CHARACTERIZING VOICE TRANSMISSION PERFORMANCE FOR EVOLVING BUSINESS NETWORKS

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(continued on page 80)

Today's large businesses typically have complex voice networks that combine carrier-provided network services and private facilities (customer-owned or leased facilities dedicated to a single customer). Ensuring high-quality voice transmission in these networks is a complicated task, well suited to computer modeling techniques. We describe a new approach to computer modeling of network performance, one in which the expected transmission performance of all voice calls throughout a given network can be displayed easily and analyzed collectively. We use probability of occurrence information for call origination and routing to assess network performance realistically. Our approach augments existing tools and techniques to give network designers and network managers a better understanding of the voice transmission performance experienced in today's business networks. It also gives planners of new network architectures or elements a better understanding of the performance of future business networks. We illustrate our technique with examples, explain how we represent and select connections, and describe the elements that determine voice transmission performance.

Introduction

Since at least 1902, large businesses have used dedicated switching systems for their on-site voice communications. These switching systems are called private branch exchanges (PBXs) when located on a business' premises and centrex when located in a carrier's or service provider's office. Businesses that had PBXs at several sites began to network their switches over private, dedicated facilities, but only over short distances. Technological advances in the 1950s

Panel 1. Abbreviations, Acronyms, and Terms

centrex — an exchange network business service that provides PBX-like features.
DCS — Distributed Communications System
EIA — Electronic Industries Association
ETN — Electronic Tandem Network
GOS — grade of service
IEC — inter-exchange carrier
LEC — local exchange carrier
PBX — private branch exchange
PCM — pulse code modulation
ROLR — receive objective loudness rating
SDN — Software Defined Network
TIA — Telecommunications Industry Association
TOLR — transmit objective loudness rating

popularized private facilities by enabling them to be used over long distances (i.e., between cities), by adjusting loudness levels automatically, and by repeating dialed digits. In the late 1970s, businesses began to use sophisticated PBX routing capabilities to route calls automatically during overload conditions, to allow a common 7-digit dialing plan throughout the network, and to allow centralized attendants to cover many PBXs. AT&T's Electronic Tandem Network (ETN) introduced this type of private networking. ETN was initially introduced on Dimension® PBXs.

Since the 1970s, large businesses have continued to develop their networks to include AT&T's Distributed Communications System (DCS) and AT&T's Software Defined Network (SDN) service. DCS is a private network capability that makes many station and attendant features function the same to the user regardless of whether one or more PBXs are involved in the feature's operation. SDN is an virtual private network service of AT&T. Its network switches and customer-specific databases provide private network features like 7- or 10-digit dialing and authorization codes, with pricing determined by call volume. Businesses have been changing from entirely analog to largely digital networks that carry voice, data, and video information. This change started with the availability of digital private lines in the mid-1980s. Recently, it has been fueled by businesses expanding their mostly private networks to include digital, carrier-based services. Now that networks are built in a variety of architectures and with many types of telephones, facilities, and services, the important task of ensuring high-quality voice transmission in those networks has become complex.

A key element in ensuring high-quality voice transmission is the control of echo. To satisfy this need, domestic IECs, including AT&T, have been using more echo cancellers in their switched networks. We expect that soon nearly all interexchange calls will be equipped with cancellers. Today's software-controlled PBXs, such as AT&T's Definity® communications system, have evolved to control echo and adjust voice loudness with flexibly applied deamplification, or loss. By slightly reducing signal amplitude at key locations in a connection, interference from circulating signals such as echo and feedback can be dissipated without substantially reducing the voice volume heard by the listener. Inserting loss in switching systems, including PBXs, is still an important factor in determining transmission quality. We must continue to evaluate which losses are acceptable and under which network circumstances they should be applied. Recently, the EIA and TIA have identified new PBX losses for digital trunks, as part of an ongoing effort to maintain or improve the quality of voice transmission.

In this paper, we describe a new approach to computer modeling of voice network performance. In doing so, we examine major contributors to voice transmission quality in networks involving PBXs. We discuss modeling techniques that encompass the variety of networks mentioned above and present network performance results that exhibit the range of transmission quality that can be achieved by today's business networks.

During the last few years, the rapid conversion of IEC networks to digital transmission has begun to simplify prediction of end-to-end voice transmission quality. In the coming years, digital telephones and digital LEC facilities and switches will pervade business networks. The resulting voice transmission quality would become less variable and its prediction less complex if μ-law pulse code modulation (PCM) remained the domestic standard for digitally encoding analog voice signals. This simplification will likely not be realized, however, because many evolving communications technologies use voice bandwidth compression or expansion techniques that affect transmission quality. Mobile telephone services, personal communications networks, and voice messaging networks typically use compression techniques to conserve spectrum, bandwidth, or storage requirements. Compression generally lowers transmission quality. Expansion techniques use available spectrum, bandwidth, or storage more effectively to offer "high-fidelity" voice transmission in traditional business applications, which tends to improve transmission quality. In the face of such constant change, we can use computer modeling to understand general network transmission performance.

