BEFORE THE FLORIDA PUBLIC SERVICE COMMISSION

In re: Investigation into appropriate methods to compensate carriers for exchange of traffic subject to Section 251 of the Telecommunications Act of 1996.	))))	Docket No. 000075-TP

DIRECT TESTIMONY OF

ELIZABETH A. GEDDES

ON BEHALF OF

VERIZON FLORIDA INC.

MARCH 12, 2001

DIRECT TESTIMONY

OF

ELIZABETH A. GEDDES

Q. PLEASE STATE YOUR NAME, BUSINESS ADDRESS AND OCCUPATION.

A. My name is Elizabeth A. Geddes. My business address is 2107 Wilson Boulevard, Floor 11, Arlington, Virginia 22201. I am employed by Verizon Network Services Group as a member of the Technical Staff.

Q. PLEASE DESCRIBE YOUR EDUCATIONAL BACKGROUND AND PROFESSIONAL EXPERIENCE.

A. I received a Bachelors of Science in Mechanical Engineering from University of Notre Dame and a Masters of Science in Applied Biomedical Engineering from Johns Hopkins University. I have three years of experience in the telecommunications industry.

Q. WHAT IS THE PURPOSE OF YOUR TESTIMONY IN THIS DOCKET?

A. The purpose of my testimony is to address issue 16(a): What is the definition of Internet Protocol (IP) telephony?

My testimony will focus exclusively on the technical aspects of IP telephony. Issue 16b, concerning compensation for IP telephony, will be addressed by Verizon witness Dr. Edward Beauvais.

In order to understand IP Telephony, it is helpful to first define the terms "Internet" and the underlying suite of protocols upon which the Internet relies .

Q. WHAT IS THE "INTERNET"?

A. The term "internet" refers to any collection of connected networks. The "Internet" (with a capital I) is a worldwide collection of interconnected computer networks that got started in the late 1960s when the U.S. Department of Defense's (DoD's) Advanced Research Projects Agency (ARPA) funded a research project that led to the development of ARPANET, an experimental network that demonstrated the feasibility of connecting computers via a packet-switched network. ARPANET has since evolved into the Internet, which connects thousands of networks worldwide. Today, a variety of applications such as email, file transfers, "surfing" the World Wide Web (WWW), and some forms of Internet Protocol (IP) telephony are concurrently run over the Internet.

Q. WHAT IS "INTERNET PROTOCOL"?

A. "Internet Protocol" is a standard protocol that provides a connectionless, unconfirmed transmission and delivery service.

The International Organization for Standardization (ISO), a worldwide federation of national standards bodies from some 110 countries, developed a model that permits unique systems to communicate regardless of their underlying architecture. The components that comprise this model, which I will describe in more detail, are commonly referred to as a protocol. This model is known as the Open Systems Interconnect (OSI) model, which consists of seven distinct layers. Each layer performs a distinct function that is transparent to each of the other layers, and, each layer can only communicate with the layers immediately above and below it.

The Internet relies on the Transmission Control Protocol/Internet Protocol (TCP/IP) suite of protocols, which, although not part of the OSI model, roughly corresponds to the layers in the OSI model. The OSI model consists of seven layers as follows (beginning with layer one): the physical layer, the data link layer, the network layer, the transport layer, the session layer, the presentation layer and the application layer. (Generally, layers 5 and 6, the session and presentation layer respectively, are not employed by the TCP/IP suite of protocols.) A packet is really just the data associated with the application layer wrapped inside a transport protocol packet that, in turn, is wrapped in a network protocol packet, and so forth.

