NORTHCENTRAL UNIVERSITY

A COMPARISON OF VoIP AND ANALOG PERCEIVED CALL QUALITIES

A Dissertation submitted to the Graduate Faculty

Of the Department of Business and Management

In candidacy for the degree of

DOCTOR OF PHILOSOPHY

By

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DISSERTATION ABSTRACT

Name: Kirby J. Scheer Degree: Doctor of Philosophy

Institution: Northcentral University Major Field: Business Administration

Location: Prescott, Arizona Specialization: Management Information Systems

Title of Dissertation: A COMPARISON OF VoIP AND ANALOG PERCEIVED CALL QUALITIES.

Scope of Study: This dissertation compared perceived call quality differences between voice-over Internet protocol (VoIP) and analog landline-based communications through a field experiment. This field experiment was set up so that consumers and businesses can duplicate it to compare call qualities and use it as a tool to make educated decisions in communications.

Findings and Conclusions: This dissertation provides data comparing VoIP and analog calls. Extensive research has been done on VoIP delay and jitter controls. A conclusion was made that the majority of the participants in this study believed that the VoIP calls were slightly better than the analog calls. The findings in this experiment were derived from a group of 112 participants in a blind study.

Chair's Approval: Keith Harman Date: 2/28/05

Dr. Keith Harman

NORTHCENTRAL UNIVERSITY

We recommend that the dissertation prepared under our supervision by

KIRBY J. SCHEER

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A Comparison of VoIP and Analog Perceived Call Qualities

be accepted in partial fulfillment of the requirements for the degree of

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Chapter 1: Introduction

This dissertation focused on the Voice over Internet Protocol (VoIP®) using a digital subscriber line (DSL)-enabled technology and compared the Relative Perceived Call Quality of VoIP to traditional landline-based communications. For the purposes of this research, the construct "Call Quality" was defined in terms of overall clarity, levels of voice, outside interference, background noise, and the extent to which full duplex conversation was feasible for both parties to a telephone conversation. "Relative Perceived Call Quality" was defined as the user's perception of how the call quality of a DSL-enabled VoIP product compares to that of landline-based phones. "Initial Beliefs" with respect to Call Quality are defined as the user's beliefs regarding the perceived sound quality of VoIP before it is used. This research also analyzed transmission speeds through DSL-enabled broadband connections and their relationships to the Relative Perceived Call Quality of VoIP. Transmission speed is the speed of the broadband connection a participant is using, measured in kilobits per second (Kbps). Call Participants are Call Initiators who initiate a VoIP or analog call, and Call Receivers who receive a VoIP or analog call.

VoIP is a new dynamic in transmitting voice calls over the Internet. "VoIP gained momentum in 2003, says Kate Griffin, an analyst at The Yankee Group, a telecom consulting firm" (Jain, 2004, p. 1). VoIP services function much like those of a traditional analog phone line, but are more efficient. (In this context, efficiency is the extent to which long distance charges are reduced or eliminated for voice calls transmitted over the Internet.) VoIP does not use standard phone lines. Rather, calls made through VoIP utilize the Internet to transmit the voice communication. Since the voice is transmitted over the Internet, broadband Internet connections are needed for the communication to take place in real time. Transmission speeds must be at least 90 Kbps to effectively facilitate uncompressed phone calls through VoIP (Vonage, 2004). Uncompressed calls are defined as calls that use the maximum amount of bandwidth that is necessary to facilitate the VoIP call.

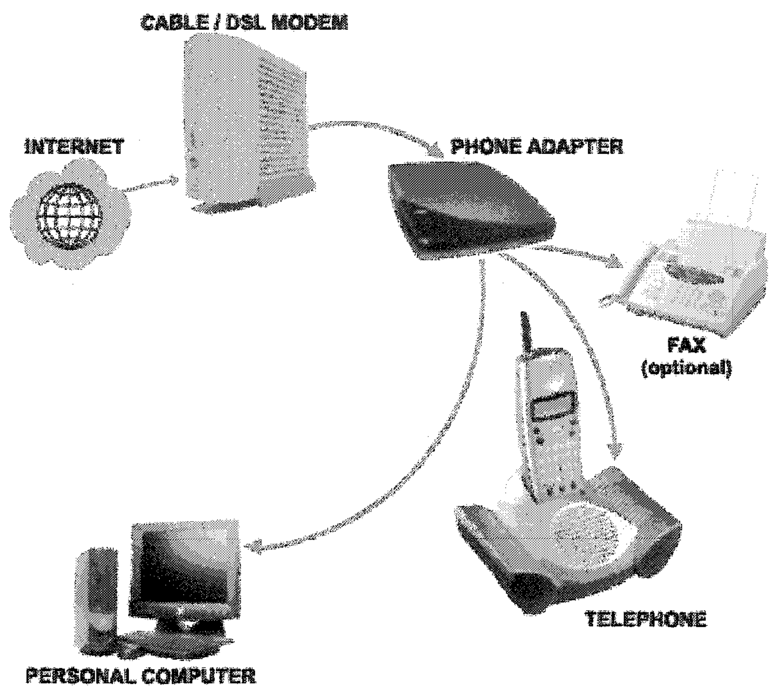

Figure 1: How VoIP Calls are Made (Vonage, 2004)

Figure 1 illustrates how VoIP calls are placed and received through the Internet public telephone network (PTN). Lenny Liebmann, an independent consulting specialist, cautions “VoIP can be a reliable technology with some concrete benefits, but you have to make sure your network is ready to handle it” (Liebmann, 2003, p. 46). VoIP calls can be placed through a regular analog phone with an adapter, which connects the phone to the broadband connection. From this point, the voice is divided into packets of information and transmitted through the broadband connection via the Internet. While the packets of information are being transmitted, the packets are reassembled through the routing system of the Internet or switches in the PTN to recreate audible voice in an analog format at the recipient's end. No additional equipment is necessary for the recipient of a VoIP network call. VoIP customers may call anywhere a landline-based customer may call. This means they can call other landline-based phones, other VoIP customers, international destinations, mobile users, and cellular users. The only caveat to this is that the VoIP carrier may restrict calls because of system capabilities. However, larger VoIP companies often open their systems so that calls can be placed anywhere. (Two of the major VoIP companies in the United States are Qwest Communications and Vonage).

A VoIP user dials a local or long distance call utilizing 11-digit dialing (Vonage, 2004). Since VoIP calls can be made from any place that has a broadband connection, users are not restricted to their homes or offices to make these calls. A VoIP customer who lives in Minnesota can take the VoIP adapter to a home in Florida, plug the adapter into a broadband Internet connection, and make calls as if the phone were in Minnesota. The local Minnesota number will even appear on the recipient's caller ID screen. Likewise, a call to the local Minnesota number would be routed to Florida, but would register as a local call. During traveling, the VoIP call will work the same as at home, provided there is a broadband connection that the VoIP adapter can utilize (Federal Communications Commission, 2004).

Statement of the Problem

In the telecommunications industry, new technologies have been built on the backbone of antiquated technologies. The backbone was defined as the existing telecommunications architecture over which traditional analog voice calls are transmitted. This reliance of new technologies on outdated ones has created an operational inefficiency in the telecommunications industry. The use of suboptimal various network paths for non-local voice calls are much too great, resulting in unnecessarily high long distance charges. Currently, VoIP services are not required to charge the regulatory charges that the Local Exchange Carriers (LECs) are required to charge. As a result, VoIP services normally are offered at a greater value than plain old telephone service (POTS) lines.

Many of the operational inefficiencies were covered from a cost standpoint before deregulation stimulated more competition in the industry. Costs were covered by the customers, who paid higher rates for telecommunication services. In 1996 the telecommunications industry was deregulated, making competition a reality. The resultant was decreased margins that exposed the operational efficiencies. With competition increasing between Competitive Local Exchange Carriers (CLECs) and Incumbent Local Exchange Carriers (ILECs), revenues have decreased, and profit margins have become smaller (Passmore, 2002). Because of these decreased profit margins, both CLECs and ILECs must increase their operational efficiencies to be able to transport the traditional voice calls in a more cost-effective way (Passmore, 2003).

The Internet is generally regarded as an efficient operational medium through which data is transmitted (Informationweek, 2004). The operational efficiency of the Internet permits VoIP calls to utilize the same transport mechanism as data. David Passmore, director of research at the Burton Group, writes, “Soon IP carriers will be able to cut the public switched telephone networks (PSTN) out of the loop” (Passmore, 2003, p. 15). However, calls between VoIP and analog phones can be problematic, since the call changes from analog to digital to analog in the conversion process (Borthick, 2002). VoIP can change the dynamics of telecommunications and address all of the operational efficiencies of the Internet. Local Exchange Carrier (LEC) revenues have continued to decrease. Even so, voice transmission over the Internet can make calls more operationally efficient and cost-effective with VoIP products. "The Internet is being modified to support voice traffic and products are being made to link the data and voice network" (Kulathumani, 1999, p. 2). LECs can achieve a new level of operational efficiency resulting in decreased transport costs for the carrier (Passmore, 2003).

This dissertation studied the Relative Perceived Call Quality of DSL-enabled VoIP with respect to landline-based analog calls through a pretest and posttest survey of 112 randomly selected individuals. The surveys measured the Relative Perceived Call Quality with respect to several factors as indicated in Table 1. Overall, there were four surveys. Two of the surveys were designed to measure the Relative Perceived Call Quality of landline-based service and VoIP service, respectively. The order in which this pair of surveys were assigned in the pretest/posttest experiments was randomized. The Pretest Survey was designed to measure the Call Initiators' and Call Receivers' Initial Beliefs–before they had used the VoIP product–of how VoIP calling might compare with their current analog phone service. The posttest survey was a Comparative Survey, in which, Call Initiators and Receivers indicated their perceptions of how VoIP-initiated calls compared with analog-initiated calls. Since the speed of the broadband connection may be a relevant factor affecting Relative Perceived Call Quality, a speed test was conducted on the broadband connection before the VoIP call was initiated. A DSL connection was used, thus providing a consistent broadband connection not subject to local area network fluctuations, through which the upload and download speeds were measured within the allowed ranges. (The only variance on a DSL line is that of random fluctuations caused by the DSL circuit). Therefore, because of the possibility of variance in connection speed, the speed test was used to determine which one of two levels was associated with the Call Initiators' connections for both upload and download speeds. The first level represented the range from 128-220 Kbps. The second level represented the range from 221-512 Kbps. The speed test tool used is available online at http://64.122.32.21 and is called the Integra Telecom speed test (Integra Telecom, 2004).

After all the surveys were conducted, a multiple forward stepwise regression analysis was used to analyze the following factors and how they may have affected the Relative Perceived Call Quality of DSL-enabled VoIP with respect to landline-based analog calls:

1. The speed of the DSL-enabled connection.
2. The Call Initiators' and Call Receivers' Initial Beliefs regarding the performance of VoIP Call quality.
3. The perceived Call Quality of analog landline-based equipment as described in the analog survey, for both Call Initiators and Call Receivers.
4. The final perceived Call Quality of VoIP as described in the VoIP survey, for both Call Initiators and Call Receivers.
5. The Call Initiators' and Call Receivers' posttest responses indicating their relative perceptions of how VoIP Call Quality compares to that of analog.
6. The Call Initiators' and Call Receivers' posttest responses indicating their relative perceptions of how analog Call Quality compares to that of VoIP.

Chapter 2: Review of Related Literature

A search of the entire ProQuest® database, along with other exhaustive searches, yielded a minimal number of primary or scholarly publications that examined VoIP quality of service (QoS) from a subjective standpoint. Sources devoted to objective testing of VoIP QoS were widely available and was the primary focus of the literature review. Much of the literature written focused on network constraints, admission control, traffic routing, delay, and jitter. The review showed that these, among other factors, can significantly impact QoS and perceived call qualities in VoIP networks.

Standards

Travers and Swale (2001) focused on why global VoIP standards are needed. They explained that standards are needed to complete calls. Without global standards, one country may have different standards than another country. If standards are not set, a call may never be completed, a packet may be lost, or delay and jitter may increase. This, among other factors, Travers and Swale argued, can have an effect on VoIP QoS.

When the International Telecommunications Union (ITU) sets standards, its primary focus is on interface requirements. Horizontal interfaces are a top priority. These are categorized in two areas: 1) inter-domain interworking, and 2) inter-technology interworking. Inter-domain interworking sets standards between two different domains. Inter-domain standards are necessary for domains to be able to understand each other. The same standard is set for the origination and termination of a VoIP call. Inter-technology interworking standards focus on compatibility between different hardware manufactures that comply with the ITU standards (Travers & Swale, 2001).

The second interface requirement pertains to vertical interfaces. Vertical interfaces, like horizontal interfaces, fall into two categories: 1) inter-technology interworking, and 2) interworking between service and network providers. Inter-technology interworking provides standards for technology within the same domain. VoIP is transmitted in packets, thus the packets must be transmitted over layers within the same network. Inter-technology interworking supports the communication among the various layers within the same domain (Travers & Swale, 2001). Interworking between service and network providers places standards on the service and network providers to ensure that communication between these providers is unified.

An explanation of standards is important to this dissertation because it provides a clear understanding of integration among hardware, domains, networks, and service providers. Without standards, a VoIP-initiated call may not be completed, or call quality may decline.

Cordell, Potter, and Wilmot (2001) focused their research on the H.323 standard. H.323 is a standard used by the telecommunications and computing industries that represents voice-over data (VoD) technologies. H.323 is better known as a VoIP standard, which enables voice, data, and video to be sent over a packet-based network (Cordell et al., 2001).

VoIP has three major standards, including: 1) H.323, or packet-based multimedia standards, 2) session initiated protocol (SIP), and 3) International Telecommunications Union (ITU) standard H.248, which focuses on decomposed multimedia gateways (Cordell et al., 2001).

Cordell et al. stated, "H.323 and SIP are competing standards because they cover the same VoIP problem space–passing voice, video, and multimedia traffic over IP networks and implementing associated services" (Cordell et al., 2001, p. 92). They explained that H.323 and SIP are derived from two different areas. H.323 comes from the telecommunications sector, and SIP comes from the computing sector (Cordell et al., 2001).

H.323 implements five major phases of call flow. These phases are:

1. call set-up,
2. initial communication and capability exchange,
3. establishment of audiovisual communication,
4. call services, and
5. call termination (Cordell et al., 2001).

"One of the main differences between H.323 and SIP is the way messages are encoded. SIP uses a text-based method of coding messages akin to HTTP" (Cordell et al., 2001, p. 103). Cordell et al.'s research concluded that H.323 and SIP are competing standards, with little dominance of one standard over the other. This research is important to the study of VoIP because it evaluated two competing standards for integrated communications.

Wisely's (2001) research focused on the session initiated protocol (SIP) and the control SIP has over VoIP calls. SIP deals with session initiation, however, SIP does not specify anything about resource reservation or security (Wisely, 2001). SIP is based on the hypertext transport protocol (HTTP), which is the same protocol used by Web applications. SIP negotiates contacts between one or more entities (Wisely, 2001). Wisely concluded that SIP is an excellent protocol for PSTN providers because, it allows PSTN providers to control the sessions (Wisely, 2001). Controlled sessions allow for greater management over admission control, which reduces the risk of network overload, dropped packets, and delay. Wisely's research is important because it enabled the PSTN providers to ensure that network traffic moves smoothly.

Stephens and Cordell (2001) focused their research on interworking VoIP networks with SIP and H.323. SIP and H.323 are two separate technologies. However, for the two technologies to work together, generic standards must be obtained (Stephens & Cordell, 2001). Stephens and Cordell concluded that interworking is technically feasible even though interested parties from different standards bodies are far from this view (Stephens & Cordell, 2001).