Background. Work done from the late 1960s through the 1970s led to techniques for predicting the subjective opinion of transmission quality.[1] That work was enhanced and adopted by standards bodies in the 1980s.[2] Predicting perception of voice transmission quality for comparison purposes is straightforward for networks in which:

- Absolute delay does not get too long to hinder voice communication (under about 300 milliseconds round-trip delay)
- Analog or μ-law PCM switching and transmission are used
- Standard, well-designed telephone sets and battery feed circuits are used.

From a subjective model, we can predict user opinion of transmission quality based on the loss, noise, echo, and delay characteristics of the connection, and on the electrical and acoustic characteristics of the telephone sets and their loops.

Techniques have been developed to measure the critical transmit and receive characteristics of a telephone set and its loop (including battery feed current for certain telephones). When these characteristics are converted to a form easily combined with electrical losses (analog) or equivalent signal levels (digital), they are commonly referred to as transmit objective loudness rating (TOLR) and receive objective loudness rating (ROLR). TOLR and ROLR represent the telephone's efficiency in converting speech to electrical levels or encoded signal levels in the transmit direction, and in reversing the process in the receive direction.

Network design would be simple if telephones were ideal and if echoes did not occur within networks. We would choose the acoustic-to-electrical and electrical-to-acoustic efficiencies of telephone sets and a constant amount of loss in the PBX to make talking and listening volumes comfortable. In past and present networks, however, we commonly encounter four-wire to two-wire conversions, which cause a reflection, or echo, of part of the signal energy passing through the conversion point. These conversions occur as the signal leaves a four-wire switching system, such as a digital PBX or a LEC's digital switch, and enters a two-wire loop connected, for example, to an analog telephone. The conversion occurs in the switch's line or trunk interface circuitry. Echo returning to the talker is called talker echo; it is generally the most important echo in determining subjective opinion of transmission quality. Listener echo is energy from the talker that reflects twice before proceeding back to the listener — once at the listener end and then at the talker's end. Echo is harmful to the perceived performance of a connection, and the performance decreases as the echo's delay increases.

Today, we can use echo cancellers to reduce to insignificance the echoes caused in four- to two-wire conversions. Although IEC networks are beginning to use echo cancellers in all connections, echo cancellers are not generally required or considered cost-effective in private networks or LEC networks. Those networks typically have delays small enough to make echo effects manageable by inserting loss. There are certainly exceptions to this; a geographically spread private network with long delay will have improved performance with echo cancellers. The most practical method to maximize the performance of a connection without adding echo cancellers is to apply transmission loss in the connection. If we insert too much loss, performance will suffer because the delivered signal will be inaudible. If we insert too little loss, we will not adequately minimize echo's ill effects. As delay in the connection grows (e.g., because of circuit length), the loss needed to overcome echo eventually disrupts normal conversation. In these cases, echo cancellers are required.

Over the years, AT&T has worked with other industry participants in standards organizations to develop network loss plans that specify the location and quantity of loss to be inserted. In the past, this plan encompassed LEC and IEC network switches and PBXs. Loss insertion is no longer required, however, for digital IEC networks with pervasive echo cancellation. As a result, the end-to-end performance variations described later in this paper stem from the private and local-exchange network portions of connections and from the remaining nondigital access lines to IECs, where loss, noise, and echo variations continue.

As we stated earlier, the conversion from a four-wire switch to a two-wire analog telephone loop has been a major source of echo. Digital telephones, which are functionally four-wire devices, are not subject to the four-wire to two-wire conversion. The increased use of digital telephones, therefore, will significantly reduce the generation of echo. Voluntary industry standards control generation of echo within ISDN telephones to the point where a low level of fixed transmission loss, independent of distance, will be satisfactory for domestic calls.[3]

Unlike loss, noise is never intentionally inserted into voice connections. The intent has always been to reduce noise to as low a level as practical. Perceived performance of a connection depends on the noise level, so it must be considered when we predict a connection's transmission performance. Although digital transmission (increasingly encountered in long distance transmission) makes noise level less of a factor than it was in the past, noise level still influences connections with lengthy analog facilities.

From human factors studies conducted at AT&T Bell Laboratories, Cavanaugh et al.[1] developed a model correlating subjective opinion of a connection to its measurable characteristics. That model allows a subjective grade of service (GOS) rating of an existing or hypothetical connection to be calculated from its loss, noise, echo, delay, and telephone set and loop characteristics. The result typically is expressed as the percent of a population of test subjects who would rate the connection as "good or better" or "poor or worse." Because opinions vary with time and between test populations, the percentage GOS result of a connection cannot be expected to match numerically a later judgment by a different test population. However, relative differences of a few percent or more can be trusted. For example, if a particular connection has a better GOS result than another, according to a given test subject population, then that connection can be expected to be judged better than the other in later testing with different test subjects.