Although the Internet consists of networks that rely on different lower layer technology (i.e., layers 1 and 2), each of these networks primarily relies on the TCP/IP suite of protocols for their higher layers (i.e., layers 3 – 7). The Internet Protocol (IP), which roughly corresponds to layer 3 of the OSI model, the network layer, is designed for routing a packet to its destination. IP is a protocol that provides a connectionless, unconfirmed delivery service. Connectionless means that no handshaking occurs between IP nodes prior to sending data. Unconfirmed means that IP sends a packet without sequencing and without an acknowledgment that the destination was reached. Instead, IP makes a best effort to deliver packets to its final destination. The IP header contains information necessary for routing the packet, including source and destination IP addresses. Because each router decides independently where to forward a packet, a packet's path between two sites is not necessarily the same as the next packet's path. Additionally, because of various transit delays, each packet can arrive in a different order from which it was sent. Higher layer protocols may be employed for reliable transport of IP packets. For example, the Transmission Control Protocol (TCP), which roughly corresponds to layer 4 of the OSI model, the transport layer, is designed for reliable transmission of a packet. Alternatively, another transport layer protocol, User Data Protocol (UDP) is designed for "best effort," unconfirmed transport of IP packets. While IP combined with TCP is an ideal protocol suite for the transmission of data packets for email and "surfing" the Internet, most IP Telephony applications rely on IP combined with UDP, for optimal transport of real-time voice packets.

Q. PLEASE DESCRIBE THE BASIC UNDERLYING TECHNOLOGY EMPLOYED IN IP TELEPHONY.

A. IP Telephony encompasses a very diverse array of applications ranging from the somewhat crude conversation conducted between two users via their personal computers to the more innovative "click to talk" application in which a user, by selecting a hyperlink on a web page, is instantly connected to a live representative in a call center. While there may not be a single definition for IP Telephony, IP Telephony generally refers to voice or facsimile telephony services that are at least partially transported over an IP network in lieu of the traditional circuit-switched network. (While, today, the Public Switched Telephone Network (PSTN) primarily relies on a circuit-switched network, in the future, the PSTN may employ a packet-switched network in place of portions of the existing circuit-switched network. It is therefore somewhat misleading to simply contrast IP Telephony with the PSTN.) The basic steps involved in an IP telephony call are the conversion of the analog signal to a digital signal and the subsequent translation of that signal to packets of data for transmission over a packetized network. The reverse process occurs at the packets' receiving end, where the many packets are reassembled in the proper sequence, and then converted back to analog. Thus, IP telephony is typically achieved in combination with the PSTN.

Q. PLEASE DESCRIBE THE TECHNICAL CHARACTERISTICS OF IP TELEPHONY.

A. Transporting voice over an IP network, rather than over the traditional circuit switched network, increases bandwidth utilization efficiency of the network in three ways. First, it allows the consolidation of voice and data onto one single network rather than having to maintain two separate costly networks. Secondly, it only occupies bandwidth when there is data (i.e., voice packets) to transmit. In a circuit-switched network, when a user makes a telephone call, a dedicated path is allotted to those end users. In an IP network, voice packets are transmitted over a shared network in a "best effort" manner. During periods of silence in a telephone conversation, a circuit-switched network continues to reserve that bandwidth because it has been dedicated to those users even though the conversation is idle. In a packet-switched network, bandwidth is not occupied during those times of silence, leading to increased efficiency throughout the network. Thirdly, by employing complex compression algorithms in the analog to digital conversion, the voice channel may occupy significantly less bandwidth than occupied on a standard Time Division Multiplexed (TDM) telephony channel, used in circuit-switched networks. However, degraded quality of service, as compared to circuit-switched networks, is a consequence of this increased efficiency.

As I mentioned above, IP telephony is an unconfirmed delivery service. An efficiency/service quality trade-off arises because each router independently determines a packet's path and different packets may arrive at a destination at different times and out of sequence. Some packets may never even reach their destination. These factors lead to increased latency, jitter and packet loss, all of which contribute to the degradation in the quality of service. Jitter is the random variation in the time it takes a packet to reach its destination. Latency is the time it takes for a packet to cross a network connection, from sender to receiver. While latency is not generally an issue for non-real time services (e.g., "surfing" the Internet), in real-time, two-way communications such as telephony, latency over a certain threshold may lead to intolerable service quality. Similarly, if too many packets are lost, then this may lead to intolerable service quality (i.e., at the receiving end of the conversation, the sound may appear broken up).