Hutchinson (2001) analyzed the telecommunications in the United States from an historical perspective. Hutchinson argued that since the 1984 "breakup and reorganization of AT&T into seven regional Bell operation companies (RBOCs), there has been a revolution in quality management within the telecommunications industry. With divestiture," Hutchinson continued, "RBOCs were required to stop manufacturing products. Ameritech, Verizon (formerly Bell Atlantic), BellSouth, Nynex, Pacific Bell, Southwestern Bell and US West were required to purchase all materials from suppliers, as they needed a way to control the quality of products shipped to them. As a result, Telcordia Technologies (formerly known as Bellcore), the RBOC owned consulting organization, initiated a supplier quality management group for the industry" (Hutchinson, 2001, p. 33). In the late 1980s the Customer Supplier Quality Program (CSQP) emerged from Telcordia (Hutchinson, 2001). The CSQP approach meets quarterly with RBOCs and Telcordia to review supplier progress. This approach was maintained for five to seven years, until a shift occurred as a result of this approach (Hutchinson, 2001). First, there was slow progress in certifying companies in CSQP. Second, the program was expensive. And last, emergence of ISO 9000 became available (Hutchinson, 2001).

In 1997, a new approach, the development of the TL9000 standard, took precedence within the industry. The TL9000 standard developed a new forum known as the QuEST Forum (Hutchinson, 2001).

Dandekar and Hutchinson (2002) explained the rationale behind forming the Quality Excellence for Suppliers of Telecommunications Leadership (QuEST) Forum. The QuEST Forum is designed as a joint effort between telecommunications service providers and suppliers centered around one common goal, which is to improve the quality of products and services for end-user(s) (Dandekar & Hutchinson, 2002).

The QuEST Forum introduced a single quality standard called TL9000. Telecommunications service providers and suppliers designed the TL 9000 standard. Over the course of a year and one-half, service providers and suppliers analyzed available quality standards and implemented the best standards into the new TL9000 standard (Dandekar & Hutchinson, 2002). TL9000 standards are composed of both requirements and measurement (Dandekar & Hutchinson, 2002). The measurement aspect of TL9000 is the most unique, Dandekar and Hutchinson showed, in that companies need to have a required system to measure, analyze, and report measurements (Dandekar & Hutchinson, 2002).

The QuEST Forum originally had membership based in North America; however, over the past few years it has been a globally accepted organization that focuses on improving the quality of products and services globally (Dandekar & Hutchinson, 2002). Dandekar and Hutchinson's research on QuEST is important to this dissertation because it focuses on creating standards and working globally to bring the telecommunications providers and suppliers together. Making standards that enable providers and suppliers to work together is accomplished by utilizing new metrics for measurements and requirements. QuEST realizes that without uniformity protocols, technologies such as VoIP may have to be researched on an infinite basis to provide QoS.

VoIP Related Research

Jiang (2003) utilized the Bernoulli, 2-State Gilbert, and General Markov Models in his research. These models are based on analyzing the real-time delays that affect the Quality of Service (QoS) standards of VoIP. The Bernoulli Model used by Jiang examines packet loss, which is defined as data loss normally resulting in congestion buildup. The 2-State Gilbert Model has two states: one corresponds to the number of packets that have been received, and the other corresponds to the number of packets that have been lost. The General Markov Model is used to identify dependencies on a finite number of previous events.

Jiang related his QoS standards in both objective and subjective terms. Objective QoS standards utilize metrics for loss, delay, and jitter (low or no delays in variation) on the VoIP network. Jiang's subjective data used Mean Opinion Scores (MOS), which are applied to an algorithm for measuring subjective data. The measuring instrument Jiang used was an E-model, which estimated perceived quality of network performance. Jiang derived this subjective data by measuring a random sample of participants listening to pre-recorded audio clips.

In this experiment, Jiang implemented a grading chart using a range of grades from one to five. A score of one was defined as "communications breakdown" (Jiang, 2003, p.

44). A score of two meant "strong noise or distortion, difficult to understand" (Jiang, 2003, p. 44). A score of three indicated "some audible noise or distortion, some efforts required" (Jiang, 2003, p. 44). A score of four meant "minimum requirement on most wire-line telephone networks" (Jiang, 2003, p. 44). A score of five he defined as "like a perfect AM radio reception" (Jiang, 2003, p. 44). Jiang explained that on a traditional landline network an MOS of 4.0 is the expectation, whenever the MOS drops below 3.5, users will typically start to be dissatisfied (Jiang, 2003). This means calls below 3.5 will result in less than acceptable call qualities.

Jiang indicated that the most important aspect to VoIP quality is packet loss (Jiang, 2003). Jiang concluded his study by indicating that packet loss should be kept below 3.5. An MOS value of 3.5 is the turning point where many users will become dissatisfied, as a result, packet loss rates should be kept below 3.5 to maintain an acceptable QoS level (Jiang, 2003).

Jiang's research is relevant to this dissertation because he analyzed network loss, delay, and jitter on VoIP networks and provided an example of how Relative Perceived Call Quality may be measured. "Among all Internet real-time multimedia service," Jiang stated, "VoIP is among the most critical applications as it demands high service availability" (Jiang, 2003, p. 3).

Markopoulou (2002) hypothesized that wide area networks (WAN) cause high jitter delay, periodic delay, and long loss periods. Markopoulou's research focused on measurements taken over the WAN to point out potential problems in the WAN network carrying VoIP transmission. She conducted an experiment using user perceived quality measures. Markopoulou concluded that many of the problems in WAN-based VoIP transmission stemmed from reliability, network protocols, router operation, and insufficient consideration of network traffic load. Paths of the same provider were observed to have similar delays with regard to loss, patterns, and ranges in short or long distances. This was hypothesized since many paths of the same providers may share elements within the same network (Markopoulou, 2002). Markopoulou's research is important because it analyzed the operational efficiency of the Internet backbone used to transmit real-time applications such as VoIP. Markopoulou concluded that many applications on backbone networks are sufficiently provisioned for data applications, but that this was not always the case for voice and video traffic (Markopoulou, 2002).

Houck, Kim, Uzunalioglu, and Wehr (2003) focused their research on presenting an algorithm that rejects VoIP call requests in congested gateways. According to Houck et al., the same network infrastructure has witnessed the rise of Internet protocol (IP) networks, as a result, a strong movement was made towards unifying the transport of voice and data within these networks (Houck et al., 2003). Houck et al.'s research was dedicated to placing an algorithm in VoIP gateways to determine the degree of congestion before a VoIP call is made. When network congestion occurs, this will affect all calls in progress, as a result, the calls will experience packet loss, delay, and jitter (Houck et al., 2003). This packet loss will affect the quality of VoIP calls placed on the network.

The measurement-based call admission control (MBCAC) algorithm operates in the gateways of the network. The algorithm works through the real-time transport protocol (RTP) and observes call quality statistics and flow at different paths if congestion occurs. Voice gateways are converted from the PSTN switch to IP packets using time division multiplexing (TDM) (Houck et al., 2003). The IP packets follow different network paths to arrive at the endpoint. Houck and colleagues' research is significant because it focuses on allocating VoIP calls to different networks if congestion occurs, reducing delay, jitter, and packet loss. Many key problems have prevented VoIP from a wide deployment by the service providers. The key reoccurring problems are packet loss, delay, and delay jitter constraints, which ultimately affect quality of service (QoS) (Houck et al., 2003).

Three other researchers, Bordetsky, Brown, and Christianson (2001), tried to address this problem by focusing their work on a feedback control model for quality of service in multimedia communications. They based their research on the observation that there are no guarantees for minimum or maximum delays on packet-switched networks (Bordetsky et al., 2001). Bordetsky et al. focused their attention on research conducted on real-time applications such as video and audio. As such, their research is relevant to VoIP by using Real Time Control Protocol (RTCP). RTCP performs quality of distribution monitoring, intermedia synchronization, and participant identification (Bordetsky et al., 2001). Receiver reports (RR) are generated by each receiver; these reports indicate loss ratios, jitter, and the highest sequence number from the source. These reports also allow the participants of a call to obtain valuable information and thus focus their efforts on exploring network deficiencies (Bordetsky et al., 2001). By identifying these deficiencies, the participants may be able to identify certain areas within their network to enable better call quality.

Doshi, Eggenschwiler, Rao, Samadi, Wang, and Wolfson (2003) researched mechanisms that provide quality of service (QoS) in a VoIP environment. Their research lists six critical elements necessary for VoIP to achieve the same grade of service as public switched telephone networks (PSTN).

In particular, Doshi and colleagues advised, "to achieve a PSTN grade of service in VoIP networks the following requirements are critical:

1. Global reach of PSTN and VoIP endpoints via multiple administrative domains (Ads).
2. Signaling mechanisms that support rich and flexible services.
3. Guarantee PSTN-grade voice quality as measured by mean opinion score (MOS).
4. Mechanisms to make sure the calls in progress are not adversely affected by new calls.
5. Traffic/resource engineering to provide a bound of the fraction of call setup requests denied at busy hour load.
6. Hardware and software architecture of network elements, routing and alternate routing mechanisms, and network management to enable high service availability.

Given good codecs, requirement three above is met if the packet delay, jitter, and losses are kept sufficiently small" (Doshi et al., 2003, p. 42). Codecs are defined as the algorithms used for transforming voice transmitted into bits at various speed rates. The focus of their research was to develop AbsoluteQOS™, which is obtained by making sure that calls in progress are not affected by new calls, and that call requests are denied on a busy load (Doshi et al., 2003). AbsoluteQOS is defined as the ability to provide a QoS guarantee on call blocking and a quantitative guarantee that calls-in-progress will receive (Doshi et al., 2003). There are certain considerations when looking at QoS in a VoIP environment. Voice users expect a rapid call establishment within fractions of a second when picking up the phone, and the connection to be made within seconds after dialing all digits (Doshi et al., 2003).

There are three main metrics in a VoIP environment for QoS standards. “Loss” is defined as packets that are lost between the transmitter and receiver, reducing the call quality of the call in progress. “Delay” is defined as the transformation of the voice to packets (or “mouth to ear”) causing a delay in the voice transmission, which affects QoS standards. The last metric is “Delay Jitter.” Delay jitter measures the variations of delay in the data units (Doshi et al., 2003). Delay jitter can contribute to packet loss and delay, causing the quality of the conversation to diminish.

Absolute QOS focuses on reserving space on a circuit with a predetermined specified bandwidth range for the call in progress. Allocating certain bandwidth requirements, Doshi et al. proposed, will keep jitter, delay, and delay jitter within certain specifications, thus providing a quality VoIP call. A counter is proposed to keep the circuit from receiving too many calls. If there are too many calls on the circuit, the call is denied to ensure that other calls in progress are not affected by an overload of calls coming in. Doshi et al.’s research is important, because it focused on network and admission control to the network that the VoIP conversation operates over, affecting the quality of the conversation. Public switched telephone network (PSTN) grade quality must be achieved in a packet switched network without excessive over-provisioning (Doshi et al., 2003).

Thorne’s (2001) research focused on several aspects of delays that happen on VoIP networks. “There are three main reasons why delay matters in a voice network,” Thorne stated. These include “the effects of echo, the impact upon conversations and the requirements for achieving interconnection between networks” (Thorne, 2001, p. 33).

Echo is defined as a delay that exceeds a few milliseconds (ms) in the conversation. As a result of excessive delay, the speaker will hear a delayed version of the conversation. Echo delays need to be kept under a minimum of five ms, otherwise, the conversation is very difficult to understand (Thorne, 2001).

Conversational interaction is defined as natural pauses in a conversation between the call initiator and call receiver. When the delay exceeds a certain value, the call quality will diminish. As a result, the tempo of the conversation can become upset. The ITU guidelines suggest keeping end-to-end delays below 150 ms to avoid this problem (Thorne, 2001).

Interconnect agreements are agreements that enable connection among carrier networks. Proper connections–established through interconnect agreements–links one network to another, enabling entire calls to be completed. In order to achieve an acceptable service quality, the delays need to be known (Thorne, 2001). Delay in the transmission of VoIP must be planned. "It is commonly assumed that delay is only due to packetisation effects" (Thorne, 2001, p. 34). There are several elements that must be considered when analyzing the delay elements. "The main elements of delay include," according to Thorne,

- encoding, the time taken to digitally encode the voice samples–this can be significant for aggressive compression schemes, particularly those that use block encoding techniques,
- packetisation, the time taken to accumulate a packet's worth of data,
- serialisation time, the time taken to clock the packet's contents on to the transmission line,
- packet queuing–as VoIP systems support multiple simultaneous conversations, any given packet may have to wait for access to the transmission system, and furthermore, as the service offering would normally be voice and data, voices packets may have to queue behind data packets,
- DSL transmission delay–the 'speed–of-light' delays across an access network are typically very small, and so the DSL transmission delay is dominated by the transmission system itself,

- core network delay–the time taken to transfer the packet to the far-end access system or interworking unit,
- decoding/transcoding delays at the far end or some intermediate point(s),
- dejittering buffer–although some of the above delay components are fixed, some are variable, e.g., queuing and network delays, and it is therefore necessary to provide a jitter buffer that essentially allows for the worst-case variable delay, otherwise output buffer starvation can occur which results in voice samples not being available for play-out; this can have very bad effects of perceived quality. (Thorne, 2001, p. 34)

Thorne's research delineated the potential causes of delays that are possible in a VoIP network, particularly copper networks, and is, thus, highly valuable for this dissertation. Thorne lamented, "Various mechanics have been devised to overcome such problems but as yet these are not widely implemented. In fact, the most common way of providing 'QoS' in IP networks is to significantly over-provision bandwidth. This is simply not an option in a bandwidth-restricted copper access network" (Thorne, 2001, p. 34).

Uhl (2004) focused his research on two significant problems that happen in VoIP networks: 1) runtime discrepancies among packets and 2) packet loss. Runtime discrepancies between packets are the loss in real-time transmission within the packets which carry the voice call. Packet loss means the loss of packets during a conversation from the call initiator to the call receiver. Uhl's research proposed four methods to achieve Quality of Service (QoS) in VoIP.

The ITU recommended the Mean Opinion Score (MOS), one of Uhl's four recommended methods, in 1996. MOS evaluates the quality of speech in a VoIP communication based on subjective appraisal from a number of subjects (Uhl, 2004). Uhl comments, "This method was the first method of note for measuring quality of speech" (Uhl, 2004, p. 3). A MOS test has three different segments: the listening opinion test, the interview, and the survey test.

The Perceptual Analysis Measurement System (PAMS) was developed by British Telecom to evaluate perceived speech quality. PAMS was the first model to use objective methods to evaluate speech quality in end-to-end networks (Uhl, 2004). The PAMS model measures two different aspects of call quality: 1) listening quality and 2) listening efforts.

KPN Research in the Netherlands developed the Perceptual Speech Quality Measure (PSQM). PSQM evaluates the speech quality of a telephone conversation through the human sense of hearing. This method is used most commonly in networks using speech compression with regard to packet based transmission (Uhl, 2004).

Perceptual Evaluation of Speech Quality (PESQ), developed by British Telecom and KPN Research, is a method used to evaluate the quality of speech in telephone networks. PESQ focuses on speech quality and speech codecs. In the PESQ model, the human element is replicated by transmitting a reference signal over both endpoints of the telephone line (Uhl, 2004). Uhl writes, "Any interference acting on the speech signal during transmission over the connection being tested will cause the test signal to differ from the reference signal. So that the signals can be compared with each other, a perceptive hearing model is used. This model is a replica of the human hearing apparatus and is actually very good" (Uhl, 2004, p. 3).

Uhl's research is relevant to this dissertation because network loss and measuring devices used to measure the network loss are researched. If packets cannot be handled within a certain time frame, jitter will increase, resulting in a decreased quality of transmission (Uhl, 2004).