Understanding Overall Network Performance. GOS performance analysis involving private networks traditionally has been conducted on a single connection at a time. When predicting the effects of a change, such as altering switch-inserted losses, GOS ratings typically were calculated for a set of connections affected by the change. It was hoped that the chosen connections represented a cross-section of telephone calls in the network. Manually specifying the parameters for GOS calculations is tedious. The seemingly infinite variety of calls made it impractical to analyze the impact of a change on all possible connections in a network. The computer program described in this paper makes analysis of transmission performance possible for an entire network. It also organizes the numerous GOS results into a picture of network-wide performance. These graphs can be used to predict how changes in echo canceller deployment, loss plan, facility delay, switch delay, noise, etc., will affect transmission performance for all or a subset of network users.

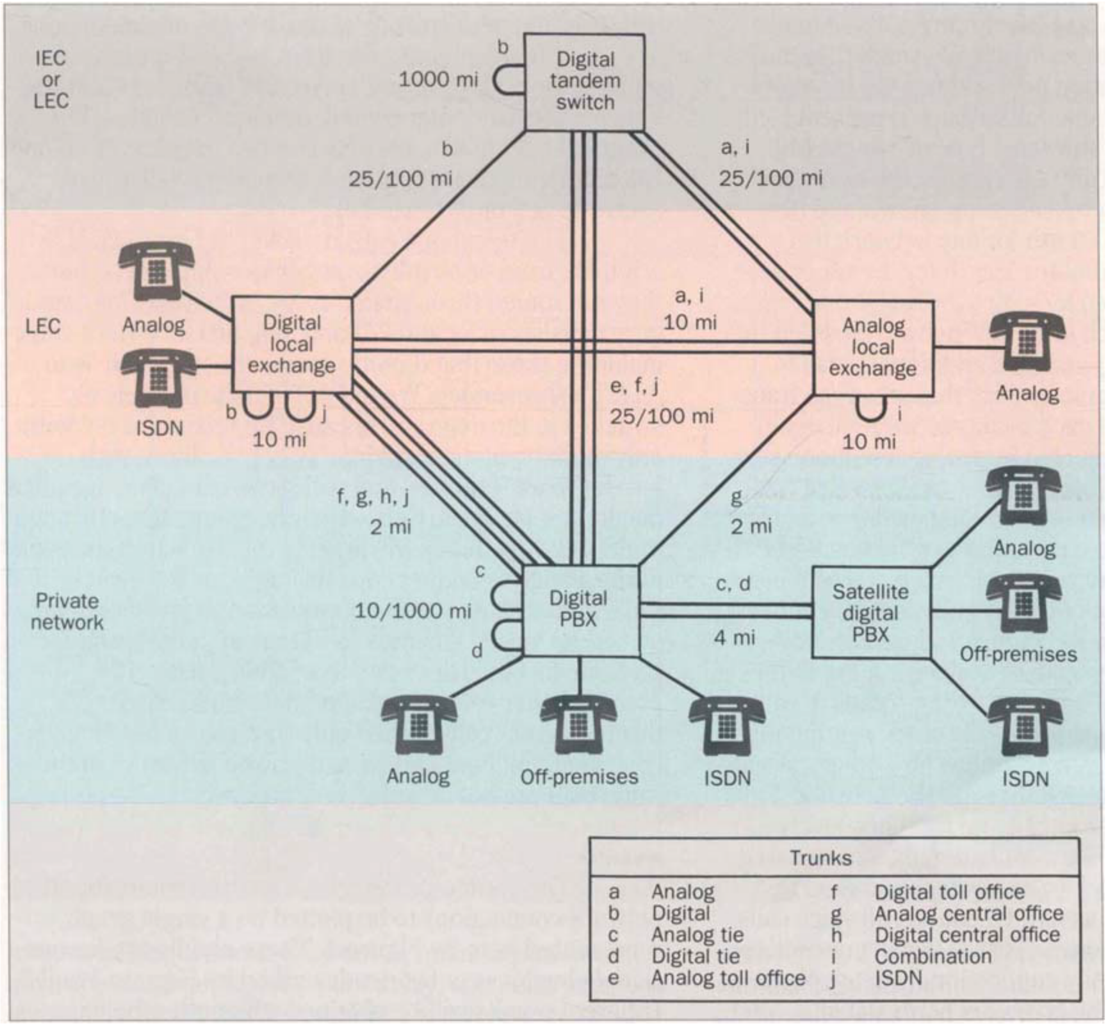

Figure 1. A connectivity diagram showing each type of switch that can be encountered in a telephone network, and the types of facilities that connect them. Small loops represent facilities that connect the same type of switch. Mileages shown at the facilities define facility lengths for later examples. Where pairs of mileages are separated by a slash, the first one defines the facility length for one network in an example, and the second for the other network.

