The Effect of Waveform Substitution on the Quality of PCM Packet Communications

ONDRIA J. WASEM, STUDENT MEMBER, IEEE, DAVID J. GOODMAN, CHARLES A. DVORAK, MEMBER, IEEE, AND HOWARD G. PAGE, MEMBER, IEEE

Abstract—**Missing packets are a major cause of impairment in packet voice networks. While it is easiest to allow these gaps in received speech to appear as silent intervals in reconstructed speech, speech quality is improved by filling the gaps with estimates of the transmitted waveform. We have investigated several estimation techniques for packets containing 16 ms of speech in a pulse code modulation format. The simplest method, packet repetition, extends from two percent to five percent, the acceptable ratio of missing packets. Here, acceptability is defined as a mean opinion score midway between "fair" and "good" on a five-point opinion scale. The most effective estimation technique (although not the most complex) is pitch waveform replication. It extends the acceptable ratio of missing packets to ten percent.**

I. Introduction

A. Missing Packets

PACKET communication techniques were originally devised for digital data transmission [1]. In packet data networks, excess traffic leads to delays in delivery of information. When a network is congested, packets are held in queues at entry points and switching nodes. In voice communication, on the other hand, long delays are intolerable and network delay budgets have a strong influence on the design of packet voice systems. Packets delayed beyond a certain limit are ignored by the receiving terminal or perhaps dropped at an intermediate node. Hence, congestion in packet voice systems leads to gaps in received speech.

The simplest way of dealing with the gaps is to treat them as silent intervals in the transmitted speech. This requires no signal processing at the receiving terminal nor does it require the terminal to distinguish gaps due to lost packets from gaps introduced by a speech activity detector at the transmitting terminal. On the other hand, spurious silent intervals are audible and disturbing to listeners [2], [3].

B. Background

To increase the tolerance of packet voice systems to lost packets, we have explored techniques for estimating the contents of missing packets. Our methods are based on "waveform substitution" which involves searching received packets for waveform segments that resemble the waveforms of missing packets. The simplest technique, *packet repetition*, uses the most recent received packet as an estimate of the missing one. Another approach, based on *pattern matching*, extracts packet-length segments from the received speech and uses them to replace the missing packets. A third approach is to estimate the pitch of the received speech and to *replicate* prior *pitch waveforms* for the duration of a missing packet. In contrast to packet repetition, pitch waveform replication and pattern matching tend to maintain phase continuity at the boundaries of substitution packets and prior received packets.

Generally speaking, waveform substitution requires the following operations.

1) Classification of transmission gaps as missing packets or silent intervals introduced by speech activity detectors. This classification can be based on time stamps and/or serial numbers in packet headers.

2) Storage of 16–48 ms of received speech.

3) Signal processing to extract from stored speech a pitch estimate or a waveform segment that will replace the missing packet.

With packet repetition, the signal processing task 3) is essentially eliminated and the required speech storage 2) is limited to one packet. The memory and signal processing requirements of pattern matching and pitch waveform replication are within the capabilities of commercial digital signal processing chips [4].

In our initial studies of the missing packet problem [5], we concluded that packet repetition improves speech quality relative to silence substitution. Informally, we judged the quality of speech derived from pitch waveform replication and pattern matching to be nearly equal to each other and higher than that produced by packet repetition.

C. The Scope of This Paper

The purpose of this paper is to describe a formal subjective test performed to obtain reliable measures of the merits of the various techniques. Before presenting the tests in detail in the following sections, we report our major findings.

Relative to silence substitution, packet repetition extends the maximum tolerable missing packet rate by a factor of approximately 2.5. Pitch waveform replication pro-

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O. J. Wasem is with the Department of Electrical Engineering and Computer Science, Massachusetts Institute of Technology, Cambridge, MA 02139.