Reynolds and Rix focused their efforts on network design choices and how network design can limit the perceptions of VoIP. Customers will strongly influence the future of

VoIP by evaluating the call quality with that of the PSTN (Reynolds & Rix, 2001). Reynolds and Rix identified three influences on a user's perception of quality:

- the VoIP device,
- network performance, and
- end-to-end delay, which includes both termination points.

VoIP has the potential to deliver voice and data services more efficiently than that of the PSTN (Reynolds & Rix, 2001). Reynolds and Rix used two methods to test end-to-end speech quality:

- Subjective tests, which seek the average opinion of a group of human users.
- Objective tests, which use comparison methods based on a known reference signal from endpoint-to-endpoint (Reynolds & Rix, 2001).

An MOS was calculated by averaging the votes of all subjects (Reynolds & Rix, 2001). Objective tests are used to measure the physical factors of the system. This can be accomplished by emitting test signals or monitoring live traffic.

Reynold's and Rix's research is important because it was focused on achieving high quality VoIP calls. Because there are various applications that are transmitted over the network, there is no easy way to specify how to achieve this. One application may be specified, however, this may be under- or over-specified for another application (Reynolds & Rix, 2001).

Amiri (2003) focused his research on bandwidth packing problems in telecommunications networks using an M/M/1 model. Packing problems refer to the proper routing of telecommunications calls during peak periods when alternative routes may be incorporated. The rationale for carriers incorporating alternative routes is to maximize profits. "Typically, the network topology, the capacities of the links, the unit flow costs on the links, the maximum allowable queuing delay in the network, the call table and revenue and traffic requirements of demands of each call are given" (Amiri, 2003, p. 113).

Amiri used a series of calculations that were taken from 500 examples. The M/M/1 model represents a good approximation for the actual queuing delay and for any other measurement of network performance (Amiri, 2003). Amiri concluded that by using a unified model of the problem, which considers all possible paths for each call, congestion can be minimized and streamlined. This research is relevant to the dissertation because it focused on traffic congestion and resolutions during peak times.

Peelen, Zivkovic, Bijwaard, and Teunissen (2003) focused their research on supporting quality of service (QoS) standards in wireless and wired access environments. Traffic must be regulated to protect it from fluctuating bandwidth environments. QoS in networks is attributed to the classification of packets and the flow of these packets (Peelen et al., 2003). By controlling the packets, Peelen et al. discovered that the bandwidth allocation could also be controlled.

Peelen et al. focused their research in a variety of areas, notably on fluctuation in available bandwidth. Available bandwidth will randomly fluctuate with network traffic. As a result, the available bandwidth is also random (Peelen et al., 2003). Peelen et al. proposed a device that allows for the admission control of traffic. This device monitors bandwidth and accepts or denies various sources from using this bandwidth. For example, if a heavy bandwidth application tries to enter a busy network, access may be denied, but if a low bandwidth source enters a busy network, access may be accepted. The admission is based on how much bandwidth the network has to allocate during a specific point in time. This research is relevant to this dissertation because it investigates bandwidth resources and the proper allocation of the bandwidth that is needed to effectively transmit VoIP-enabled calls.

Kraimeche (2001) analyzed cell loss and call blocking at an ATM-based DSL access multiplexer, which integrates two types of calls on a single channel, through a common buffer. Type-1 calls are voice calls; the calls have stringent delays, but relaxed cell loss Quality of Service (QoS) requirements. Type-2 calls represent Internet data; the calls have stringent cell loss, but have relaxed delay QoS requirements (Kraimeche, 2001).

Asynchronous Transfer Mode (ATM) is the most efficient packet-switching technology for backbone networks (Kraimeche, 2001). ATM technology multiplies various applications, as a result a QOS is guaranteed (Kraimeche, 2001). The Central Office (CO) ADSL modems demodulate the data and voice ATM cell streams, which, in turn, are buffered into a multiplexer (Kraimeche, 2001). The multiplexer, along with the ADSL modem is known as the DSL access multiplexer (DSLAM) (Kraimeche, 2001). After analyzing the two variables as stated above, Kraimeche concluded that ATM-over-xDSL is a viable solution to integrate voice, data, and other applications onto the xDSL access network (Kraimeche, 2001). Kraimeche's research is useful because it supports the integration of voice and data onto one network.

Mandjes and Uitert (2000) focused their research on utilizing the measurement-based admission control (MBAC) algorithm for "bursty" sources to gain admission control. Bursty sources are large numbers of calls coming in at once. The algorithm will attempt to regulate the networks' loads to provide the customer with sufficient quality of service (QoS) (Mandjes & Uitert, 2000).

The research analyzed the robustness of the MBAC algorithm with respect to the characteristics of the traffic patterns (Mandjes & Uitert, 2000). The approach used in this research was conducted using a series of measurements based on both the call and burst levels. Mandjes and Uitert concluded that more research needed to be done on MBAC algorithms. This research is important because the MBAC algorithm provides admission controls for the caller. With an overloaded or an uncontrolled network, this may have a significant impact in regard to call quality.

Parr and Herron (2001) evaluated VoIP technology with regard to a hybrid PSTN/ VoIP application solution. The recorded information distribution equipment (RIDE) system was deployed in the late 1980s, as a result, economic mass access to recorded announcements was provided (Parr & Heron, 2001). In the 1990s the RIDE architecture was revamped to integrate application and voice-processing capabilities such as VoIP (Parr & Heron, 2001). A common constraint within VoIP QoS is the routing system; the RIDE architecture enables a routing table to determine the route the call will take. RIDE specifically addresses VoIP engineering in three key areas: 1) using an IP multicast, which minimizes bandwidth, 2) utilizing network equipment that supports QoS standards, including integrating different servers and multi-protocol label switching, and 3) applying traffic engineering to achieve optimal network design (Parr & Heron, 2001). "The RIDE IP network is a 'closed' network with predictable traffic flows (unlike the public Internet), and the use of multicasting helps to minimize this traffic" (Parr & Heron, 2001, p. 155). This research is important because a local area network that does not have constraints on traffic flow, especially over an IP platform, can impact the call quality as perceived by users.

Massoulie´ and Roberts (2000) focused their research on bandwidth sharing, with regard to allocating more bandwidth to accommodate traffic. Traffic can be categorized into two types: 1) stream and 2) elastic. Stream flows carry voice and video traffic that are preserved as the flow passes through the network. Elastic flows transfer the digital object at a rate up to the limits of the link and system capacity (Massoulie´ & Roberts, 2000).

Controlling admission to the network is a commonly used practice by telecommunication carriers. Admission control allocates certain bandwidth requirements on the network, and many times, to keep all traffic flowing through the network, reduces the amount of bandwidth available for a certain application. Admission controls may slow the network down during peak periods, and applications such as VoIP will experience delays in the voice transmission. For elastic traffic, it is commonly understood that it is better to reduce overall flow rather than to reject new flow (Massoulie´ & Roberts, 2000).

VoIP calls are broken down into packets of information and forwarded through the Internet. The possibility exists that a congested link may see repeated usage for new flows, even though an alternative path that may offer much greater throughput may be open (Massoulie´ & Roberts, 2000). The research concluded that the admission should admit no more than the flows that are compatible with the minimum acceptable throughput (Massoulie´ & Roberts, 2000).

Patek, Liebeherr, and Yilmaz (2003) addressed two distinct network designs meant to achieve guarantees on end-to-end delay. The two designs they researched are called class-level aggregation and path-level aggregation. Recently, much work has gone into devising QoS guarantees in packet networks (Patek et al., 2003). Deterministic services, which guarantee worst-case end-to-end delay bounds for traffic, produce an inefficient use of network resources. Statistical services, on the other hand, which violate QoS specifications, achieve an increase in the utilization of network resources (Patek et al., 2003).

Class-level aggregation, also known as the "jitter control method," utilizes nodes to specify a buffer size, and uses an algorithm to determine a minimum bandwidth for each class of traffic. When traffic arrives, it is provisioned to the length of the buffer; if the traffic is in violation of the buffer size, the traffic is dropped. A disadvantage of class-level aggregation is its requirement for jitter control at each node (Patek et al., 2003).

Path-level aggregation utilizes the same buffer, but all traffic has to be in the same traffic class and have identical end-to-end routes (Patek et al., 2003). Hence, the name "path-level aggregation," as this traffic class needs to follow the identical path. "At the network entrance, there is one traffic conditioner for each pipe. The traffic conditioner discards that portion of the aggregate traffic which does not comply to a given policing function" (Patek et al., 2003, p. 21). A separate buffer for each pipe exists at the network node. As a result, flows of the same traffic are buffered only if they have the same end-to-end path. A disadvantage of path-level aggregation is its inability to perform jitter control (Patek et al., 2003).

Patek et al.'s research concluded that class-level aggregation is worth the price paid in terms of enforced delay. Path-level aggregation continues to diminish as the network load increases. As a result, worst case scenario QoS guarantees are implied (Patek et al., 2003). Because this research concentrates on jitter control, which can reduce QoS in VoIP, it is important to this dissertation.

Klincewicz, Schmitt, and Wong (2002) researched ways to incorporate QoS into IP enterprise network designs. Currently, many companies are integrating voice and data onto one unified network. QoS over IP can be treated based on the class of service, or provide guaranteed performance for certain classes (Klincewicz et al., 2002). The treatment for class of service allows certain classes to receive priority over other classes. In contrast, guaranteed performance guarantees certain traffic classes to share the available resources (Klincewicz et al., 2002). Klincewicz et al. proposed that an IP network have various classes or services, different routing paths based on the class of service, and models that queue and prioritize differential delays. Although advanced algorithms were not used in this research, the proposed methodology does help offset network congestion. Network congestion causes a delay in both voice and data applications, ultimately affecting QoS.

Tobagi, Markopoulou, and Karam (2002) assessed the question "Is the Internet ready for VoIP?" (Tobagi et al., 2002) Their research was designed to measure packet loss and delay jitter characteristics. RouteScience Technologies, Inc. measured seven providers throughout five United States cities, including San Jose, CA, Thornton, CO, Newark NJ, Ashburn, VA, and Andover, MA (Tobagi et al., 2002).

The first measurement they made was of packet loss characteristics. Packet loss is broken down into 1) elementary packet loss and 2) complex loss events. Elementary packet loss indicates that one or more packets are lost, and these consist of consecutive probes separated over long periods of time. Complex loss, on the other hand, means that several elementary probe loss events have occurred over short periods of time (Tobagi et al., 2002).

Tobagi et al. analyzed elementary packet loss, in which the path incurred 27 elementary events, 20 of which were single packet losses, while seven consisted of 17 to 24 packets lost. The identical path incurred five complex events within a time frame of 20 to 60 seconds, with a loss rate calculated at 19 to 24%. The complex loss event was measured over a period of 50 seconds. As a result, the packet loss rate during that period was 24.6% (Tobagi et al., 2002).

Delay and delay jitter measurements were analyzed. Using the identical network, Tobagi et al. discovered that spikes occur approximately every ten minutes. This time frame coincides with the packet loss observed above (Tobagi et al., 2002).

Quality voice communications are referred to as Mean Opinion Score (MOS), which is a subjective rating given by listeners (Tobagi et al., 2002). MOS is measured on a scale based on a numerical value of one to five. A score of 1.0 is not recommended; 2.6 indicates that nearly all users are dissatisfied; 3.1 indicates that many users are dissatisfied; 3.6 is acceptable, however, some users are dissatisfied; 4.0 indicates a satisfied and desirable level; and 5.0 includes very satisfied users (Tobagi et al., 2002).

Quality of speech is affected by the encoding process, loss of speech, echo, and mouth-to-ear delay, which is also known as delay (Tobagi et al., 2002). Tobagi et al., like many other researchers, concentrated studies on loss of speech, which is an occurrence affected by delay and packet loss.

Tobagi et al. concluded that VoIP is a feasible network. However, much more work needs to be done on network reconfiguration, router and internal operations, and protocol exchanges. These aspects need to be fixed before VoIP can become a quality telephone network (Tobagi et al., 2002).

Karam and Tobagi (2000) researched different traffic types of services over the Internet. The Institute of Electrical and Electronic Engineers (IEEE) and the Internet Engineering Task Force (IETF) have proposed protocols and mechanisms that can enable the Internet to provide different support for traffic classes (Karam & Tobagi, 2000).

Karam and Tobagi used a computer simulation experiment consisting of a network scenario and traffic scenario (Karam & Tobagi, 2000). Karam and Tobagi researched several aspects, but the most important for this dissertation is the measurement they took that combined voice and data over a T1 link, with platforms of 10Base-T and 100Base-T links. They concluded that voice and data applications on a T1 link are nearly impossible if delay requirements are to be satisfied (Karam & Tobagi, 2000). Their study indicated that voice and data must be separated, or, at a minimum, related to different classifications of traffic (Karam & Tobagi, 2000).

Fraleigh, Tobagi, and Diot (2003) suggested that latency sensitive traffic can be differentiated and suggested that this traffic should be prioritized to provide enough bandwidth to support the most stringent delay requirements (Fraleigh et al., 2003). Traffic on the Internet is very dynamic. Some applications such as voice require real-time capability, otherwise, delay and jitter are evident. In contrast, some applications such as email do not have the same stringent requirements.

Traffic classification can prove to be very costly from both a network and support aspect (Fraleigh et al., 2003). The alternative method is to provision the network to allow enough bandwidth to support all applications. The backbone network is connected by sets of nodes, also know as Points-of-Presence (PoPs), which are connected by high speed links (Fraleigh et al., 2003). Internet Service Providers (ISP) must set bandwidth requirements.

The ISP must know the traffic demand set on each pair of POPs, as well as the network that each of these POPs follows (Fraleigh et al., 2003).

Fraleigh et al. introduced a Capacity Assignment (CA) problem to determine the sufficient bandwidth that each network must support between the POPs and the network. After analyzing several aspects of the network, Fraleigh et al. concluded that provisioning 5 to 15% on the Sprint IP network would allow the network provider to reduce the delay by 4 ms. Yet even with this decrease, voice quality would not reflect significant improvement (Fraleigh et al., 2003).

Markopoulou, Tobagi, and Karam (2002) analyzed the quality of VoIP over Internet backbones. They took voice quality measurements using an Emodel, standardized by ITU-T. The Emodel is used to predict the subjective quality of packetized voice (Markopoulou et al., 2002).

Markopoulou et al. classified two call durations: 1) 150 short calls, which had a 3.5-minute mean and 2) 50 long calls, which had a ten-minute mean. During the time between 14:00 and 15:00, the worst rating using a MOS scale was 1.1%. Ten percent of the calls had a MOS equal to or greater than a MOS of 1.4%. 50% of the calls were equal to or greater than a MOS rating of three. The remaining calls had a MOS rating that was equal to or greater than 3.75%, which is defined as acceptable on the MOS scale (Markopoulou et al., 2003). This data was based on 200 calls.

Markopoulou et al. concluded that backbone networks are over-provisioned, however, voice should be given preferential treatment and a playout buffer scheme needs to be used to match the delay pattern from end-to-end.

Similar research was conducted by Markopoulou, Tobagi, and Karam (2003). This research analyzed the quality of voice over Internet backbones. Internet communications is a universal system that carries voice, data, and video. Voice, however, seems to be of greatest importance because of the revenue the telephony industry has generated (Markopoulou et al., 2003). The research concluded that backbone networks are sufficiently provisioned for data, however, voice traffic exhibited a different result (Markopoulou et al., 2003). Markopoulou et al. reported that some backbone networks have fairly good characteristics, but some networks have problems that are directly correlated with reliability and network operations (Markopoulou et al., 2003).

Kado, Ishida, and Yajima (2000) introduced quality of service guarantees involving different flows of network traffic. Kado et al. suggest that the flows on a single transmission route are limited by the node capacity (Kado et al., 2000). Kado et al.'s research is important to this dissertation because it analyzed different classes of services of the old architecture and proposed that the Classed and Chained Queues (CCQ) method be used. "Three technical preconditions necessitate the CCQ method: support for an increasing number of flows; flexible allocation of bandwidth values, including dynamic reallocation; and, for delay guarantees required for real-time applications, highly precise control of cell interval, with minimal inconstancy even after completion" (Kado et al., 2000, p. 156).