Modeling the Network

To model a telephone network that we wish to study, we need to represent two things: first, how the equipment and facilities are connected, and second, how calls are routed through the network. The first item is represented in our model as a connectivity diagram. The second is represented as a routing diagram. We show sample networks to describe the technique, but the diagrams may be changed to represent other business networks.

The Connectivity Diagram. Figure 1, the connectivity diagram, illustrates each type of switch that can be encountered in a telephone network and the types of facilities that connect them. Small loops represent facilities that connect two of the same type of switch. Mileages shown define a facility's length for the examples that follow. (Where two mileages are shown, the first number defines facility length for one network in a graph, and the second defines length for the other network.) The diagram also indicates the telephone types and connections for each switch. When we modeled this diagram in software, we included additional detail to define transmission characteristics that affect the transmission performance of each facility or piece of equipment. Performance is affected by analog telephone sets and their loops, ISDN telephone sets, facilities that connect switches, echo cancellers on long facilities, and the switching fabric and interface ports of switches. In brief, the connectivity diagram summarizes the physical possibilities for connecting telephone equipment in a network.

The Routing Diagram. Figure 1 illustrates how switches and telephones may be connected, but it does not show the number of switches or the routes of calls through the network; that is the role of the routing diagram. The routing diagram describes how voice calls are routed from switch to switch through the network. Calls may follow very simple routes (e.g., from one PBX to another), or calls may follow more complex routes (e.g., from a PBX to a LEC switch to IEC switches, and on to another LEC switch and a PBX). Of course, all voice calls begin and end at telephones. Arrows depict connections from switch to switch. Any connection from telephone to telephone that follows the arrows is permissible.

We can represent all possible routes of telephone calls in a single routing diagram. However, we can gain insight from studying a subset of those calls. Figures 2a and b are examples of subset routing diagrams. Figure 2a shows routes of private network calls. Figure 2b shows routes of calls sent to LECs or IECs. Each subset can be studied separately using our model to determine the transmission performance of those types of connections.

The telephone calls illustrated in Figure 2a originate from and terminate at PBX telephone sets and use only private, customer-owned, or leased facilities. We excluded telephone calls between two telephones on one PBX from the private routing category, as well as calls routed to LEC or IEC switches.

The telephone calls depicted in Figure 2b also originate from or terminate at PBX telephone sets, but they are routed through LEC or IEC switches. This category consists of local (LEC) and long distance (IEC) calls, including those that directly access the IEC, such as to AT&T's SDN service. We did not include multiple IEC switches in the diagram, because we model the IEC with only digital switches and interswitch facilities, with no loss between switches, and with echo cancellers installed on all their facilities. Consequently, connections through multiple IEC switches will provide the same transmission performance as connections through one IEC switch. Calls in this category do not traverse private, customer-owned, or leased facilities for the most part, except for a private facility from a PBX to a satellite PBX.

Other routing subsets (e.g., calls routed only through a LEC, calls routed only through an IEC, or calls concatenating both carrier and private networks in the same call) are not included in this paper.

Results

The computer model allows GOS results for all network connections to be plotted on a single graph, represented here by Figure 3. These results are for our sample business networks described by Figures 1 and 2. Different results will be obtained when other businesses' telephone networks are modeled. An exhaustive algorithm searches for all calls in the network defined by the connectivity diagram that also meet the routing diagram requirements. The computer program calculates percent good or better ratings for each of the calling and called telephones on every call. This rating is based on the ROLR of the evaluator's telephone and the acoustic-to-acoustic