Q. IS THERE A DIFFERENCE BETWEEN IP TELEPHONY AND PACKET-BASED TELEPHONY?

A. Yes. It is important to make a distinction between packet-based telephony and IP Telephony. Packet-based telephony is a more general term for IP Telephony, indicating that the underlying network is based on IP rather than some other type of network (*e.g.*, ATM or Frame Relay). (To make matters even more complicated, IP packets may be carried as payload inside ATM cells or Frame Relay frames.) Many types of packetized telephony fall under the purview of packet-based telephony, including IP Telephony, Voice over Asynchronous

Transfer Mode (VoATM), and Voice over Frame Relay (VoFR).

Q. IS IP TELEPHONY CARRIED OVER THE SAME INTERNET USED FOR "SURFING" THE WWW AND EMAIL?

A. A common misconception is that IP Telephony only refers to telephony carried over the Internet (with a capital I), which is the network used to "surf" the WWW and to send and receive email. In reality, the underlying IP network used in IP telephony just as easily may be a private internet as the Internet. In fact, in many cases, a private internet is used in IP telephony in order to increase the quality of service. There is a term, Internet Telephony, that encompasses only telephony sent over the Internet. Internet Telephony is a subset of IP Telephony. However, for simplicity, for the remainder of these comments, I will use the term Internet to include both the Internet and private internets.

Q. PLEASE DESCRIBE THE DIFFERENT CONFIGURATIONS OF IP TELEPHONY.

A. There are many different possible configurations of IP Telephony. IP Telephony may be offered between two Personal Computers (PCs), between two telephones or between a telephone and a PC. Following is a brief overview of these three different configurations of IP telephony.

Q. WHAT IS PC-TO-PC IP TELEPHONY?

A. Originally, IP Telephony was a telephony application between two Personal Computers (PC). For PC-to-PC IP telephony, each PC requires an active connection to the Internet, a sound card, a microphone, and speakers. Additionally, for the most part, both PCs need to be running the same application software. (For example, a user running DialPad software could not successfully make a call to another user with a PC running Net2Phone software since the two pieces of software are not interoperable.) Typically, the caller "dials" a person by selecting someone from a list of users currently on-line who are able to receive calls. Since the PSTN is not used to switch the call, user names rather than the traditional 7- or 10-digit North American Numbering Plan (NANP) telephone numbers are used to identify the desired terminating party. In fact, the only PSTN resources used in this service are the facilities used to connect to the Internet via an Internet Service Provider (ISP).

Communication between users is limited to the set of users who have an active connection set-up to the Internet, and further limited to the subset of users equipped with identical application software running on their PCs. Because of these limitations, PC-to-PC IP telephony, although a rudimentary form of telephony, probably cannot serve as a substitute for the PSTN.

Q. WHAT IS PC-TO-PHONE IP TELEPHONY?

A. PC-to-Phone IP telephony employs a single gateway. With the introduction of gateways, IP Telephony could be offered as a telephony service between a PC and a conventional telephone, significantly expanding the range of the service. (A gateway is software or hardware that permits communications between two different networks based on different protocols. For example, an IP telephony gateway translates IP packets to Pulse Code Modulated (PCM) traffic suitable for travel over the PSTN and vice versa.) In PC-to-Phone IP Telephony, beyond the gateway, the PSTN will be used to switch the call to the termination telephone. Therefore, users now must "dial" a terminating party by inputting a 7- or 10-digit NANP telephone number. Additionally, the PC-to-Phone configuration requires only one party, the calling party, to have a PC and an active Internet connection.

Q. PLEASE PROVIDE EXAMPLES OF APPLICATIONS OF IP TELEPHONY THAT RELY ON A PC-TO-PHONE CONFIGURATION.

A. An application of the PC-to-Phone configuration, which is gaining popularity in the e-commerce world, is "Click to Talk." In this application, by simply clicking on a designated web page hyperlink, a user may be instantly connected to a live representative in a call center to answer questions or provide additional information. In this scenario, the user "dials" by the click of a button. For dial-up users with one telephone line for voice and data, this permits users to have their questions answered while on-line, rather than having to disconnect to make the phone call.