Knowles, Oates, and Corne (2000) suggested that evolutionary algorithms need to be applied in telecommunications. The two algorithms Knowles et al. researched are multi-objective PAES (M-PAES) and Pareto archived evolution strategy (PAES) (Knowles et al., 2000). PAES is essentially a climbing procedure that allocates one current single solution, then adds another solution. As a result, PAES maintains the current solution as a base to test the new proposed solution (Knowles et al., 2000). Knowles et al. further states, "M-PAES is a recently proposed memetic multi-objective algorithm, which has already been found to outperform sophisticated alternative approaches on some difficult multi-dimensional knapsack problems" (Knowles et al., 2000, p. 55). The research that pertains to this dissertation focused on offline routing on packet-switched networks. Offline routing are alternative routes that traffic can seek from a cost and congestion scenario. As a result, if an offline route is available, this can minimize congestion (Knowles et al., 2000). Knowles et al.'s research is relevant because it focused on the routing problems that may be affecting voice calls made over IP networks today. Strength pareto evolutionary algorithm (SPEA) uses both an internal and external population. Knowles et al. concluded that M-PAES and PAES are superior to SPEA in the majority of the tests run.

Whitehead and Williams (2002) examined overload controls for adaptive networks. "In telephone networks, overloads can be caused by the following (either singly or in combination):

- Media-stimulated mass-calling events – such as televotes, charity appeals, competitions and marketing campaigns,
- Emergencies,
- Network equipment failures,
- Auto-scheduled calling.

In the absence of effective controls, such overloads would threaten the stability of network systems, and cause a severe reduction in successful call completions" (Whitehead & Williams, 2002, p. 32).

Overload controls are regulated by the node suppliers and network operators. Two standards of overload controls are being advanced by the ITU-T. The first is adaptive ISUP ACC. The second is H.248, which is designed as a feedback control to adaptively throttle from fresh demand (Whitehead & Williams, 2002). Williams and Whitehead conclude that overload controls need to be automatic, fast-acting, and adaptive (Whitehead & Williams, 2002). Both internal and external overload controls are important in telecommunications to avoid network overload. Network overloads produce spikes of high call volume. As a result, packet loss and delay jitter occur with VoIP-initiated calls, resulting in poor call qualities.

VanderBrug (1999) described that customers may have a high expectation with VoIP in regards to quality of service (QoS). The high QoS demands of VoIP correlate directly with the reliable circuit switched networks that consumers have become accustomed to (VanderBrug, 1999). VanderBrug's research concluded that packet-switched hardware does not match the maturity of the circuit switched world (VanderBrug, 1999). The circuit switched world has a long history of reliability; as a result, consumers demand the same reliability and QoS for VoIP.

De Oliveira (2000) analyzed different methods of end-to-end speech quality. The three categories are: 1) subjective tests, 2) subjective measurements, and 3) objective measurements. The most notable for this dissertation are subjective tests and objective measurements. Subjective tests have an advantage of providing a subjective view, however subjective tests can prove to be quite costly and take a long time to perform (De Oliveira, 2000). The E model, which includes basic metrics to determine quality of service (QoS) levels is an objective measurement device. Objective tests may be best suited for long-time

QoS measurements (De Oliveira, 2000). The research is important to this dissertation because the Internet is generally regarded as a best effort service with few QoS guarantees. Because VoIP uses the Internet to transmit over, other quality designs must be implemented to provide QoS guarantees for VoIP services. The research concluded that QoS guarantees should be provided by telephony operators from a global evaluation system. The proposed global system would monitor real-time QoS, along with end-to-end QoS (De Oliveira, 2000).

Michels (2003) analyzed proper designs to achieve an acceptable VoIP network. Quality of Service (QoS) must minimize end-to-end delay, jitter, and prevent packet loss. QoS cannot be guaranteed on links that share bandwidth with other devices. The other devices can take valuable bandwidth away from the devices attempting to implement QoS (Michels, 2003). To achieve QoS in a VoIP environment the networks and equipment must be carefully analyzed and prepared to provide quality VoIP services. Michels states "IP-telephony implementations have proven successful, when accompanied by proper planning, preparation and a systematic approach" (Michels, 2003, p. 48).

Gautam (2002) centered his research on the tradeoff that is associated with the balance between controlling user traffic and guaranteeing QoS. QoS is defined in terms of packet loss rate, delay, delay jitter, and bandwidth. As a result, QoS must be guaranteed from start to finish (Gautam, 2002). A policing mechanism for QoS is a "leaky bucket." A leaky bucket is known as a management system that allocates traffic control onto the networks (Gautam, 2002). The "leaky bucket" uses two mechanisms for admitting traffic onto the network, a data buffer and a token pool. The data buffer allows external data traffic onto the network. The token pool operates generally at a fixed rate, however if the token pool is full, new tokens will be discarded (Gautum, 2002). If there are tokens in the token pool, the incoming tokens pass an equal amount of tokens into the network. In contrast, if the token pool is empty, two alternatives are introduced.

The first alternative is a buffered leaky bucket, whereby the packets wait in an infinite capacity data buffer until the tokens arrive. The second alternative is an unbuffered leaky bucket, whereby there is no data buffer. As a result, the packets arrive, however, if the packets do not match a token, a "violation" occurs, resulting in dropped packets when congestion happens (Gautam, 2002). Therefore, Gautum wrote, "the leaky bucket is usually located at the user end" (Gautum, 2002, p. 38).

Gautum's research concluded that a "leaky bucket" can be solved on the source end, which is usually owned by the same organization. However, more research needs to be done on network-wide global optimization, particularly relating to all classes of traffic and at different nodes on a private network. Gautum's research also demonstrated the introduction of a "leaky bucket" on both ends (Gautum, 2002).

Rosen (2001) described the Megaco architecture protocol as a gateway. Megaco was designed to be used in a wide variety of gateways. Thus, Megaco has an extension called a "Package" that accommodates the different varieties of gateways (Rosen, 2001). A Package is defined as a document that may come from a standards organization (Rosen, 2001). Rosen's description of the Megaco architecture is important to this dissertation because there are varying standards of gateways in telecommunications, and standards need to be defined for uniformity. Uniformity and standards in gateways may help in QoS measurements of VoIP calls.

Al-Shaer and Tang (2002) studied path monitoring for multicast networks. Al-Shaer and Tang argue, "Efficient monitoring tools are necessary to observe the health of the multicast delivery trees, fault report, and performance problems, like high latency or packet loss in the delivery path, unreachable members, and abnormal disconnections, which may occur due to routing misconfiguration or bugs in the protocol implementation" (Al-Shaer & Tang, 2002, p. 358).

Al-Shaer and Tang proposed the implementation of the monitoring using SMRM, which is defined as SNMP-based multicast reachability monitoring for multicast reachability (Al-Shaer & Tang, 2002). Al-Shaer and Tang argued that the SMRM architecture detects and discovers the cause of network problems, such as high latency or packet loss on a multicast network (Al-Shaer & Tang, 2002). The SMRM uses an interface which allows users to create one or more SMRM monitoring sessions, with the ability to create remote SMRM agents. Furthermore, SMRM allows users to view the results in real time or on a post-mortem basis (Al-Shaer & Tang, 2002).

Zubey, Wagner, and Otto (2002) focused on consumer aspects of voice communications technologies. Through a survey format, their research investigated which technologies are most preferred by customers. The surveys were sent to the customers via the Internet, with awareness created by news groups. The minimum sample size was set at one hundred (Zubey et al., 2002). The surveys included factors such as price, reliability, accessibility, voice quality, and services. There were 254 participants who completed the survey, of which 49% ranked reliability as the most important VoIP attribute (Zubey et al., 2002). Zubey, Wagner, and Otto's survey concluded that the consumers surveyed thought the call quality is an important attribute (Zubey et al., 2002).

Jiang, Koguchi, and Shulzrinne (2003) evaluated VoIP end-points. The research of Jiang et al.focused on the IP phones, which is the area of their research most relevant to this dissertation. Their research included the evaluation of IP phones from three major vendors –Cisco, 3Com, and Pingtel–and it analyzed the mouth-to-ear delay (M2E) for each type of phone. The experiment used a digital audio book tape (Jiang et al., 2003). After performing a large number of tests, the researchers concluded that the hardware IP phones achieve a low M2E delay, and have acceptable packet loss concealment performance (Jiang et al., 2003).

Wu, Hou, Li, and Chao (2001) examined integrated services through packet networks. "As the Internet transforms from the traditional best-effort service network into QoS-capable multi-service network, it is essential to have new architectural design and appropriate traffic control algorithms in place" (Wu et al., 2001, p. 135). Wu, Hou, and Chao's research included scheduling, call admission control, and buffer management on an integrated services (IntServ) network. To guarantee zero packet loss for guaranteed service (GS), an appropriate buffer must be allocated for each GS flow (Wu et al., 2001). Controlled-load service (CL) flows do not have hard delay requirements like GS flows, because CL flows operate on the premise of actual traffic. As a result, more bandwidth must be allocated (Wu et al., 2001). Call admission controls (CAC) are designed to maximize the network while simultaneously providing QoS guarantees for the admitted call flows. The research concluded that providing these, among other controls, resolves several problems, as opposed to the traditional class-based approach on IntServ networks (Wu et al., 2001).

Jiang and Shulzrinne (2002) compared forward error correction (FEC) to codec robustness. Because the Internet is a best-effort network, users must rely on end-to-end techniques to improve quality (Jiang & Shulzrinne, 2002). FEC recovers lost packets by transmitting redundant data. As a result, data delays and bandwidth increases occur (Jiang & Shulzrinne, 2002). After running a series of measurements utilizing the same bandwidth using a Mean Opinion Score metric, Jiang and Shulzrinne concluded, FEC is better when ignoring the delay (Jiang & Shulzrinne, 2002). When delay is considered, the robust codec performs better under low loss conditions (Jiang & Shulzrinne, 2002).

Jiang and Schulzrinne (2003) examined several metrics to assess VoIP service availability on the Internet. VoIP has increased in popularity, but the PSTN has a long history of reliability (Jiang & Shulzrinne, 2003). The assessment conducted by Jiang and Schulzrinne found: 1) packet losses are not rare events, 2) international paths generally represent a greater packet loss rate, 3) network outages are relatively short, though there have been long term outages, 4) the mean time to restore a network can be very high, and 5) the Internet two, which is primarily used for research, has lower delays and packet loss than the public Internet, but network outages are similar to those of the public Internet (Jiang & Schulzrinne, 2003).

Hei and Tsang (2002) researched the Earliest Deadline First (EDF) scheduler, along with a buffer management scheme (CHOKe). "In order to support communication services with QoS guarantee, network resources need to be managed in a systematic manner" (Hei & Tsang, 2002, p. 350). The FCFS-Droptail scheme is commonly implemented in IP routers (Hei & Tsang, 2002). FCFS limits the buffer size; as a result, packets are dropped at the router if there is an insufficient buffer size (Hei & Tsang, 2002). TCP performance can be increased using Random Early Detection (RED), which allows the average queuing delay to be controlled, while transient queue management queue-size fluctuation is allowed (Hei & Tang, 2002).

Scheduling schemes can be implemented to help with the aforementioned situation. "The packetized version of the Generalized Processor Sharing Scheduler (PGPS) can guarantee the minimum per-connection throughput and delay bound with flow protection, but it needs to maintain per-connection states and reserve large bandwidth for a small delay bound.... EDF has an advantage over PGPS in that it is able to schedule real-time traffic in terms of system scalability and utilization" (Hei & Tsang, 2002, p. 351).

Hei and Tsang used a Network Simulator to conduct an experiment using a single congested link. The designs that were studied in this experiment were the FCFS-Droptail, FCFS-RED, FCFS-CHOKe, EDF-Droptail, EDF-RED, and EDF-CHOKe.

Hei and Tsang concluded, "the Earliest Deadline First (EDF) scheduler schedules the real-time traffic. Simulation results show that the proposed EDF scheduler working with an active buffer management scheme can achieve a better delay performance and, at the same time, make a better bandwidth allocation between real-time TCP and UDP connections than the First Come First Serve (FCFS) scheduler with the Drop-Tail buffer management" (Hei & Tsang, 2002, p. 358). This research is important because it focuses on controls to guarantee QoS in a real-time environment.

Holly (2004) focused her research on three key types of measurements applicable to the analysis of VoIP's operational efficiency: pervasive, continuous, and comprehensive. Pervasive measurements were applicable to the entire enterprise network, continuous measurements were taken throughout time, and comprehensive measurements were implemented through the use of both active and passive monitoring. Active monitoring simulates actual usage of calls being made over the VoIP network. This is a proactive measure to focus on problems such as delay that can cause real-time transmission problems while using the VoIP system. Passive monitoring is taken on a continual basis to identify if any calls have poor quality and, if so, to identify the cause(s). Monitoring on a continual basis ensures any problematic issues with regard to the network are addressed immediately.

Pervasive measurements of the enterprise network were taken with hardware and software implementations on the origination and delivery points of the VoIP network using an enterprise-wide approach. This ensured that all of the points on the VoIP network were controlled. Holly concluded that proper quality control procedures must be taken to test all phases of the VoIP infrastructure. By doing so, businesses will be able justify their usage of VoIP technologies from both a cost and quality standpoint. However, Holly stated that "not doing so leaves mission-critical voice applications at risk for crippling service degradations and outages" (Holly, 2004, p. 4).

Comprehensive measurements of the enterprise network are used to provide meaningful data with respect to the transmission quality of VoIP. Holly's article did not provide descriptions of any statistical tests used for the measurements. Holly's research is relevant to this dissertation because it focuses on monitoring and testing measures that can be implemented to ensure a high quality VoIP enterprise network. "Enterprises everywhere are looking carefully at VoIP these days," Holly contended. "And no wonder: the economic advantages of sending voice and data over the same low-cost, IP network can be attractive" (Holly, 2004, p. 1).

Supporting Research

The supporting research ties VoIP, broadband, and long distance technologies together. As stated before, VoIP is a technology that can be efficient and cost effective, however, VoIP is a dependent technology particularly with regard to broadband. The supporting research is explained from a philosophical point of view and briefly described.

Roach (2004) analyzed the broadband increases on a global basis. The Yankee Group estimates that by 2008 there will be 325 million broadband subscribers, which is an increase of one hundred million from a report taken at the end of 2003 (Roach, 2004). The broadband increases are important to VoIP because VoIP rides over broadband networks.

Gubbins (2002) focused on innovative broadband Internet through a wireless network with no latency. (Latency is defined as delay time related to the transmission of data in a network.) Innovative broadband is defined as new ways that broadband resources can be transmitted in a non-traditional way to reach more users in an efficient way. Non-innovative broadband is defined as traditional media, such as cable or a DSL, in which broadband resources are transmitted. These networks operate with speeds as high as 1.5 megabytes per second (Mbps). Internet broadband speeds are an important factor in the performance of the VoIP technology because broadband Internet is the backbone over which VoIP must operate.

Leisen and Vance (2001) focused their research on the assessment of telecommunications service quality in the USA and Germany. The sample size consisted of 200 Germans, and 76 Americans who have telecommunications service in their respective countries (Leisen & Vance, 2001). Leisen and Vance conclude that there is much work to be done to provide acceptable service across multiple nations (Leisen & Vance, 2001).

Much of the variances resulted from different standards as indicated by the consumers. For example, in Germany the consumers indicated that reliability, responsiveness, and empathy were the most important factors, while in the USA the consumers indicated that reliability was the most important (Leisen & Vance, 2001).