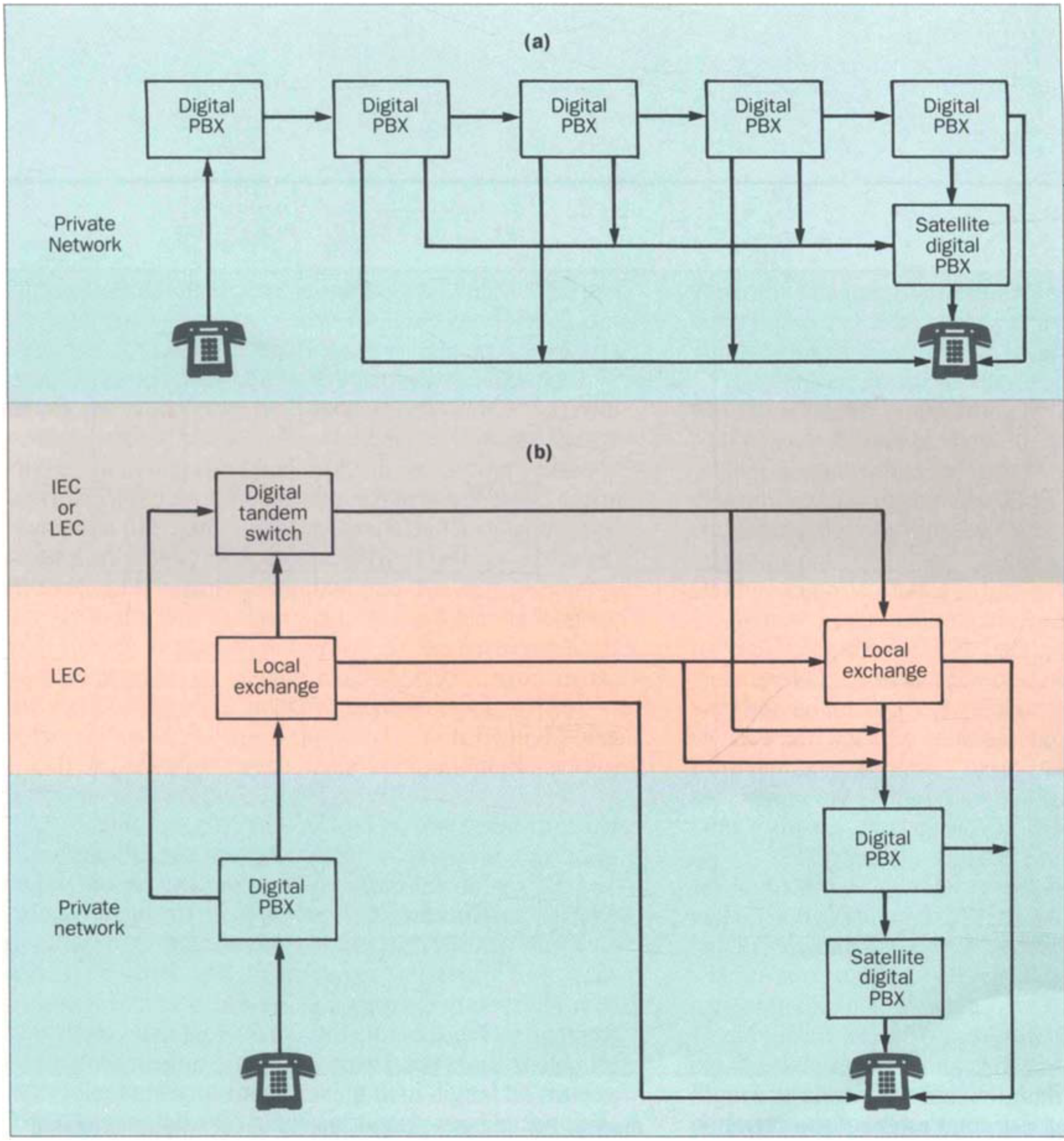

Figure 2. (a) A sample routing diagram showing how private network calls that use only private (customer-owned or leased) facilities are routed from switch to switch. All calls begin and end at telephones. The arrows depict the permissible connections from switch to switch. (b) A routing diagram for calls routed through carrier network switches. Telephone calls originate from or terminate at PBX telephones and are routed through LEC or IEC switches.

loss, switching noise, facility noise, quantization noise, environmental noise, echo path delay, and echo path loss characteristics of the connection. The percent good or better rating is a GOS measurement defined as the percentage of reference test subjects who rate a particular call as good or excellent from the choices excellent, good, fair, poor, and unsatisfactory. The percent good or better rating in these graphs is based on the Murray Hill reference set of test subjects.[1,2] The x-axis of the graph shows the percent good or better result for each connection. The y-axis represents individual calls as a cumulative distribution function (CDF) of 0 to 100 percent of the calls, obtained by ordering the percent good or better ratings to increase monotonically. Sorting the ratings into a CDF organizes what is otherwise a cloud of several thousand data points into an orderly curve that can be easily understood and interpreted visually.

