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Voice over Internet Protocol (VoIP): The Dynamics of Technology and Regulation

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Abstract: *The appearance of VoIP comes at a moment when telecommunications system has already turned in a large-scale, complex system with multiple, challenging substructures. VoIP, however, greatly supplements the nested complication by affording a technology that enables multiple architectures and business models for delivering the same voice (and often converged voice and data) service, while remaining nonbeliever to the underlying infrastructure. The VoIP-enabled architectures have very different capabilities and costs from one another. Most develop the economic arbitrage opportunities by escaping access charges and universal service contributions. Added to this is the combination of reduced asset specificity due to VoIP's layered architecture and a global standard based ubiquitous IP technology that frees the service providers of the need to own the delivery infrastructure, and enables them to offer service from anywhere globally.*

Keywords*: voIP, universal service, global, ubiquitous, IP technology*

I. INTRODUCTION

This paper is an attempt to provide a framework forunderstanding how voice over Internet protocol (VoIP) technology will impact regulatorychoices, without speculating on the nature of the new regulatory regime.On the technical side, Internet Protocol (IP) being agnostic to the physicalmedium provides a way to run VoIP as an application on wired or wireless networks. Thewired network could be a public switched telephone network (PSTN), cable, digitalsubscriber line (DSL) or the Ethernet. The wireless network could be the wirelesscarrier's network, such as code division multiple access (CDMA), time division multipleaccess (TDMA) or GSM network, or private networks such as Wi-Fi, Bluetooth orWiMaX. There are multiple, different architectures under which a service provider canoffer a VoIP based voice communications service. At one extreme, it is possible to offerVoIP as an application that utilizes any infrastructure that offers the Internet connectivity. The application provider in this case need not own any parts of the infrastructure. On theother, there can be a complete vertical integration of service where the provider owns theinfrastructure and all the components necessary to deliver service. Therefore, the choiceof architecture determines the service provider's underlying costs, capabilities andlimitations. This necessitates the study of infrastructure ownership when discussingoptions forregulating various scenarios under which VoIP services is delivered tocustomers.On the regulatory side, voice communications service has been subjected to a100-year-old regulatory regime. The Internet on the other hand has been exempt fromregulation. As the VoIP bridges the two worlds of PSTN and the Internet, the question forthe regulators is: should VoIP service be regulated as a common-carrier regulation, justlike a PSTN telecommunication service provider, left unregulated like the Internet. In this paper, we will first discuss a way to classify the current panoply of VoIPofferings and the challenges they pose if the current regulatory regime were to apply tothem. We will then examine the case of Communications Assistance for the LawEnforcement Act (CALEA) also known as the wiretapping act to study itsImplications on VoIP. A system dynamics model is used for the analysis.

II. TECHNOLOGY AND REGULATION

Voice communication carried out using the Internet Protocol (IP) for the transportis known as Voice over Internet Protocol (VoIP). Traditional phone networks, known asPublic Switched Telephone Networks (PSTN) used circuit-switching. In Circuit-Switching, resources are reserved along the entire communication channel for the duration of the call. Conversely, Internet Protocol (IP) uses packet-switching. In Packet-Switching, information is digitally transmitted into one or more packets. Packets knowtheir destination, and may arrive there via different paths.Implementing VoIP requires a range of protocols from those needed to do call Signalling for call establishment and more, to transport real-time voice across the network,to do quality-of-service-aware routing, resource reservation, Qos-aware of the network management and billing. We will examine evolution of each of these protocols to understand how they fit the currently popular architectures.