Ford and Jackson focused their research on competition and investment in telecommunications. The 1996 Telecommunications Act required incumbent local exchange carriers (ILECs) to allow competitive local exchange carriers (CLECS) the use of their networks at regulated rates. "Three methods of entry relying upon wholesale access are: a) combining incumbent distribution with an entrant-supplied switching plant (UNE-Loop); b) using both incumbent distribution and switching plant (UNE-Platform); and c) reselling the incumbents' retail services (Resale)" (Ford & Jackson, 2004, p. 71).

Loube (2003) focused his research on the Universal Service Fund. The Telecommunications Act of 1996 altered the regulatory environment of the telecommunications industry; as a result, regulatory pricing was exchanged for lower market pricing (Loube, 2003). The Universal Service Fund (USF) helps support rural and high-cost area carriers. Two mechanisms, Model Support and Interstate Access Support, help both the larger carriers and the affiliated smaller carriers. USF is used to help fund the increased costs of the smaller rural carriers by offsetting the costs of telecommunications line charges. Determining the appropriate amount of universal service funding is a function that is difficult, partially due to changing costs in maintaining the networks (Loube, 2003). Large carriers that serve rural and high cost areas have sufficient funds available. Loube's research is relevant to this dissertation because it looked at the financial support needed to keep the rural markets and high cost areas afloat, which indicates the increasing expense of landline-based communications.

Couper, Heikel, and Wolman examined the boom and bust within the telecommunications industry. On the technological side, passage of the 1996 Telecommunications Act produced rapid growth in fiber optic technology. As a result, data capacity increased vastly (Couper et al., 2003). The growth of the Internet resulted in a greater demand for telecommunication services (Couper et al., 2003).

From 1996-2000 telecommunications companies in communications averaged $135 billion of profit per year (Couper et al., 2003). The final quarter of 2000 was the start of a seven quarter negative investment growth for telecommunications, the low point coming in the fourth quarter of 2001, at under $93 billion (Couper et al., 2003).

The boom in telecommunications came from tremendous advances in technology, e.g. the Internet, which stimulated a tremendous growth of fiber and new applications. However, despite the growth in technology, many of the basic elements of telecommunications remained the same. Copper wire continues to connect many consumers and businesses to the local switching offices. Much of the copper was used to transmit voice calls while much of the circuit remained unused. With the rise of digital communications many of the telecommunications providers are shifting to packet switched calls (Couper et al., 2003). After 1996 many competitive local exchange carriers (CLECs) entered the marketplace, but after July 2000 the CLECs revenues fell 63% (Couper et al., 2003).

With VoIP dependent on broadband access, the state of telecommunications companies is at the forefront. Without widely deployed broadband networks, including remote/rural areas, and increased network efficiency, VoIP cannot succeed as a universal service. The Telecommunications Act of 1996 promoted competition, however it decreased revenues for the carriers. This resulted in an overabundance of financial burdens for many of the carriers. With these financial burdens, broadband access in many rural/remote areas has been slow to deploy.

The Long Distance (LD) segment has been partially analyzed, and the results illustrate how it has driven LD prices down for the supporting local exchange carriers (LECs). Creswell (2004) stated that 40% of the voice traffic could be on VoIP by 2009. The LD research will be used as a guide for testing of the VoIP product. The LD research is critical, since this segment's operations continue to be inefficient, and because VoIP can create a greater operational efficiency.

Much of the VoIP related research focused on objective data, with regard to admission control and packet loss. By understanding how to control admission onto the network, researchers can have a better understanding of call flow and how to efficiently handle different volumes of call flow. When too many calls are allowed onto the network, congestion is created, resulting in decreased call quality or dropped calls.

When call flow volume is high, packet loss is ultimately affected. Because voice calls are broken down into packets of data, it is important to understand where the packets of data are being lost. With packet loss, the conversation greatly diminishes, affecting the quality of the call. Packet loss will result in delay and jitter, disrupting the normal flow and quality of the conversation.

Another important aspect to VoIP quality is uniform standards. Standards are needed to support admission control, which ultimately affects packet loss. Without standards a call may never reach the intended destination. Standards can help manage call flow, resulting in better call quality through a more efficient network, ultimately resulting in less packet loss.

Void In The Related Research

The void within the related research includes the lack of measurement, analyses, and conclusions of the difference in Relative Perceived Call Quality from the user's perspective. Subjective data, defined as data that is derived from human participation, has not yet been analyzed for Relative Perceived Call Quality of VoIP for responses provided by human Call Initiators. The purpose of this dissertation is to identify whether DSL-enabled VoIP calls are much worse, worse, the same, better, or much better than traditional landline-based calls from DSL users' perspectives. This assessment was achieved by randomly sampling a segment of the target population and using the resulting subjective data to analyze the Relative Perceived Call Quality of the DSL-enabled VoIP product in comparison to that of landline-based phones. This study is significant, since it measures both the users' initial preconceptions and the actual perceptions of the relative call quality of VoIP calls versus traditional analog calls.

Chapter 3: Methodology

The design methodology utilized was a pretest/posttest design. There were 56 randomly chosen Call Initiators and 56 randomly chosen Call Receivers who were asked to participate in this study. The assignment of the participants to the Call Initiators and Call Receivers groups were randomized. This randomization was accomplished for each pair of participants by placing two slips of paper in a non-transparent container, one of which was labeled with CI (indicating Call Initiator) and one of which was labeled with CR (indicating Call Receiver). One participant from each call round drew a slip of paper from the non-transparent container. The slip of paper drawn indicated whether the drawing participant was the Call Initiator (CI) or Call Receiver (CR). The other participant in the same call round was automatically designated to the opposite role. Both Call Initiators and Call Receivers were randomly chosen from the student population, faculty, and the general population. Any person who was willing to participate and was over the age of 18 was allowed to participate in this experiment. The student and faculty populations were comprised of students and faculty at the University of Mary, Fargo, North Dakota. The general population included any other persons who were willing to participate in this study either as a Call Initiator or as a Call Receiver.

A blind study was conducted, in which, neither the Call Initiators nor the Call Receivers knew whether a given call was an analog or a VoIP-initiated call. Only the researcher, in order to catalog the types of surveys distributed to the Call Initiators and Call Receivers, knew the type of call placed. The surveys did not indicate whether they were intended for a VoIP or analog-initiated call. The calls were randomized between VoIP-initiated and analog-initiated calls by placing 56 slips of paper in a non-transparent container, 28 of which were labeled with a -1 (indicating an analog-VoIP call order), and 28 of which were labeled with a +1 (indicating a VoIP-analog call order). The Call Initiator drew the slip of paper from a non-transparent container and handed the slip of paper immediately to the researcher without knowing whether it was a VoIP-initiated or analog-initiated call.

Call Initiators and Call Receivers each completed four surveys. The four surveys are located in Appendix A1. The Pretest Survey was designed to measure the Call Initiators' and Call Receivers' prior beliefs regarding VoIP Call Quality before actually using the VoIP product. The Pretest Survey had one additional question, which was designed to record whether or not the Call Initiators and Call Receivers had ever used VoIP service.

The next two surveys, Survey One and Survey Two, were designed to measure the Relative Perceived Call Quality of VoIP calls and analog landline-based calls. Both Call Initiators and Call Receivers completed each of these surveys after each call was placed and finished. In the case of VoIP calls, a measurement was conducted to record the transmission speed of the broadband connection just prior to the Call Initiators' use of the VoIP product. In this study a DSL connection was used, which was assigned strict controls to avoid any major speed fluctuations. The same DSL service was utilized for all Call Initiators in the study. Immediately after the broadband connection speed was measured, the Call Initiators placed a call using the VoIP phone to the same number used in the analog-initiated call. Finally, the Comparison Survey was designed to measure the Call Initiators' and Call Receivers' perceptions of how the two calls compared. Call Receivers simultaneously completed surveys identical to those completed by the Call Initiators, in the order previously described.

The surveys were comprised of five questions designed to help the Call Initiators and Call Receivers rate their perceptions of the overall Call Quality for each survey. The Call Initiators and Call Receivers were asked to rate each component of Perceived Call Quality on a scale of one to five. After which, they decided based on the same numeric value scale, the overall Perceived Call Quality for the telecommunications medium.

Design and Analysis

The experiment was comprised of 56 "call round" blocks, each of which consisted of a pair of analog and VoIP calls placed in random order. In 28 of the call round blocks, an analog-VoIP call order was used. A VoIP-analog call order was used in the remaining 28 call round blocks. Each participant was randomly assigned to one of the call round blocks. The experimental design was based on eight independent variables. X_{TSU}, X_{TSD}, X_{VEI}, X_{VER}, X_{CR}, X_{CT}, X_{VCQBI}, and X_{VCQBR}, and six dependent variables: Y_{ACQI}, Y_{ACQR}, Y_{VCQI}, Y_{VCQR}, Y_{PCCQVA}, and Y_{PCCQAV}. The variables are defined as follows:

- Transmission Speed Up (kilobits per second, Kbps) (X_{TSU})
- Transmission Speed Down (kilobits per second, Kbps) (X_{TSD})
- Previous VoIP Experience Initiator (X_{VEI})
- Previous VoIP Experience Receiver (X_{VER})
- Call Round Block (X_{CR})
- Call Type (X_{CT})
- Call Initiators' Initial VoIP Call Quality Beliefs (X_{VCQBI})
- Call Receivers' Initial VoIP Call Quality Beliefs (X_{VCQBR})

- Call Initiators' Analog Call Quality Perception (Y_{ACQI})
- Call Receivers' Analog Call Quality Perception (Y_{ACQR})
- Call Initiators' Final VoIP Call Quality Perception (Y_{VCQI})
- Call Receivers' Final VoIP Call Quality Perception (Y_{VCQR})
- VoIP Versus Analog Relative Perceived Comparative Call Quality (Y_{PCCQVA})
- Analog Versus VoIP Perceived Comparative Call Quality (Y_{PCCQAV})

Brief Description of Variables

Independent Variables

Transmission Speed Up

Transmission Speed Up (X_{TSU}) was analyzed before the VoIP product was used, to determine its association with the Relative Perceived Call Quality within the predefined ranges. This test measured the speed in Kbps for the Call Initiator. It is important to test the upload speed of the broadband connection because VoIP calls are actually transmitted using the upload transmission from the Call Initiator. The values for Transmission Speed Up are listed in Table 1.

Transmission Speed Down

Transmission Speed Down (X_{TSD}) was analyzed before the VoIP product was used, to determine its association with the Relative Perceived Call Quality within the predefined ranges. This test measured the speed in Kbps for the Call Initiator. It is important to test the download speed of the broadband connection because the VoIP call is actually received using the download transmission by the Call Initiator. The values for Transmission Speed Down are listed in Table 1.

Previous VoIP Experience Initiator

Previous VoIP Experience Initiator (X_{VEI}) was indicated in the Pretest Survey by the Call Initiator. The Call Initiator circled either 'yes' or 'no,' indicating whether the Call Initiator had any previous experience using VoIP services.

Previous VoIP Experience Receiver

Previous VoIP Experience Receiver (X_{VER}) was indicated in the Pretest Survey by the Call Receiver. The Call Receiver circled either 'yes' or 'no,' indicating whether the Call Receiver had any previous experience using VoIP services.

Call Round Blocks

Call Round Blocks (X_{CR}) were used for the analog and VoIP-initiated calls for both the Call Initiator and Call Receiver. The experiment was comprised of 56 Call Round Blocks, each of which consisted of a pair of analog and VoIP calls placed in random order. In 28 of the Call Round Blocks an analog-VoIP call order was used. A VoIP-analog call order was used in the remaining 28 Call Round Blocks. Each participant was randomly assigned to one of the Call Round Blocks. Call Round Blocks were used to address variations among calls made at different times of the day and any variations in equipment response times.

Call Type

Call Type (X_{CT}) was used within the Call Round Blocks variable. The Call Type variable indicated, for a given call round block, whether a VoIP or analog call was initiated; this value was known only to the researcher. The Call Type Variable had two values: a -1 indicated an analog-initiated call, and a +1 indicated a VoIP-initiated call. The Call Type values were derived by placing 56 slips of paper in a non-transparent container, 28 of which were labeled with a -1 (indicating an analog-VoIP call order), and 28 of which were labeled with a +1 (indicating a VoIP-analog call order).

Call Initiators' Initial VoIP Call Quality Beliefs

Call Initiators' Initial VoIP Call Quality Beliefs (X_{VCQBI}) was used to measure the Call Initiators' initial perceptions of how the VoIP product performed in a variety of ways. The Call Initiators completed five specific questions and indicated a "Final Call Quality Rating" at the end of the Pretest Survey, which is located in Appendix A1. The values for Call Initiators' Initial VoIP Call Quality Beliefs are listed in Table 1. The Pretest Survey had one additional question prompting the Call Initiator to indicate a 'yes' or 'no' response to determine if the Call Initiators had ever used VoIP services before. The Pretest Survey was completed by the Call Initiators before the VoIP call was placed.

Call Receivers' Initial VoIP Call Quality Beliefs

Call Receivers' Initial VoIP Call Quality Beliefs (X_{VCQBR}) was used to measure the Call Receivers' initial perceptions of how the VoIP product performed in a variety of ways. The Call Receivers completed five specific questions and indicated a "Final Call Quality Rating" at the end of the Pretest Survey. The values for Call Receivers' Initial VoIP Call Quality Beliefs are listed in Table 1. The Pretest Survey had one additional question prompting the Call Receiver to indicate a 'yes' or 'no' response to determine if the Call Receivers had ever used VoIP services before. The Pretest Survey was completed by the Call Receivers before the VoIP call was placed.

Dependent Variables

Call Initiators' Analog Call Quality Perception

Call Initiators' Analog Call Quality Perception (Y_{ACQI}) was used to measure the Call Initiators' perception of how the analog call performed in a variety of ways. The Call Initiators completed five specific questions and indicated a "Final Call Quality Rating" at the end of Survey Two, which is located in Appendix A1. The values for Call Initiators' Analog Call Quality Perceptions are listed in Table 1. The Call Initiators completed Survey Two after the analog call was completed.

Call Receivers' Analog Call Quality Perception

Call Receivers' Analog Call Quality Perception (Y_{ACQR}) was used to measure the Call Receivers' perception of how the analog call performed in a variety of ways. The Call Receivers completed five specific questions and indicated a "Final Call Quality Rating" at the end of Survey Two. The values for Call Receivers' Analog Call Quality Perceptions are listed in Table 1. The Call Receivers completed Survey Two after the analog call was completed.

Call Initiators' Final VoIP Call Quality Perception

Call Initiators' Final VoIP Call Quality Perception (Y_{VCQI}) was used to measure the Call Initiators' perception of how the VoIP-initiated call performed in a variety of ways. The Call Initiators completed five specific questions and indicated a "Final Call Quality Rating" at the end of Survey One, which is located in Appendix A1. The values for Call Initiators' Final VoIP Call Quality Perceptions are listed in Table 1. The Call Initiators completed Survey One after the VoIP call was completed. The "Final Call Quality Rating" as indicated by the Call Initiators was used to measure the linear main effects of

Transmission Speed Down (X_{TSD}) and Previous VoIP Experience Initiator (X_{VEI}), the potential two-factor linear by linear interactions of X_{TSD} and X_{VEI}, and the quadratic main effects of Call Initiators' Initial VoIP Call Quality Beliefs (X^2_{VCQBI}) in the multiple forward stepwise regression analysis.