Private Network Calls. Figure 3a is a plot of GOS results for all possible calls in our sample private network. It follows the routing rules of Figure 2a. The two

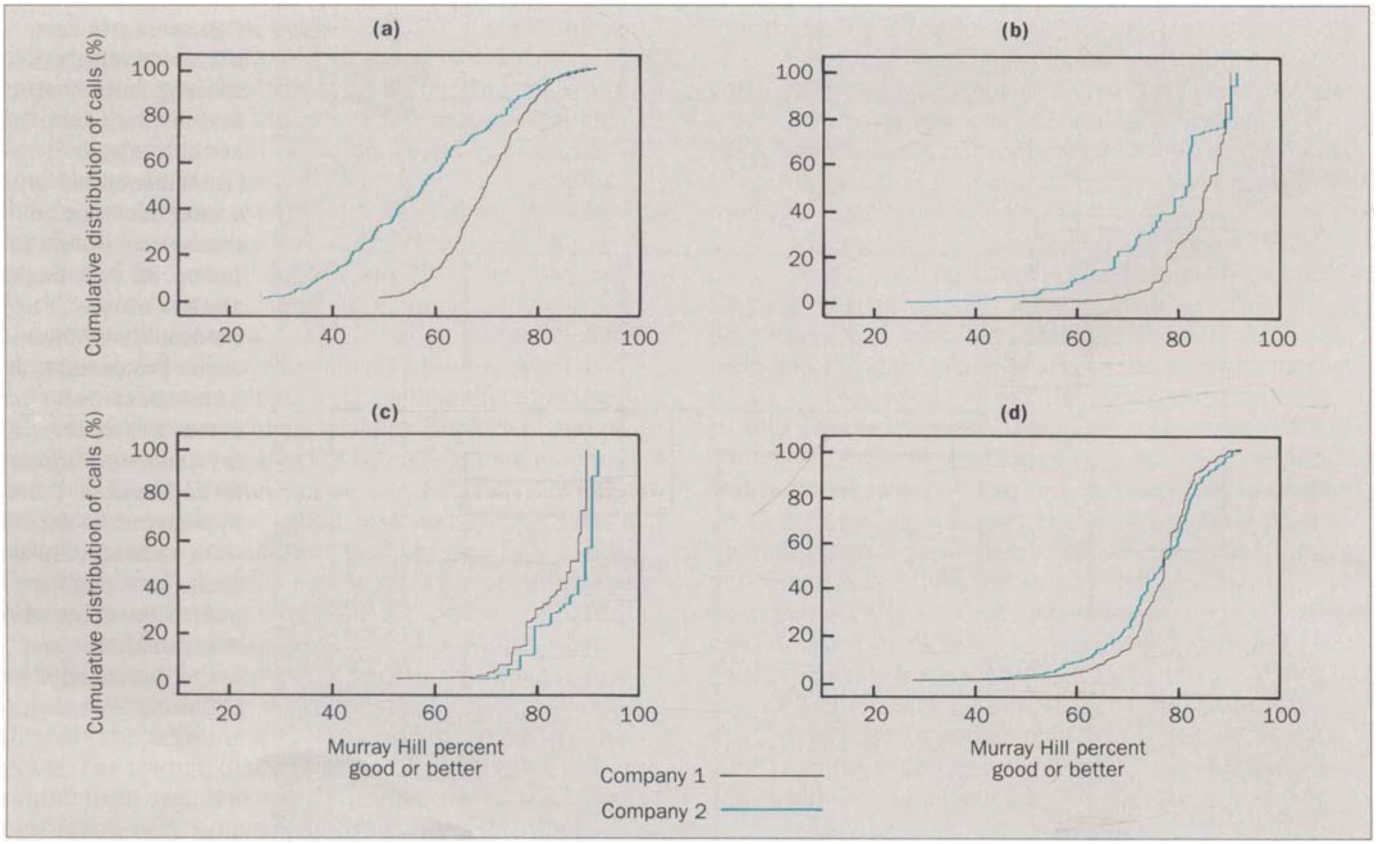

Figure 3. GOS ratings of all calls that can be made in two different (a) private networks following the routing rules of Figure 2a. The switches of Company 1 are located close together; the switches of Company 2 are widely separated geographically. (b) Each call is then weighted according to its probability of occurring. For example, we vertically stretch the data point for a two-PBX call to reflect how frequently that call occurs in this company's network. Similarly, we shorten the data point of a five-PBX call to reflect how rarely it occurs. (c) The model may be used to predict how changes to the network will affect network-wide performance. For instance, these results occur when the analog tie trunks are replaced with digital tie trunks. These weighted curves may be compared directly to the previous graph, in which half the tie trunks were analog. (d) This graph compares networks with calls routed to LECs and IECs according to the routing rules of Figure 2b. Company 1 makes mostly local calls, and Company 2 makes mostly long distance calls.

curves in the figure correspond to two hypothetical business networks. One network's switches are located close together; the other's switches are widely separated geographically. Because the results are sorted independently for each curve, no direct correspondence of connections exists between the two curves. Plotting the two curves on one graph emphasizes how varying the distance between switches changes a network's GOS performance. In the opinion of its users, a network whose GOS results fall predominantly to the right will have better transmission performance than one that falls predominantly to the left. In this example, the business network with closely located switches will get better transmission performance for most calls, since its curve is mostly to the right of the other network's curve. Because the curves cross at the top of the graph, we can tell that the widely separated network has a few more of the best performing connections.