Another application of this configuration, with a twist, is popular with customers who want to consolidate their voice and data traffic onto a single network. For example, large business customers whose voice network employs either a PBX switch on their premise or Centrex service, which is provided by their telephone carrier, may consolidate their voice network onto their existing Local Area Network (LAN). In an IP PBX configuration, a gateway compatible with their existing PBX may be deployed to translate the packetized voice traffic to traffic suitable to travel over the PSTN. In a Centrex configuration, a telephone carrier may provision an IP Centrex service in which the gateway is deployed next to the Centrex switch in the carrier's central office. In either IP PBX or an IP Centrex configuration, an IP phone may be used in lieu of a PC in a configuration similar to the PC-to-Phone configuration described above. An IP phone, used on an Ethernet LAN connection, may be designed to look and work just like a conventional Plain Old Telephone Service (POTS) phone, but it plugs into an Ethernet RJ-45 wall jack instead of the traditional RJ-11 analog telephone jack. In this scenario, the functionality of a PC used for IP Telephony is placed in an IP phone. That is, the digitization of an analog voice signal and subsequent packetization actually occurs in an IP phone rather than in a PC. Users may directly dial both users served by the PSTN and users served by other IP phones.

Q. WHAT IS PHONE-TO-PC IP TELEPHONY?

A. Phone-to-PC IP telephony also employs one gateway. To initiate a call, typically, the originating party first has to dial an access telephone number to access a gateway. Once a connection is established with the gateway, the party dials the terminating party's telephone number, again using 7- or 10- digit NANP telephone numbers from a conventional POTS telephone. The telephone number is a unique telephone number that has been assigned to a user who has registered for this particular service. The PSTN routes the call to a gateway that connects the PSTN to the Internet. In Phone-to-PC IP Telephony, beyond the gateway, the Internet will be used to route the call to the terminating party. The Phone-to-PC configuration requires the called party, rather than the calling party, (as in the PC-to-Phone configuration) to have a PC and an active Internet connection.

Q. WHAT IS PHONE-TO-PHONE IP TELEPHONY?

A. Phone-to-Phone IP telephony employs two gateways instead of just the one gateway that is used in PC-to-Phone IP telephony. With the employment of two gateways, the scope of IP Telephony was further expanded to permit IP Telephony service between two conventional telephones. In this configuration, neither party is required to use a PC or to be connected to the Internet. To initiate a call, the originating party may first have to dial an access telephone number to access a gateway. (If the party directly dials the terminating party's telephone number, the call will be routed over the default route, which is usually the PSTN.) Once a connection is established with the gateway, the party dials the terminating party's telephone number, again using 7- or 10- digit NANP telephone numbers. (In some configurations, the default route for a telephone service provider may be a packetized network through the use of gateways. In that case, there is no need to first dial an access number.) A second gateway is employed near the called party. Essentially, in this configuration, IP telephony service may appear to the user as no different from traditional circuit-switched telephony service.

Q. IS PACKET-BASED TELEPHONY A HIGHLY EVOLVED TECHNOLOGY?

A. No. Packet-based telephony, of which IP Telephony is a subset, is still a rather nascent technology, which, as I have explained, can take many forms. The more widespread deployment and use of broadband access and next generation networks (converging voice, video and data) can be expected to further drive the development of packet-based telephony. As Verizon witness Beauvais notes in his testimony, it is important for policymakers to avoid precipitous action in this area, which might hinder further innovation.

Q. PLEASE SUMMARIZE YOUR TESTIMONY.

A. The term IP Telephony encompasses a broad variety of services. IP

Telephony may be offered in various configurations (i.e., between two PCs, between a phone and a PC or between two phones). IP Telephony may be offered over a combination of different types of underlying backbone networks (e.g., the public Internet or a private managed internet). IP Telephony may also be offered over different types of access networks (e.g., corporate intranet, broadband connection or PSTN). In addition, there are other types of packet-based telephony beyond IP Telephony, and packet telephony may be offered using different underlying protocols (e.g., ATM, Frame Relay, and IP).

In its deliberations in this docket, the Commission should remain aware that packet-based telephony is still a relatively new technology and, as Dr. Beauvais notes, policy needs to be set accordingly.

Q. DOES THIS COMPLETE YOUR TESTIMONY?

A. Yes.