Call Receivers' Final VoIP Call Quality Perception

Call Receivers' Final VoIP Call Quality Perception (Y_{VCQR}) was used to measure the Call Receivers' perception of how the VoIP received call performed in a variety of ways. The Call Receivers completed five specific questions and indicated a "Final Call Quality Rating" at the end of Survey One. The values for Call Receivers' Final Call Quality Perceptions are listed in Table 1. The Call Receivers completed Survey One after the VoIP call was completed. The "Final Call Quality Rating" as indicated by the Call Receivers was used to measure the linear main effects of Transmission Speed Up (X_{TSU}) and Previous VoIP Experience Receiver (X_{VER}), the potential two-factor linear by linear interactions of X_{TSU} and X_{VER}, and the quadratic main effects of Call Receivers' Initial VoIP Call Quality Beliefs (X^2_{VCQBR}) in the multiple forward stepwise regression analysis.

VoIP Versus Analog Perceived Comparative Call Quality

VoIP Versus Analog Perceived Comparative Call Quality (Y_{PCCQVA}) was defined as the Call Initiators' and Call Receivers' direct comparisons of the VoIP calls to the analog calls. This measure was determined after all other surveys and calls were completed. The Call Initiators and Call Receivers completed five specific questions and indicated a "Final Comparative Call Quality Rating" at the end of the Comparative Survey, which is located in Appendix A1. The values for VoIP Versus Analog Perceived Comparative Call Quality are listed in Table 1. The "Final Call Quality Rating" as indicated by the VoIP Versus

Analog Perceived Comparative Call Quality was used to measure the linear main effects of 1) Transmission Speed Down (X_{TSD}), 2) Transmission Speed Up (X_{TSU}), 3) Previous VoIP Experience Initiator (X_{VEI}), 4) Previous VoIP Experience Receiver (X_{VER}), and 5) Call Type (X_{CT}). The potential two-factor linear by linear interactions of 1) X_{TSD} and $X_{CT,}$ 2) X_{VEI} and $X_{CT,}$ 3) X_{TSD} and $X_{VEI,}$ 4) X_{TSU} and $X_{CT,}$ 5) X_{VER} and $X_{CT,}$ and 6) X_{TSU} and $X_{VER.}$ The two quadratic main effects measured were Call Initiators' Initial VoIP Call Quality Beliefs (X^2_{VCQBI}) and Call Receivers' Initial VoIP Call Quality Beliefs (X^2_{VCQBR}). The listed variables were measured in the multiple forward stepwise regression analysis for VoIP Versus Analog Perceived Comparative Call Quality.

Analog Versus VoIP Perceived Comparative Call Quality

Analog Versus VoIP Perceived Comparative Call Quality (Y_{PCCQAV}) was defined as the Call Initiators' and Call Receivers' direct comparisons of the analog calls to the VoIP calls. This measure was determined after all other surveys and calls were completed. The Call Initiators and Call Receivers completed five specific questions and indicated a "Final Comparative Call Quality Rating" at the end of the Comparative Survey, which is located in Appendix A1. The values for the Analog Versus VoIP Perceived Comparative Call Quality are listed in Table 1. The "Final Call Quality Rating" as indicated by the Analog Versus VoIP Perceived Comparative Call Quality was used to measure the linear main effects of 1)Transmission Speed Down (X_{TSD}), 2) Transmission Speed Up (X_{TSU}), 3) Previous VoIP Experience Initiator (X_{VEI}), 4) Previous VoIP Experience Receiver (X_{VER}), and 5) Call Type (X_{CT}). The potential two-factor linear by linear interactions of 1) X_{TSD} and $X_{CT,}$ 2) X_{VEI} and $X_{CT,}$ 3) X_{TSD} and $X_{VEI,}$ 4) X_{TSU} and $X_{CT,}$ 5) X_{VER} and $X_{CT,}$ and 6) X_{TSU} and $X_{VER.}$ The two quadratic main effects measured were Call Initiators' Initial VoIP Call

Quality Beliefs (X^2_{VCQBI}) and Call Receivers' Initial VoIP Call Quality Beliefs (X^2_{VCQBR}). The listed variables were measured in the multiple forward stepwise regression analysis for Analog Versus VoIP Perceived Comparative Call Quality.

With VoIP, much of the PTN can be bypassed. However, the purpose of this research was to determine perceptions of how Call Quality compared between analog and VoIP calls on a DSL network. This study concentrated on a field experiment to test if VoIP Call Quality is much worse, worse, the same, better, or much better than that of analog. The findings of this field study may help determine whether VoIP companies can compete, in terms of delivered call quality, with the LECs in telecommunications.

Table 1

Construct Variables

Variable	Levels	Values	Call initiators' and call receivers' ratings
Transmission speed up (X_{TSU})	2	-1 (128 Kbps – 220 Kbps) +1 (221 Kbps – 512 Kbps)	N/A N/A
Transmission speed down (X_{TSD})	2	-1 (128 Kbps – 220 Kbps) +1 (221 Kbps – 512Kbps)	N/A N/A
Previous VoIP experience initiator (X_{VEI})	2	-1 (No) +1 (Yes)	Yes / No
Previous VoIP experience receiver (X_{VER})	2	-1 (No) +1 (Yes)	Yes / No
Call round block (X_{CR})	56	1-56	N/A
Call type (X_{CT})	2	-1 (Analog) +1 (VoIP)	N/A
Call initiators' initial VoIP call quality beliefs (X_{VCQBI})	5	Unbearable, below average, average, above average, excellent	Scaled 1-5
Call receivers' initial VoIP call quality beliefs (X_{VCQBR})	5	Unbearable, below average, average, above average, excellent	Scaled 1-5
Call initiators' analog call quality perception (Y_{ACQI})	5	Unbearable, below average, average, above average, excellent	Scaled 1–5
Call receivers' analog call quality perception (Y_{ACQR})	5	Unbearable, below average, average, above average, excellent	Scaled 1–5
Call initiators' final VoIP call quality perception (Y_{VCQI})	5	Unbearable, below average, average, above average, excellent	Scaled 1–5
Call receivers' final VoIP call quality perception (Y_{VCQR})	5	Unbearable, below average, average, above average, excellent	Scaled 1–5
VoIP versus analog perceived comparative call quality (Y_{PCCQVA})	5	Much worse, worse, the same, better, much better	Scaled 1-5
Analog versus VoIP perceived comparative call quality (Y_{PCCQAV})	5	Much worse, worse, the same, better, much better	Scaled 1–5

Possible Outcomes of the Experiment Design Analysis

A multiple forward stepwise regression analysis was conducted on the data collected from the user surveys contained in Appendix A1. The analysis yielded modeling equations for all the dependent variables. Call Initiators and Call Receivers each completed four surveys which were used to compute the values for the following six dependent variables: Y_{ACQI}, Y_{ACQR}, Y_{VCQI}, Y_{VCQR}, Y_{PCCQVA}, and Y_{PCCQAV}.

These six variables each represented a particular measure as indicated by the Call Initiators' and Call Receivers' survey responses, shown in the "Final Call Quality Rating." Y_{ACQI} represents the Call Initiators' Analog Call Quality Perception, taken from the "Final Call Quality Rating" provided by the Call Initiators in Survey Two. Y_{ACQR} represents the Call Receivers' Analog Call Quality Perception, taken from the "Final Call Quality Rating" provided by the Call Receivers in Survey Two. Y_{VCQI} represents the Call Initiators' Final VoIP Call Quality Perception, taken from the "Final Call Quality Rating" provided by the Call Initiators in Survey One. Y_{VCQR} represents the Call Receivers' Final VoIP Call Quality Perception, taken from the "Final Call Quality Rating" provided by the Call Receivers in Survey One. Y_{PCCQVA} represents VoIP Versus Analog Relative Perceived Comparative Call Quality, taken from the "Final Call Quality Rating" provided by the Call Initiators and Call Receivers in the Comparative Survey. Y_{PCCQAV} represents Analog Versus VoIP Relative Perceived Comparative Call Quality, taken from the "Final Call Quality Rating" provided by the Call Receivers and Call Initiators in the Comparative Survey.

The response data derived from these surveys yields a possible range of values from 1 to 5. Their corresponding levels are indicated below in Table 2. Since the Call Initiators and Call Receivers are predefined and were independent groups, orthogonal coding was used to analyze the data derived from the Call Initiators and Call Receivers, as indicated in the "Final Call Quality Ratings" of each of the four surveys. SPSS® was used to analyze the data from the survey responses and perform the multiple forward stepwise regression modeling for the construct variables (SPSS, 2001). The significance levels used to qualify a term for entry were .01, .05, and .10, which were also used as the criteria for including any of the possible linear main effects, linear by linear two-factor interactions, as well as quadratic main effects. The pairs of variables that were analyzed for potential linear by linear two-factor interactions with each other were: 1) X_{TSD} and X_{CT}, 2) X_{VEI} and X_{CT}, 3) X_{TSD} and X_{VEI}, 4) X_{TSU} and X_{CT}, 5) X_{VER} and X_{CT}, 6) X_{TSU} and X_{VER}. The quadratic main effects that were analyzed were X^2_{VCQBI} and X^2_{VCQBR}.

Table 2

Relative Perceived Call Quality Levels

Value	Rating
Much worse	1
Worse	2
The same	3
Better	4
Much better	5

After the analysis was completed, the following questions were answered:

- How does the perceived quality of VoIP calls compare to that of traditional landline-based communications in the perceptions of DSL broadband users?
- Are the Call Initiators' and Call Receivers' Initial Beliefs with regard to the Call Quality of the VoIP product correlated with their actual posttest Relative Perceived Call Quality ratings of VoIP?

- Are differences in transmission speeds correlated with the perceived quality of the VoIP calls?

Limitations of the Study

The limitations of this study were the participant sample size and the type of broadband connection used. The broadband connection was confined to a DSL connection on a controlled network. This was due solely to technological limitations and resources. The experiment was done at one geographical location, the University of Mary, Fargo, North Dakota, where all calls were initiated and received in two separate rooms. The latter restriction was required to ensure that all Call Initiators used the same analog and DSL-enabled VoIP lines throughout the experiment. The experiment was designed to simulate a real-world experiment with a VoIP adapter, DSL connection, along with the analog phone lines and phones. The participant sample was comprised of individuals randomly contacted and selected from college students, college faculty, and persons not connected with the campus. Call Initiators and Call Receivers who were not affiliated with the campus were defined as the "general population." The general population participated in the study in the same manner as the college students and college faculty. Any person who was willing to participate and was over the age of 18 was able to participate in this experiment. The majority of the participants were between the ages of 30 and 45. The results of this study may have varied with different population samples. However, this research was designed and conducted to ensure that it still met the basic technological and statistical design criteria for testing the Call Quality of VoIP services on a DSL-enabled connection.

Statistical Design and Analyses

Implementing and analyzing the data from the experiment design for this study generated data and results from a group of 112 randomly chosen persons who agreed to participate. All participants were restricted to campus while engaged in the experiment. The statistical methodology employed multiple forward stepwise regression modeling for estimating the linear main effects, as well as all potential two-factor interactions, and quadratic main effects. All potential interaction terms to be included in the regression model are summarized in the section entitled "Possible Outcomes of Variable Design."

Estimated Power Analysis

To verify that the sample size was appropriate for this experiment, an estimate of its power was calculated.

The statistical analyses for this experiment were applied to data obtained from a group of 56 randomly sampled Call Initiators and 56 randomly sampled Call Receivers. Though the data was not analyzed using a t-test, if it were, the sample size of 56 at a significance level of .01 yielded a power value of 85% using a two-tailed test (Lenth, 2004). To achieve a power of 85% a sample size of 56 Call Initiators and 56 Call Receivers were used for this experiment.

Instrumentation

The instrumentation used in this experiment was identical for all users. Quality of Service (QoS) standards were implemented to ensure uniformity for all the technological components of the experiment, and the same equipment was used throughout the experiment.

The research data was obtained via four survey instruments. The format of the survey questions was multiple choice, and each Call Initiator and Call Receiver was given identical surveys. The surveys did not disclose whether analog or VoIP Call Quality was being measured. The preliminary questions in each survey helped the Call Initiators and Call Receivers identify and evaluate several aspects of Call Quality. These questions were followed by a single question at the end of each survey asking the Call Initiators and Call Receivers to provide an overall rating of Perceived Call Quality. Copies of all the surveys are attached in Appendix A1.

The technological instrumentation used in this research consisted of two hard-wired analog phones, a VoIP adapter, and a DSL connection. The same hardwired analog phone was used by the Call Initiators to make a call over the public telephone network (PTN), and the VoIP adapter was used by the Call Initiators to make a VoIP-enabled call. The DSL connection was the transmission medium over which the VoIP call was placed. The second hard-wired analog phone consisted of the same make and model as used by the Call Initiators. The second hard-wired analog phone is the phone the Call Receivers used. Both phones utilized for this experiment were AT&T 210 analog phones.

Reliability of Instrumentation

The reliability of the technological instrumentation was tested before Call Initiators made the analog and VoIP calls. The analog phone lines were checked for proper decibel levels in milliwatts (dBm). This was to ensure that the proper specifications were met for acceptable line conditions. The minimum standard for loss requires that the decibel level be less than 8.5 dBm (Borowski, 2004). Loss is defined as the acceptable dBm level where certain thresholds are obtained to achieve proper volume levels. The Call Receivers' phone line tested at 3.7 dBm, while the Call Initiators' phone line tested at 7.7 dBm.

The VoIP adapter allowed the call to be switched over the VoIP network. The medium used to gain access to the VoIP network was a DSL broadband connection. To ensure the reliability of the DSL connection, a speed test was conducted to measure and document the speed of the DSL connection before the Call Initiators placed a call on the VoIP network. Both upload and download speeds were checked. Strict controls were placed on the DSL connection to ensure there were no major speed fluctuations. To ensure consistent speed measurements, the DSL connection was not connected to any local area network (LAN).

Ethical Assurances

The research in this experiment was centered on the completion of four separate surveys by 56 randomly selected Call Initiators and 56 randomly selected Call Receivers. Call Initiators and Call Receivers participated in the field experiment as indicated in the methodology section. All initially-identified potential Call Initiators and Call Receivers were asked to review and provide informed consent forms. (A blank informed consent form is located in Appendix B2). The potential participants were informed of their right to elect to participate in this study and of their potential roles and responsibilities.

The participants in this study, along with their individual survey responses, were confidential. The study was held at the University of Mary, Fargo, North Dakota. Every effort was made to comply with the standards for conducting research with human participants.

Reliability and Validity

The reliability of the field experiment was dependent on its internal consistency, as the experiment was conducted at certain points in time. It was the intent of this research to produce a field experiment that can easily be replicated by consumers and businesses. The reliability was guaranteed by virtue of its construct validity, which is defined as "the degree to which a measurement device accurately measures the theoretical construct it is designed to measure" (Cozby, 2004, p. 369). Multiple forward stepwise regression modeling was used, as there were 14 variables, as indicated in Table 1.

A possible threat to internal validity was a regression threat. Regression threat is defined as "a statistical phenomenon that causes a group's average performance on one measure to regress toward or appear closer to the mean of that measure than anticipated or predicted. A regression threat will bias estimates of the group's posttest performance and can lead to incorrect causal inferences" (Trochim, 2001, p. 351). The pretest is based solely on a hypothesis. The null hypothesis was the perceived quality of VoIP calls were worse than the quality of analog calls, while the alternative hypothesis was the quality of VoIP calls were better than the perceived quality of analog calls. The potential multiple regression threat was mitigated by performing a regression analysis with randomizing both call round blocks and/call type ordering within the call blocks to mitigate the threat of call times and equipment variances. The pretest served as a measurement criterion against which to base the experiment. However, the data was correlated with the posttest study.

The 56 chosen Call Initiators and 56 chosen Call Receivers constituted the sampling frame. The members of the study participated voluntarily and were not pre-screened. The first 112 individuals who agreed to participate became the participants in this study.