One can easily be misled by this graph, however. Although the results incorporate all possible telephone calls, they do not suggest how often the different types of calls are made on each network. Some calls occur infrequently (e.g., calls connected through five PBXs in series), whereas other calls occur frequently (e.g., calls from one PBX to another). The graph also does not reflect how often each type of equipment is used. For instance, a company's business network may have more analog than ISDN telephones, or more analog than digital tie trunks. The GOS graph would more accurately predict the opinion of network users if it accounted for how often each type of equipment is used and how often each call routing occurs.

To make the performance plots more useful, the remaining graphs are weighted by estimates of how frequently telephone calls will occur. This technique emphasizes calls expected to occur frequently and deemphasizes calls expected to occur rarely. These weights are incorporated into the plot as a width on the y-axis. If a certain type of connection is judged to occur 25 percent of the time, then its GOS data point will be stretched vertically to a height of 25 percent of the y-axis. Connections that occur infrequently will be plotted as short data points to show their relative insignificance.

Figure 3b shows the result of weighting the Figure 3a connections according to their probability of occurring. Figure 3b reflects hypothetical, but realistic, probabilities of occurrence for each element in the connectivity diagram and for each route on the routing diagram. By comparing the two figures, we observe that network user opinion would be better than Figure 3a suggests. Thus, for this scenario, the calls that perform better occur more frequently than those that perform poorly. The weights used in this exercise were estimated by experienced people attending an industry PBX standards meeting. These weights model a typical business network. However, the weights of a specific business network may be different from our assumptions. Therefore, values that pertain to a network being studied must be applied to the results before any conclusions can be drawn.

This analysis tool can predict how a proposed network change will affect network-wide performance. For instance, we assumed that the networks of Figure 3b had half-analog and half-digital tie trunks. What would happen if these two companies were to upgrade their analog tie trunks to digital? What amount of service improvement would they realize? We can calculate the answer by changing the tie trunk weights and rerunning the program. Figure 3c shows the results. Both networks improve, but performance for the company with widely separated PBXs improves markedly. Its performance surpasses that of the company with closely located switches. Why?

Analog facilities are noisy, and the amount of noise increases in proportion to the length of the facility. Therefore, the network with widely separated switches benefitted greatly by converting the long analog trunks to digital. Replacing the noisy trunks unmasked an advantage that the widely separated network had all along: echo cancellers. Long facilities have long transmission delays that cause even small echoes to be distracting. Thus, we installed echo cancellers on the 1000-

mile tie trunks in our example to cancel the echoes. The closely located network could match the widely separated network's performance if it added echo cancellers to its tie trunks. Although transmission delays in the close network are short enough that users will not hear distinct echoes, users prefer having echo cancellers even on short facilities. Performance of the closely located network would improve by 8 percent in this example if echo cancellers were installed.

The half-digital and all-digital tie trunk mixes assumed in Figures 3b and c represent extreme cases. In recent years, the half-digital network was typical. IEC portions of long tie trunks tended to be analog, and access from smaller customer sites commonly was analog. Today, only a few private networks have converted to using all-digital tie trunks. Although the IEC portion of today's tie trunk is usually digital, digital access is still not uniformly affordable for smaller customer locations.

Calls Routed to LEC and IEC Networks. Figure 3d shows GOS results for sample networks that route calls to LECs and IECs. One business makes predominantly local calls, while the other business makes mostly long distance calls. LEC switches and access facilities to both LECs and IECs were weighted with mixtures of analog and digital technologies expected to be typical in 1992. Again, the weighting heavily affects the outcome. It must be tailored to match the expected usage of analog or digital access facilities, the services and equipment of the LEC and IEC, and the expected routing of calls.

Computer Program Description

To find and construct telephone calls of interest, the computer program creates two general graph data structures in memory: the connectivity graph and the call-routing graph. These graphs are constructed in memory from data files that describe the connectivity and routing diagrams, which may be changed to describe other network connectivities or call routes of interest. For example, Figures 1 and 2a were used to plot GOS results for private network calls, and Figures 1 and 2b were used to plot GOS results for calls routed through the LEC and IEC networks.

Exhaustive Search Algorithm. After the computer program builds general graph data structures for the connectivity and call-routing graphs, a recursive algorithm searches out all possible calls in the described network. The program begins at a PBX telephone and systematically routes calls from all facilities leaving that PBX. It then continues routing them from succeeding switches, building calls that meet the connectivity and routing constraints. The connectivity graph ensures that telephones, lines, switches, and facilities are connected properly, and it supplies the transmission characteristics needed to calculate the GOS. The routing graph ensures that each call is routed correctly, for example, avoiding routes that contain an infinite number of switches or routes that travel from an IEC to a LEC and back. The recursive algorithm investigates every alternative as it searches for valid connections. When the program encounters another telephone, it calculates GOS ratings for both the calling and called telephones from the transmission characteristics (described later) of that call's components. Finally, the computer program sorts and plots the GOS results in the network transmission performance graph.

