in mind, Ortel would like to recommend that the cost to be levied for Internet Telephony would be preferably at zero cost ,reason being it would be a duplication of payment when the user is paying for bandwidth charges to avail the services of Internet telephony. .

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

Answer 5 : The Internet Telephony call is initiated either in the form of SIP Protocol with Best Effort (BE) traffic or in MGCP protocol with Under Guaranteed Service (UGS) Traffic. Hence, the termination charge, would be calculated based on the public internet used per call to terminate into Internet Telephony network. The calculation would be 30 channels per 2 Mbps link terminated into the network. The charges would be nominal.

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?

Answer 6 : The termination charges should be negligible as the Internet Telephony uses public Internet Data to transmit the voice. The necessary cost is being paid by the ISPs to the TSPs for the internet bandwidth. Thus in the opinion of the Ortel there should not be any termination fees for terminating at wireline or wireless network.

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

Answer 7: There is no demarcation between the International or National as the Internet Telephony Call travels using public internet network. This internet network is global in nature and boundary limitation is not there. The ISPs who hires Internet Bandwidth is for Global Service and the internet telephony uses part of this Internet Bandwidth to deliver voice. Thus either international, national or local termination is the same. The ISPs while hiring the voice minutes terminating through a SIP server, pays necessary rental to the service provider for all countries. It is therefore in the opinion of Ortel no charges to be levied for international termination.

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?

Answer 8: As mentioned at Answer 7 above, Internet Telephony by definition itself has no geographic restriction. The soft switch from where the call originated and customer is hooked is based on the public Internet. Since soft switch, has a public IP customer can use the soft switch, till such time customer is connected to public internet. Thus SDCA concept is not applicable for Internet Telephony and mobility is unlimited and full mobility is available.

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?

Answer 9: As mentioned at Answer 7 and Answer 8, yes the last mille for an Internet Telephony is the Public Internet and it has no boundary limitation. Till such time public internet is available and soft switch connected to public IP any call can be originated by the Internet Telephony subscriber from any here. Whereas, PSTN subscribers are bounded by the legacy of NLDO and ILDO with a SDCA concept.

Question 10: What should be the framework for allocation of numbering resource for Internet Telephony services?

Answer 10: Numbering Frame work for internet telephony to be followed as per E164, a document issued by ITU. The document is known as "A Framework for E.164 Number to IP Address Mapping". We would intend to follow international numbering system so that, a homogeneity is maintained at all level. Salient feature of numbering system is as under :-

a. Internet telephony service providers will obtain blocks of E.164numbers from numbering plan administrations for their subscribers.
b. For scalability purpose, subscriber-related data may be partitioned and distributed among multiple servers of the same type. These servers may be owned by individual service providers or by authorised industry third party service providers.
c. An Internet Telephony(IT) subscriber with a specific E.164 number will ordinarily subscribe to the service of one IT service provider. The service provider or a third party provider will maintain the data related to that subscriber, including the IP address that the subscriber can currently be reached.
d. If the subscriber uses the same E.164 number for the same service from more than one provider, other criteria need to be used for determining which service provider's subscribe database to be consulted for call delivery. It is emphasized that the service provider whose database was queried needs not carry the call.

e. Emergency calls to be provided as per ITU guide lines.
f. Details on numbering plan can be obtained from the E164 ITU document.

Question 11: Whether Number portability should be allowed for Internet Telephony numbers? If yes, what should be the framework?

Answer 11: Yes, it may be allowed. Frame work would be used as per E164 ITU document and its supplement document on number portability. The following general routing scheme is assumed as the routing model for calls routed to a ported customer regardless of the network (GSTN, NGN and IP) being used to provide the transport.
The first step/solution discussed for number portability is often that the donor network maintains the portability information, i.e., the complete address to the recipient network for ported-out numbers, and re-routes incoming calls to ported-out numbers onward towards the recipient network, according to onward routing.

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?

Answer 12: Yes it is possible to provide location information to the police station, when the subscriber is making an Internet Telephony Call. For wireline Internet usage it gives location of the device installed physically, whereas for wireless usage nearest AP is tracked and accordingly the customer device location is given. Presently in USA all cable operators like Comcast, Time Warners and others provide VOIP service with emergency dialing facility.

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 ?

Answer 13: Since it is possible to provide emergency number, there is no need to inform subscriber in advance.

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.

Answer 14: QoS (Quality of Service) is a major issue in VOIP implementations. The issue is how to guarantee that packet traffic for a voice or other media connection will not be delayed or dropped due interference from other lower priority traffic.

Things to consider are

- **Latency**: Delay for packet delivery
- **Jitter**: Variations in delay of packet delivery
- **Packet loss**: Too much traffic in the network causes the network to drop packets
- **Burstiness of Loss and Jitter**: Loss and Discards (due to jitter) tend to occur in bursts

VOIP QoS Requirements

Latency
Callers usually notice roundtrip voice delays of 250ms or more. ITU-T G.114 recommends a maximum of a 150 ms one-way latency. Since this includes the entire voice path, part of which may be on the public Internet, your own network should have transit latencies of considerably less than 150 ms.

Jitter
Jitter can be measured in several ways. There are jitter measurement calculations defined in:
IETF RFC 3550 RTP: A Transport Protocol for Real-Time Applications
As per CISCO Jitter buffers (used to compensate for varying delay) further add to the end-to-end delay, and are usually only effective on delay variations less than 100 ms. Jitter must therefore be minimized.

Packet Loss
VOIP is not tolerant of packet loss. Even 1% packet loss can "significantly degrade" a VOIP call using a G.711 codec and other more compressing codecs can tolerate even less packet loss. As per CISCO the default G.729 codec requires packet loss far less than 1 percent to avoid audible errors. Ideally, there should be no packet loss for VoIP

Generally for maintaining good quality voice, it is required to maintain UGS bandwidth of 64 Kbps.

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