External validity was assured because the field experiment was designed to be accurate and reproducible. However, with this experiment the external validity may have been compromised because the population sample was comprised of 112 participants. A controlled environment was used as the Call Initiators and Call Receivers were placed in two separate rooms when placing and receiving calls. While placing and receiving calls, the Call Initiators and Call Receivers could only hear each other during the duration of the calls. The experiment was controlled for reproducibility by having the participants fill out identical surveys for both the pretest and posttest and by using accurate, proven resources to measure the transmission speed of the broadband connection. A telecommunications field technician checked both phones' lines prior to the experiment, to ensure certain thresholds were achieved as described in the "Reliability of Instrumentation" section.

Chapter 4: Findings

Overview

The findings section contains the data derived from the experiment, along with the multiple forward stepwise regression results for both Call Initiators and Call Receivers. Tables 3 through 10 contain the data collected from the survey results for both Call Initiators and Call Receivers. Table 11 is a regression coefficient table used to identify the coefficients in the four cases that were analyzed. These four cases were analyzed:

1. Call Initiators' Final VoIP Call Quality Perception (Y_{VCQI})
2. Call Receivers' Final VoIP Call Quality Perception (Y_{VCQR})
3. VoIP Versus Analog Relative Perceived Comparative Call Quality (Y_{PCCQVA})
4. Analog Versus VoIP Relative Perceived Comparative Call Quality (Y_{PCCQAV})

The analysis and evaluation of findings section contains a complete explanation for each case that was significant. This includes linear main effects, two-factor potential interactions, and quadratic main effects. The coefficients are given for each case that produced a significant (for a chosen significance level) regression. Along with multiple forward stepwise regression, a chi-square test was also employed. Table 12 is a chi-square table for Analog Versus VoIP Relative Perceived Comparative Call Quality (Y_{PCCQAV}) and VoIP Versus Analog Relative Perceived Comparative Call Quality (Y_{PCCQVA}). Table 13 is a chi-square table for the Participants' Pretest and Posttest VoIP Surveys. A paired *t* test is used to further explain Table 13.

The summary is a condensed version of the entire chapter. The summary includes the main points from the findings, along with the analysis and evaluation of findings section.

Findings

The experiment revealed that 82% of the Call Initiators and 75% of the Call Receivers had not previously used VoIP service.

Table 3 indicates the results from the Pretest Survey completed by the Call Initiators. This survey was completed before the VoIP call was placed.

Table 3

Call Initiators' Initial VoIP Call Quality Beliefs

Call initiators' ratings	Frequency	Percent
Unbearable	0	0
Below average	4	7.1
Average	33	58.9
Above average	12	21.4
Excellent	7	12.5
Participants	56	100.0

Table 4 indicates the results from the Pretest Survey completed by the Call Receivers. This survey was completed before the VoIP call was received.

Table 4

Call Receivers' Initial VoIP Call Quality Beliefs

Call receivers' ratings	Frequency	Percent
Unbearable	1	1.8
Below average	12	21.4
Average	31	55.4
Above average	10	17.9
Excellent	2	3.6
Participants	56	100.0

Table 5 indicates the results from Survey One, which was a blind survey designated as the VoIP survey completed by the Call Initiators after the VoIP call was placed.

Table 5

Call Initiators' Final VoIP Call Quality Perception

Call initiators' ratings	Frequency	Percent
Unbearable	0	0
Below average	10	17.9
Average	16	28.6
Above average	22	39.3
Excellent	8	14.3
Participants	56	100.0

Table 6 indicates the results from Survey One, which was a blind survey designated as the VoIP survey completed by the Call Receivers after the VoIP call was received.

Table 6

Call Receivers' Final VoIP Call Quality Perception

Call receivers' ratings	Frequency	Percent
Unbearable	0	0
Below average	12	21.4
Average	28	50
Above average	12	21.4
Excellent	4	7.1
Participants	56	100.0

Table 7 indicates the results from Survey Two, which was a blind survey designated as the analog survey completed by the Call Initiators after the analog call was placed.

Table 7

Call Initiators' Analog Call Quality Perception

Call initiators' ratings	Frequency	Percent
Unbearable	0	0
Below average	2	3.6
Average	29	51.8
Above average	18	32.1
Excellent	7	12.5
Participants	56	100.0

Table 8 indicates the results from Survey Two, which was a blind survey designated as the analog survey completed by the Call Receivers after the analog call was received.

Table 8

Call Receivers' Analog Call Quality Perception

Call receivers' ratings	Frequency	Percent
Unbearable	1	1.8
Below average	9	16.1
Average	22	39.3
Above average	19	33.9
Excellent	5	8.9
Participants	56	100.0

Table 9 indicates the results from the VoIP Versus Analog Comparative Survey, which was completed by the Call Initiators and Call Receivers after the completion of both VoIP and analog calls.

Table 9

VoIP Versus Analog Relative Perceived Comparative Call Quality

Participants' Ratings	Frequency	Percent
Much worse	3	5.4
Worse	14	25.0
The same	20	35.7
Better	11	19.6
Much better	8	14.3
Participants	56	100.0

Table 10 indicates the results from the Analog Versus VoIP Comparative Survey, which was completed by the Call Initiators and Call Receivers after the completion of both VoIP and analog calls.

Table 10

Analog Versus VoIP Relative Perceived Comparative Call Quality

Participants' ratings	Frequency	Percent
Much worse	1	1.8
Worse	14	25.0
The same	27	48.2
Better	11	19.6
Much better	3	5.4
Participants	56	100.0

A multiple forward stepwise regression analysis was conducted at the .01, .05, and .10 significance levels to evaluate if, and if so, how the dependent variables correlated with the independent variables. In all regression cases, linear main effects, linear by linear two-factor interactions, and quadratic main effects were tested. Only the highest significance level that yielded a regression coefficient are shown in Table 11.

Table 11 summarizes the results of the regression analysis conducted at .01, .05, and .10 significance levels for the following four cases:

1. Call Initiators' Final VoIP Call Quality Perception (Y_{VCQI})
2. Call Receivers' Final VoIP Call Quality Perception (Y_{VCQR})
3. VoIP Versus Analog Relative Perceived Comparative Call Quality (Y_{PCCQVA})
4. Analog Versus VoIP Relative Perceived Comparative Call Quality (Y_{PCCQAV})

Table 11 interaction terms:

1. X_{TSDCTI} is the linear by linear two-factor interaction of Transmission Speed Down (X_{TSD}) and Call Type (X_{CT}) for the Call Initiators.
2. X_{VEICTI} is the linear by linear two-factor interaction of Previous VoIP Experience Initiator (X_{VEI}) and Call Type (X_{CT}) for the Call Initiators.
3. X_{TSUCTR} is the linear by linear two-factor interaction of Transmission Speed Up (X_{TSU}) and Call Type (X_{CT}) for the Call Receivers.
4. X_{VERCTR} is the linear by linear two-factor interaction of Previous VoIP Experience Receiver (X_{VER}) and Call Type (X_{CT}) for the Call Receivers.

5. $X_{VEITSDI}$ is the linear by linear two-factor interaction of Previous VoIP Experience Initiator (X_{VEI}) and Transmission Speed Down (X_{TSD}) for the Call Initiators.

6. $X_{VERTSUR}$ is the linear by linear two-factor interaction of Previous VoIP Experience Receiver (X_{VER}) and Transmission Speed Up (X_{TSU}) for the Call Receivers.

Table 11

Regression Coefficients

	Sig. level	X_{VEI}	X_{VER}	X_{TSU}	X_{TSD}	X_{CT}	X_{TSDCTI}	X_{VEICTI}	X_{TSUCTR}	X_{VERCTR}	$X_{VEITSDI}$	$X_{VERTSUR}$	X^2_{VCQBI}	X^2_{VCQBR}	Constant
Y_{VCQI}	.10	**NS**[1]	**NA**[2]	**NA**[2]	**NS**[1]	**NA**[2]	**NA**[2]	**NA**[2]	**NA**[2]	**NA**[2]	**NS**[1]	**NA**[2]	**NS**[1]	**NA**[2]	
Y_{VCQR}	.10	**NA**[2]	**NS**[1]	**NS**[1]	**NA**[2]	**NA**[2]	**NA**[2]	**NA**[2]	**NA**[2]	**NA**[2]	**NA**[2]	**NS**[1]	**NA**[2]	**NS**[1]	
Y_{PCCQVA}	.05	**NS**[1]	**NS**[1]	**NS**[1]	**NS**[1]	**NS**[1]	**NS**[1]	**NS**[1]	**NS**[1]	**NS**[1]	**NS**[1]	**NS**[1]	**NS**[1]	**-.432**[3]	**-.300**[3]
Y_{PCCQAV}	.05	**NS**[1]	**NS**[1]	**NS**[1]	**NS**[1]	**NS**[1]	**NS**[1]	**NS**[1]	**NS**[1]	**NS**[1]	**NS**[1]	**.488**[3]	**NS**[1]	**NS**[1]	**-.138**[3]

[1] NS denotes the variable was entered in the regression analysis. However, at the given significance level, no regression coefficients were statistically significant in the analysis.
[2] NA denotes the variable was not entered in the regression analysis.
[3] Denotes the coefficient used in the corresponding regression equation is found under the section entitled "Analysis and Evaluation of Findings" section for each case.

Table 12 summarizes the results for the chi-square test for:

1. Analog Versus VoIP Relative Perceived Comparative Call Quality (Y_{PCCQAV})
2. VoIP Versus Analog Relative Perceived Comparative Call Quality (Y_{PCCQVA})

Table 12

Chi-Square Test Results for the Comparative Surveys

	Both groups expected N[1]	Analog versus VoIP call observed N[1]	VoIP versus analog call observed N[1]
Much worse	11.2	1	3
Worse	11.2	14	14
The same	11.2	27	20
Better	11.2	11	11
Much better	11.2	3	8
Chi-square		38.286	14.536
Degrees of freedom		4	4
Significance		p. < .01	p. < .01

[1] N represents the number of participants that, under the null hypotheses of equally likely ratings, either expected to or actually did assign a given rating.

Table 13 summarizes the results for the chi-square test for the Participants' Pretest and Posttest VoIP Surveys.

Table 13

Chi-Square Test Results for Participants' Pretest and Posttest VoIP Surveys

	Pretest survey expected N[1]	Pretest survey observed N[1]	Final VoIP calls expected N[1]	Final VoIP call observed N[1]
Unbearable	22.4	1		
Below average	22.4	16	28	22
Average	22.4	64	28	44
Above average	22.4	22	28	34
Excellent	22.4	9	28	12
Chi-square		107.554		20.857
Degrees of freedom		4		3
Significance		p. < .01		p. < .01

Analysis and Evaluations of Findings

Four cases were analyzed using a multiple forward stepwise regression, and in each case the .01, .05, and .10 significance levels were used. The resulting regression coefficients are listed in Table 11. In each case, the highest significance level that yielded a regression coefficient was listed in Table 11 and is discussed below. The chi-square test

[1] N represents the number of participants that, under the null hypotheses of equally likely ratings, either expected to or actually did assign a given rating.

results were given for the Comparative Surveys (Table 12), as well as the Pretest and Posttest VoIP Surveys (Table 13). The results are discussed in the section entitled "Chi-Square Test Results". The section entitled "Paired *t* Test Results" explains any differences between the Pretest and Posttest VoIP Surveys.

Y_{PCCQVA} Case

In the case of VoIP Versus Analog Relative Perceived Comparative Call Quality (Y_{PCCQVA}), the quadratic main effects of Call Receivers' Initial VoIP Call Quality Beliefs (X^2_{VCQBR}) were statistically significant at the .05 level. Over 14% of the variation of VoIP Versus Analog Relative Perceived Comparative Call Quality can be explained by the quadratic main effects of the Pretest Survey results from the Call Receivers. $R^2 = .143$, $t(26) = -2.082$, $F(1, 26) = 4.336$, $p = .047$. The regression equation is $Y_{PCCQVA} = -.432X^2_{VCQBR} - .300$. Based on the model, the posttest results for VoIP Versus Analog Relative Perceived Comparative Call Quality was about 43% lower than the Initial Pretest Beliefs for the Call Receivers. This makes sense because VoIP deficiencies will often be more noticeable to the Call Receiver than the Call Initiator. This happens because Call Receivers can sometimes detect a slight delay in the conversation. The low R^2 value of 14% implies that the Pretest Survey from the Call Receivers was not a good predictor for VoIP Versus Analog Relative Perceived Comparative Call Quality.

Y_{PCCQAV} Case

In the case of Analog Versus VoIP Relative Perceived Comparative Call Quality (Y_{PCCQAV}), the interaction between Previous VoIP Experience Receiver (X_{VER}) and Transmission Speed Up (X_{TSU}) resulting in the interaction term of $X_{VERTSUR}$ being statistically significant at the .05 level. Over 22% of the variation of Analog Versus VoIP Relative Perceived Comparative Call Quality can be explained by the linear by linear interaction of Previous VoIP Experience Receiver and Transmission Speed Up. $R^2 = .228$, $t(26) = 2.769$, $F(1, 26) = 7.665$, $p = .01$. The regression equation is $Y_{PCCQAV} = .488X_{VERTSUR} -.138$.

The interaction term between Previous VoIP Experience Receiver and Transmission Speed Up resulted in a perceived increase of 48% for Analog Versus VoIP Relative Perceived Comparative Call Quality. This can be interpreted to mean that when the interaction between Previous VoIP Experience Receiver and Transmission Speed Up strengthens, the analog scores will increase. An increased analog score means that more participants would favor the analog call over the VoIP call. Obviously this does not make sense, because the interaction between Previous VoIP Experience Receiver and Transmission Speed Up would realistically do nothing to impact an analog call. The low R^2 value of 22% implies that the interaction of Previous VoIP Experience Receiver and Transmission Speed Up was not a good predictor of Analog Versus VoIP Relative Perceived Comparative Call Quality. An interpretation can be made that the participant may have scored the calls differently because they did not know the difference in call types. This proves that the blind study was appropriate for this experiment.

Chi-Square Test Results

The chi-square test results for the case of analog versus VoIP indicated the test was significant, $\chi^2(4, N=56) = 38.286$, $p<.01$, critical value = 13.28, with 4 degrees of freedom. As a result, the test implies that the observed responses from the participants and the expected responses from the participants were not proportionately similar to each other. The blind study along with the randomization of call types may have had an effect on this.

The chi-square test results for the case of VoIP versus analog indicated the test was significant, $\chi^2(4, N=56) = 14.536, p<.01$, critical value = 13.28, with 4 degrees of freedom. As a result, the test implies that the observed responses from the participants and the expected responses from the participants were not proportionately similar to each other. The blind study along with the randomization of call types may have had an effect on this.

The chi-square test results for the Participants' Final VoIP Call Quality Perception indicated the test was significant: $\chi^2(3, N=112) = 20.857, p<.01$, critical value = 11.34, with 3 degrees of freedom. It can be interpreted that the observed responses from the participants and the expected responses from the participants were not proportionately similar to each other. A conclusion can be made that the randomization of the experiment, along with the blind study was effective.

The chi-square test results for the Participants' Pretest Survey indicated the test was significant: $\chi^2(4, N=112) = 107.554, p<.01$, critical value = 13.28, with 4 degrees of freedom. It can be interpreted that the observed responses from the participants and the expected responses from the participants were not proportionately similar to each other. A conclusion can be made that 75% of the Call Receivers and 82% of the Call Initiators had not previously used VoIP services before the experiment.

Paired t Test Results

A paired *t* test was conducted to evaluate if there was a statistically significant difference with the Pretest VoIP Survey and the Posttest VoIP Survey for all participants. The results indicated that the mean for the Posttest VoIP Survey (M = .3214, SD =. 91252) was greater than the mean for the Pretest VoIP Survey (M = .1964, SD = .81472), $t(111)$ = 1.212, p = .228, critical value at infinity = 2.576, p<.01. Since the null hypothesis is that there was no statistical difference between the Pretest and Posttest VoIP Surveys, the critical value of (2.576), obtained value of (1.212), and large p value (.228), allow for the null hypothesis to be accepted. The variation between the two surveys may best be explained by chance and the fact that only 82% of Call Initiators and 75% of Call Receivers had ever used VoIP services before the experiment.