Determinants of Performance. The model represents the singular transmission characteristics of each type of switch, facility, and telephone. These are combined for each telephone call as input parameters for the subjective GOS equation.

Switch characteristics. The characteristics of PBXs, LEC switches, and IEC switches that affect transmission performance are noise, delay, echo, and insertion loss. These are attributes of either the switch fabric or the switch interfaces.

Digital switching introduces delay but no noise into calls, while analog switching introduces noise but insignificant delay. Either type of switch will add quantization noise if it must convert signals from analog to digital. Quantization noise is caused by digitally encoding the level of the analog signal. We assume that neither voice bandwidth compression nor expansion systems, which would add noise and delay, is used in the networks.

Impedance mismatches in the hybrid circuitry at four-wire to two-wire interfaces cause echo. The echo is attenuated by an amount known as transhybrid loss, which is used in determining the most significant echo in a connection. The most significant echo of a hybrid is reflected toward the four-wire facility.

Switches insert loss between some, but not all, ports to control echo or adjust loudness levels. The amount of loss inserted is described in a table called a loss plan, which is defined for each type of switch. An industry standard loss plan for digital PBXs is specified jointly by the EIA and the TIA.[4]

Facility characteristics. For each facility, the model assumes a fixed (if any) amount of loss and a fixed length in miles. Losses conform to current network design practices. The model assumes that analog facilities are designed for fixed loss rather than via net loss. Via net loss administration guidelines for trunks were adopted by the Bell System in 1953,[5] but fixed loss values are being used today. For the two networks, close and widely separated, we chose the facility lengths shown in Figure 1. Facility length affects round-trip delay, the decision to include echo cancellers, and, for analog facilities, the amount of noise introduced. The model includes echo cancellation on all facilities longer than a specified length in miles and assumes that all facilities between IEC switches have echo cancellers. For simplicity, the model assumes that echoes passing through a facility with echo cancellers are completely cancelled.

Telephone set characteristics. The model accounts for the characteristics of telephones and their loops. TOLR and ROLR were described earlier. A telephone also allows a small amount of the received signal to be reflected back into the network. The reflection passes from speaker to microphone (called acoustic coupling) as vibrations carried through the handset, the air, or the listener's head, or the reflection may occur in the electronics within the telephone. In ISDN telephones, the combined reflection is small, and it is modeled as a large attenuation called echo return loss. For analog telephones, these reflections typically are insignificant compared to echoes caused in their adjoining switch port circuits by hybrid imbalance. They are therefore disregarded in the model. With an all-digital connection, the reflections discussed here will be the only echoes that occur in the connection.

Summary

In this paper, we presented a method for evaluating the voice transmission performance of large, complex business networks. We illustrated the method in private business networks and business networks that use LEC and IEC networks. The method applies in a new way the transmission quality subjective models designed by others to evaluate individual voice calls. Our method introduces three new techniques that allow us to evaluate performance of entire networks. First, we use a general graph data structures technique to represent a business voice network. To select connections to be evaluated, we apply a recursive algorithm restrained by a second general graph. Second, we factor in a probability of occurrence, or weighting, for each evaluated connection to account for its importance in the overall network performance. Third, we present the results in a format that displays an entire network's performance in a single figure. This approach lets network planners or designers predict the effects of network changes on overall network performance. By comparing the resultant network performance graphs of several alternatives, they can choose the change that gives them the best network performance.

References

1. J. R. Cavanaugh, R. W. Hatch, and J. L. Sullivan, "Models for the Subjective Effects of Loss, Noise, and Talker Echo on Telephone Connections," *The Bell System Technical Journal*, Vol. 55, No. 9, November 1976, pp. 1319–1371.
2. The International Telegraph and Telephone Consultative

Committee (CCITT), "Models for Predicting Transmission Quality from Objective Measurements," *Blue Book, Volume V, Telephone Transmission Quality*, Series P Recommendations, Supplement No. 3, Geneva, 1989.
3. Electronic Industries Association, SP-1920, *Acoustic-To-Digital and Digital-To-Acoustic Transmission Requirements for ISDN Terminals*, Draft 9, October, 1989.
4. Electronic Industries Association, EIA/TIA 464-A, *Private Branch Exchange (PBX) Switching Equipment for Voiceband Application*, February 2, 1989.
5. H. R. Huntley, "Transmission Design of Intertoll Telephone Trunks," *The Bell System Technical Journal*, Vol. 32, No. 5, September 1953, pp. 1019–1036.

Biographies (continued)

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