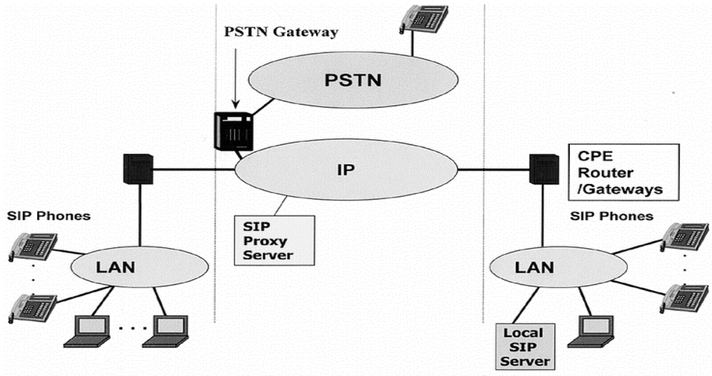

Figure 1. End-to-end VoIP

The purest VoIP implementation uses IP capable end-user equipment such as IP phonesor a computer and does not rely on a standard telephone switch. Figure 1 is a simplified diagram of an IP telephone system connected to a wide area IP network. IP phones areconnected to a LAN. Voice calls can be made locally over the LAN. The IP phones include codes that digitize and encode (as well as decode) the speech. The IP phones also packetize and depacketize the encoded speech into IP packets. Calls between different sites can be made over the wide area IP network. Proxy servers perform IPphone registration and coordinate call signalling, especially between sites. Connections tothe PSTN can be made through VoIP gateways.As voice communication has been around for about 100 years, there exists a verywell developed industry around the circuit-switched PSTN. There are many establishedincumbents with large customer bases. In the early days of VoIP, PSTN incumbentsconsidered it a threat to their business, and an opportunity to the data networking vendorssuch as the Internet Service Providers (ISP). Over time, the PSTN incumbents and the new entrants to voice communications alike view VoIP as an opportunity to providevoice service at a significantly reduced cost.

III. CALL SIGNALLING

VoIP requires a means for potential communications partners to find each other and tosignal to the other party their desire to communicate. This functionality is referred to asCall Signalling. The need for signalling functionality distinguishes Internettelephony from other Internet multimedia services such as broadcast and media-on demandservices.VoIP, when used for synchronous voice or multimedia communication betweentwo or more parties, uses signalling that creates and manages calls. The called can definea call as a named association between applications that is explicitly set up and torn down.Examples of calls are two-party phone calls, a multimedia conference or a multi-playergame. A call may encompass a number of connections, where a connection is a logicalrelationship between a pair of end systems in a call. For example, a non-bridged threeparty audio only call will have three connections, creating a full mesh among theparticipants. A media stream or sessionis the flow of a single type of media among a setof users. This flow can either be unicast (in which case it is between two users), ormulticast (more than two users). A media session is associated with one or more ofconnections. In the above three party call example, if the media is distributed usingunicast, there will be one audio session per connection. If the audio is distributed via multicast, there will be one audio session associated with all three connections. It is not required that calls have media streams associated with them, but this is likely to be thecommon case.Internet telephony signalling may encompass a number of functions: name translation and user location involves the mapping between names of different levels of abstraction, feature negotiation allows a group of end systems to agree on what media toexchange and their respective parameters such as encoding, call participant management for participants to invite others on an existing call or terminate connections with them, feature changes that make it possible to adjust the composition of media sessions during the course of a call, either because the participants require additional or reduced Functionality or because of constraints imposed or removed by the addition or removal of call participants. Media Gateway Control Protocol (MGCP), and Mega co/H.248. H.323 and SIP are peer-to-peercontrol-signalling protocols, while MGCP and Mega co are master–slave control-signalling protocols. MGCP is based on the PSTN model of telephony. H.323 and Mega co are designed to accommodate video conferencing as well as basic telephony, but they are stillbased on a connection-oriented paradigm similar to circuit-switching, despite their use for packet communications systems. H.323 gateways have more call control function thanthe media gateways using MGCP, which assumes that more of the intelligence resides in a separate media gateway controller. SIP was designed from scratch for IP networks, and accommodates intelligent terminals engaged in not only voice sessions, but other applications as well.

IV. H.323

The ITU-T recommended H.323 protocol suite has evolved out of a video telephony Standard. When early IP telephony pioneers developed proprietary products2, there was an industry call to develop a VoIP call control standard quickly so that users and service providers would be able to have a choice of vendors and products that would interoperate. The Voice-over-IP Activity Group of the International Multimedia Telecommunications of Consortium (IMTC) optional H.323, which had been developed for multimedia communications over packet data networks. These packet networks might contain LANs or WANs. The IMTC held the view that VoIP was aspecial case of IP Video Telephony. Although not all VoIP pioneers agreed that video telephony would quickly become popular, the H.323 protocol suite became the early leading standard for VoIP implementations.H.323 entities may be included into personal computers or routers are implemented in stand-alone devices. Inaddition, the VoIP gateway may perform speech transcending and compression, and it isusually capable of generating and detecting dual tone multiple frequencies (DTMF) signals.