Summary

The experiment revealed that 82% of the Call Initiators and 75% of the Call Receivers had not previously used VoIP service. Tables 3 through 10 summarized the data collected from the surveys completed by both Call Initiators and Call Receivers. A multiple forward stepwise regression analysis was conducted at the .01, .05, and .10 significance levels. The regression coefficients, along with their constants were listed in Table 11. Four cases were analyzed for linear main effects, linear by linear two-factor interactions, and quadratic main effects. Two out of the four cases resulted in statistically significant regressions. The highest significance level for each case was further explained that produced regression, along with the proper regression equation.

In both cases the R^2 value proved that the models did not fit the data well. As a result, there were no large correlations found between the independent variables including linear main effects, linear by linear interactions, or quadratic main effects with their respective dependent variables. According to SPSS, the higher the R^2 value, the stronger the relationship is between the independent and dependent variables (SPSS, 2001). In this analysis, much of the variance between the independent and dependent variables remained unexplained as represented by the low R^2 values. In each of the regression cases the models were further explained at the highest significance level.

Table 12 summarized was the Chi-square test results for Analog Versus VoIP Comparative Call Quality and VoIP Versus Analog Comparative Call Quality. Table 13 summarized the chi-square results for Pretest and Posttest VoIP Surveys. Along with the chi-square results a paired *t* test was conducted to analyze if there were any statistical differences between the Pretest and Posttest VoIP Surveys.

Chapter 5: Summary, Conclusions, and Recommendations

Summary

With VoIP much of the research has been conducted in regards to admission control, delay, delay jitter, packet loss, and standards. There has been very little research done on analyzing how human participants perceive the VoIP call. In this study, human participants were used to make or receive a VoIP and analog call. The study focused on three research questions that were answered and are given in the conclusions section of this chapter. The three research questions follow:

- How does the perceived quality of VoIP calls compare to that of traditional landline-based communications in the perceptions of DSL broadband users?
- Are the Call Initiators' and Call Receivers' Initial Beliefs with regard to the Call Quality of the VoIP product correlated with their actual posttest Relative Perceived Call Quality ratings of VoIP?
- Are differences in transmission speeds correlated with the perceived quality of the VoIP calls?

In this experiment a blind study was conducted using VoIP and analog-initiated calls. The experiment was randomized between VoIP and analog calls. The experiment was set up on a controlled network using a DSL connection, two analog phones, and a VoIP adapter. Transmission speeds were taken for both upload and download speeds prior to each VoIP call.

The data derived from the participants was analyzed using a multiple forward stepwise regression analysis at the .01, .05, and .10 significance levels. The multiple regression analysis analyzed whether there were any statistically significant correlations with the dependent and independent variables. Linear main effects, linear by linear two-factor interactions, and quadratic main effects were analyzed using SPSS. In all regression cases the highest significance level was further explained.

Conclusions

Neither the Call Initiators nor Call Receivers were influenced by the type of call made. The call type was randomized, along with the participants' role within the experiment. This indicated that the blind study, along with the randomization of participants, was an appropriate design for this experiment. The experiment was successful because of the stringent conditions under which the experiment was conducted. This included measuring the line specifications before the experiment was conducted, as well as consistent measurement of the DSL connection before each VoIP call was placed. This ensured that the corresponding transmission speed with respect to upload and download transmissions for each VoIP-initiated call was recorded.

This dissertation focused on three specific research questions. The first of these research questions will be answered by the data derived from the Comparative Surveys completed by the Call Initiators and Call Receivers using a chi-square test.

- How does the perceived quality of VoIP calls compare to that of traditional landline-based communications in the perceptions of DSL broadband users?

Table 14 indicates the chi-square test results from the Comparative Surveys taken by the Call Initiators and Call Receivers. For 28 of the rounds a VoIP versus analog call was rated by 28 Call Initiators and 28 Call Receivers. The other 28 rounds were comprised of an analog versus VoIP call that was rated by a different group of 28 Call Initiators and

28 Call Receivers. As stated in Chapter Three, the call order, along with the participants were randomized in the experiment.

The conclusion section of Table 14 indicates the interpretations from the participants' data. The rating of "The Same" indicates a neutral rating for the participants. This is defined as a rating that does not favor one technology over another. However, an interpretation can be made from the ratings found in Table 14 that fell in the categories of worse, much worse, better, and much better. Regardless of call order, an equal number of participants rated the calls as worse and better. A conclusion can be made from the participants that rated the calls as much worse and much better. Almost 2% of the participants' believed that the analog call was much worse than the VoIP call as opposed to 5% of the participants' believed the VoIP call was much worse than the analog call. However, over 5% of the participants' believed that the analog call was much better than the VoIP call as opposed to over 14% of the participants who believed the VoIP call was much better than the analog call. It can be concluded that the majority of the participants in this study believed the VoIP calls were slightly better than the analog calls.

Table 14

Chi-Square Summary for the Comparative Surveys

	Both groups expected N[1]	Analog versus VoIP call observed N[1]	VoIP versus analog call observed N[1]	Conclusion
Much worse	11.2	1	3	The expected outcome for both groups was 11.2, both groups had an observed outcome much lower than the expected outcome.
Worse	11.2	14	14	The observed outcomes for both groups were slightly higher than the expected outcomes.
The same	11.2	27	20	The observed outcomes for both groups were much higher than the expected outcomes.
Better	11.2	11	11	The observed ratings for both groups were very close to the expected outcomes.
Much better	11.2	3	8	The observed outcomes for both groups were lower than the expected outcomes, however the number VoIP versus analog participants were closer to the expected outcome.
Chi-square		38.286	14.536	Critical value is 13.28 at a .01 sig. level with 4 degrees of freedom resulting in a conclusion that the participants' distribution was not equal to the expected values.
Degrees of freedom		4	4	
Significance		p. < .01	p. < .01	

The second research question will be answered from the multiple forward stepwise regression analysis for the cases of Y_{VCQI} and Y_{VCQR}, and the paired *t* test results for the Participants' Pretest and Posttest VoIP Surveys.

[1] N represents the number of participants that, under the null hypotheses of equally likely ratings, either expected to or actually did assign a given rating.

- Are the Call Initiators' and Call Receivers' Initial Beliefs with regard to the Call Quality of the VoIP product correlated with their actual posttest Relative Perceived Call Quality ratings of VoIP?

At the .01, .05, and .10 significance levels there was no significant correlation with the actual Pretest Survey and the posttest results for either the Call Initiators or Call Receivers. At the .01 significance level the paired *t* test also indicated that there was no statistically significant correlation with the Pretest and Posttest VoIP Surveys.

The third research question will be answered from the multiple forward stepwise regression analysis for the case Y_{PBCCAV}.

- Are differences in transmission speeds correlated with the perceived quality of the VoIP calls?

In the regression case of Y_{PBCCAV} the transmission speed proved to have a small, but statistically significant correlation with the Perceived Call Quality of VoIP. Over 22% of the variation of Analog Versus VoIP Relative Perceived Comparative Call Quality can be explained by the significant interaction effect of Previous VoIP Experience Receiver and Transmission Speed Up. It can be concluded that transmission speeds could affect the quality of the VoIP call. Transmission speeds may not have played roles in other variables, because the transmission speeds were kept above the minimal acceptable threshold of 90 Kbps (Vonage, 2004) for VoIP calls.

Recommendations

The VoIP experiment was designed to be easily replicated by future researchers. As stated in the limitations section, this research was confined to a DSL-enabled connection at one geographical location. The experiment may be used in the areas of business and further research. As VoIP services become more popular, a business may want to set up a similar experiment to test VoIP services before committing to a large expense.

VoIP testing should be done by the personnel that will use the services each day. It is important to test VoIP with the type of broadband connection that will be used. The quality of VoIP service is very much dependent on the broadband connection. If the speeds of the broadband connection are limited, the end user may gain a different perspective about VoIP than if ample bandwidth is available. Furthermore, it should be stated that "burstable" services such as wireless may significantly degrade the quality of service of VoIP. Burstable services are services that produce high amounts of bandwidth for a limited time. After which, the service significantly decreases in speed depending on available bandwidth. Therefore, more work can be done on different types of broadband connections and at different geographical locations. Connection types may include broadband cable, T1s, and wireless connections.

Geographical locations may impact the perceptions of VoIP services. In a heavily congested area, the identical group of participants may have indicated a very different outcome than they did in this less congested environment. Heavily congested areas can increase packet loss, resulting in a significant decrease in the quality of the VoIP call. Dedicated networks that have experience handling VoIP calls can increase the quality of the VoIP call. Furthermore, businesses may be able to find "guaranteed" VoIP services. This is normally done on a network designed for VoIP that transmits the call through a dedicated circuit from one central switching office to another. This guarantee may be conditional on the type of connection users have from their business to the central office.

Using human participants can prove to be quite time consuming and costly. A recommendation for future researchers is to conduct a similar experiment in which the researcher may have access to a large group of participants. By having access to a large group of participants this can help mitigate the time and expense occurred in an experiment. Another recommendation is to vary the study throughout different times of the day. Network congestion may be heavier in the morning than in the evening, thus affecting the participants' perception of VoIP service.

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Appendix A1
Survey Instruments

Pretest Survey - Initial Beliefs of VoIP

Date ____/_____/2004
Effective Date 8/1/2004

Instructions
Please answer the following questions before using the VoIP phone.
The questions to be answered will help you rate different aspects of overall call quality.
Descriptions of these call quality aspects are in italics. At the end of this survey
you will be asked to rate the overall call quality based on what you believe you will observe.

Have you ever used VoIP service before? Yes No *(circle one)*

1- Unbearable 2 - Below Average 3 - Average 4 - Above Average 5 - Excellent

Use the values above to rate the current call quality in the following categories.

How clear do you think the call will be? ***Please Circle***
Overall Clarity **1 2 3 4 5**

How do you think the volume will be?
Volume Level **1 2 3 4 5**

Do you think there will be any other noise on the lines such as cross talk (hearing other conversations), fading (the conversation going in and out), or other noise that affects your ability to hear the other party?
Outside Interferences **1 2 3 4 5**

Do you think there will be any noticeable background noise, such as static, humming, or buzzing, in the conversation?
Background Noise **1 2 3 4 5**

How clear do you think the conversation will sound to the other party?
Full Duplex Conversation **1 2 3 4 5**

Based on the questions you have just answered, what do you think the overall quality of VoIP will be?
Final Call Quality Rating **1 2 3 4 5**

Survey 1

Date ____/_____/2004
Effective Date 8/1/2004

Instructions

The questions to be answered will help you rate different aspects of overall call quality. Descriptions of these call quality aspects are in italics. At the end of this survey you will be asked to rate the overall call quality based on what you have observed.

1- Unbearable 2 - Below Average 3 - Average 4 - Above Average 5 - Excellent

Use the values above to rate the current call quality in the following categories.

How clear is the call? ***Please Circle***
Overall Clarity **1 2 3 4 5**

How is the volume of the incoming call?
Volume Level **1 2 3 4 5**

Is there any other noise on the lines such as cross talk (hearing other conversations), fading (The conversation going in and out), or other noise that affects your ability to hear the other party?
Outside Interference **1 2 3 4 5**

Is there any background noise such as static, humming, or buzzing?
Background Noise **1 2 3 4 5**

How does the conversation sound to the other party? Please ask the other party how clear your voice is.
Full Duplex Conversation **1 2 3 4 5**

Based on the questions you have just answered, how would you rate the overall quality of this call?
Final Call Quality Rating **1 2 3 4 5**

Survey 2

Date ____/_____/2004
Effective Date 8/1/2004

Instructions

The questions to be answered will help you rate different aspects of overall call quality. Descriptions of these call quality aspects are in italics. At the end of this survey you will be asked to rate the overall call quality analysis based on what you have observed.

1- Unbearable 2 - Below Average 3 - Average 4 - Above Average 5 - Excellent

Use the values above to rate the current call quality in the following categories.

How clear is the call? ***Please Circle***
Overall Clarity 1 2 3 4 5

How is the volume of the incoming call?
Volume Level 1 2 3 4 5

Is there any other noise on the lines such as cross talk (hearing other conversations), fading (the conversation going in and out), or other noise that affects your ability to hear the other party?
Outside Interference 1 2 3 4 5

Is there any background noise such as static, humming, or buzzing?
Background Noise 1 2 3 4 5

How does the conversation sound to the other party? Please ask the other party how Clear your voice is.
Full Duplex Conversation 1 2 3 4 5

Based on the questions you have just answered, how would you rate the overall quality of this call?
Final Call Quality Rating 1 2 3 4 5

Comparative Survey

Date ____/_____/2004
Effective Date 8/1/2004

Instructions

The questions to be answered will help you compare different aspects of overall call quality. Descriptions of these call quality aspects are in italics. At the end of this survey you will be asked to rate the overall call quality analysis based on what you have observed.

1- Much Worse 2 - Worse 3 - The Same 4 - Better 5 – Much Better

Use the values above to compare the call quality of call 1 to that of call 2, in the following categories.

How did call 1 compare to call 2 in terms of call clarity (how clear did the call sound)? ***Please Circle***
Call Clarity **1 2 3 4 5**

How was the volume level of call 1 compared to the volume level of call 2?
Volume Level **1 2 3 4 5**

Was there any outside interference such as fading (the conversation volume going softer and louder), or crosstalk (hearing other peoples conversations) on call 1 in comparison to call 2?

Outside Interference **1 2 3 4 5**

Was there any noticeable background noise such as static, humming, or buzzing on call 1 compared to call 2?

Background Noise **1 2 3 4 5**

Did call 1 sound clearer without any breakups in comparison to call 2?

Full Duplex Conversation **1 2 3 4 5**

Based on the questions you have just answered, how would you compare the overall call quality of call 1 versus call 2?

Final Call Quality Rating **1 2 3 4 5**

Appendix B2

Informed Consent Form

Informed Consent Form

A Comparison of VoIP and Analog Perceived Call Qualities

Purpose. You are invited to participate in a research study. The purpose of this study is to examine the perceived call qualities between VoIP and analog-initiated calls. I am interested in your opinions and perceptions about the call quality of each call.

Participation requirements.for call initiators: You will be asked to complete 4 surveys about the different call quality perceptions along with placing two phone calls. One phone call will be VoIP-initiated while the other phone call will be an analog-initiated call. The experiment will take approximately 10 minutes.

Participation requirements for call receivers: You will be asked to complete four surveys of different call perceptions along with receiving two phone calls from the call initiators'. These phone calls will be VoIP and analog-initiated. The experiment will take approximately 10 minutes.

Research Personnel. The following individual is involved in this research project and may be contacted at any time: Kirby Scheer kscheer@arvig.net / 218-483-4282

Potential Risk/ Discomfort. There are no known risks associated with this study, and all survey answers will remain confidential and will be conducted in an anonymous fashion.

Potential Benefit. There are no direct benefits to you for participating in this research. No incentives are offered. The results will be of technological interest to the business communications and telecommunications industries.

Anonymity/ Confidentiality. The data collected in this study is confidential. All survey responses are anonymous such that your name is not associated with them.

Right to Withdraw. You have the right to withdraw from the study at any time without penalty. You may omit questions on any questionnaires if you do not want to answer them.

I would be happy to answer any question that may arise about the study. Please direct your questions or comments directly to Kirby Scheer.

Signatures

I have read the above description of the VoIP and Analog Perceived Call Qualities Study and understand the conditions of my participation. My signature indicates that I agree to participate in the experiment.

Participant's Name : ____________________ Researcher's Name: ________________

Participant's Signature: ____________________ Researcher's Signature:______________

Date:____________

Informed Consent, (NorthCentral University, 2004).