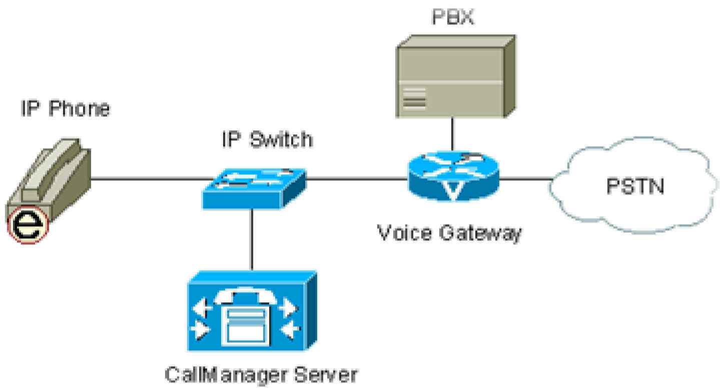

Figure 2 H.323 Gateway

developed for multimedia communications over packet data networks. A System Control Unit provides signalling for proper operation of the H.323terminal that provides for call control using H.225.0 and H.245 .H.225.0 layer formats the transmitted audio and control streams into messages,retrieves the audio streams from messages that have been received from thenetwork interface, and performs logical framing, sequence numbering, errordetection and error correction as appropriate. Registration messages and other means. Admissions control—the gatekeeper authorizes network access using H.225messages. Admissions criteria may include call authorization, bandwidth, orother policies. The Bandwidth control of the gatekeeper controls how much bandwidth a terminalmay use Zone management aterminal may register with only one gatekeeper at a time. The porter provides the above functions for terminals and gateways that have registered with it. Participation in call control signalling is optional. Directory services are optional.When an endpoint (such as a phone) is connected to the network, the Registration,Admissions and Status (RAS) channelcarries messages used in gatekeeper endpoint registration processes that associate an endpoint's aliaswith its TCP/IP address and port number to be used for call signalling. If the network has a gatekeeper, the calling endpoint sends the initial admission message to the gatekeeper using the gatekeeper's RAS Channel Transport Address. In the initial Exchange of admissions messages, the gatekeeper tells the originating endpoint weather.

V. VOICE QUALITY

There are various approaches to providing QoS in IP networks. However, the firstquestion is whether QoS is really necessary. Some Internet engineers argue that if the occupancy is low, then performance should be good. Essentially, the debate is over whether excess network capacity (including link bandwidth and routers) is less expensive than QoS implementation.QoS can be achieved by managing router queues and by routing traffic around congested parts of the network. A variety of resource management techniques may be used to achieve this, but the end result will be that some packets willreceive different service than others. This will, for example, allow service providers to offer a real-time service giving priority to the use of bandwidth and router queues, up to the configured amount of capacity allocated to real-time traffic. The appeal of DiffServ is that it is relatively simple, yet provides applications like VoIP some improvement in performance compared to "best-effort IP networks.One more come near to achieving voice quality is to use MPLS. MPLS offers IP networks the capability to provide traffic engineering as well as a differentiated services approach to voice quality.

VI. CONCLUSION

In a VoIP network, voice quality is only as good as the quality of the weakest network connection. Packet loss, delay and delay variation all contribute to degraded voice quality. In addition, because network Congestion (or more accurately, instantaneous buffer congestion) can occur at any time in any portion of the network, network excellence is an end-to-end intend concern. The QoS apparatus discussed in this project a set of mechanisms to increase voice quality of data networks by decreasing dropped voice packets during times of network congestion and minimizing both the fixed and variable delays encountered in a given voice connection. Another key driver will be the development of wireless VoIP, which will provide students and personnel with different ways to access learning environments while integrated services will open up the possibilities of everywhere computing. In future, learning may take place in a shared virtual world of common computing, alike to an online, multiplayer pastime, and VoIP will be one of the components of such an environment.

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