D. J. Goodman, C. A. Dvorak, and H. G. Page are with AT&T Bell Laboratories, Holmdel, NJ 07733.

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vides a further two-to-one extension. For example, assume that the criterion of acceptability is a mean opinion score of 3.5 on a scale from one ("unsatisfactory") to five ("excellent"). Then, with silence substitution, the maximum missing packet rate is about two percent. With packet repetition, up to five percent missing packets are acceptable. Pitch waveform replication extends this limit to ten percent.

In the test that produced these conclusions, missing packets were distributed randomly over the duration of each presentation, containing, typically 9 s of speech. In practical networks, packet losses are bursty. There can be long periods with no missing packets, interspersed with intervals with high probabilities of packet loss. If the missing packets occur at random during these high-loss intervals, our results are indicators of speech quality during the periods that a network is under stress. If, however, during these intervals, the probability of consecutive missing packets is higher than when losses occur at random, waveform substitution will be less effective than our tests suggest [5].

All of the tests reported in this paper were performed with speech represented in a pulse code modulation (PCM) format. The effect of missing packets is different with other speech codes. We are currently investigating the impact of missing packets on networks that employ adaptive differential PCM [6].

In the next section, we provide brief descriptions of the waveform substitution techniques we tested. Readers may consult [5] for more details. Section III describes the subjective test, and Section IV presents the test results, which are interpreted in Section V.

II. Signal Processing Conditions

From several points of view, including network delay, throughput, and perceptual effects of lost packets [2], [7], it appears that 8–32 ms is a desirable range of speech packet durations. In our experiment, the packet duration was 16 ms, the packet length in an experimental communications system [8]. Each 16 ms packet contains 128 consecutive speech samples (8 kHz sampling rate). We performed all of the signal processing on a minicomputer (VAX/780) with 16 bit analog-to-digital and digital-to-analog converters (Digital Sound Corporation). Thus, the PCM quantizing noise was negligible and the audible speech impairments were those due to missing packets.

The simplest treatment of missing packets is silence substitution, sometimes called zero stuffing. It requires none of the three tasks (missing packet identification, speech storage, signal processing) listed in Section I. The next simplest is packet repetition in which the receiving terminal stores the contents of the most recently received packet. When one or more subsequent packets are missing, the terminal sends this stored information to the PCM decoder. Fig. 1 shows an 80 ms section of a speech waveform. There are two missing packets, each replaced by silence. In Fig. 2, each missing packet has been replaced by the previous packet.

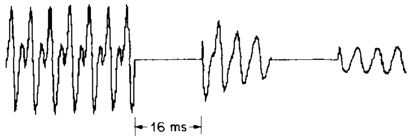

Fig. 1. Missing packets. An 80 ms section of a speech waveform divided into 16 ms packets. Two packets are missing and replaced by silence.

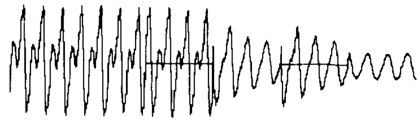

Fig. 2. Packet repetition. The missing packets in Fig. 1 have been replaced by the packets that preceded them.

A. Pattern Matching

1) One-Sided: To estimate the contents of a missing packet, the receiver uses, as a template, the last 4 ms (32 samples) of received speech. With this template, the receiver scans a 16 ms search window and detects the 4 ms segment that most resembles the template. The matching criterion is the minimum normalized absolute difference between template and candidate segment. The search window contains all of the speech in the packet preceding the most recent received packet. Therefore, the pattern-matching receiver stores the contents of the two speech packets (32 ms of speech) prior to the missing packet.

The substitution waveform is the 16 ms of speech immediately following the 4 ms segment that best matches the template. It is multiplied by a constant that makes its mean-square value equal to that of the speech in the most recent received packet. Along with other design properties, we selected the lengths of the template (4 ms) and the search window (16 ms) after evaluating waveform substitutions with a wide range of parameter values [5].

2) Two-Sided: In this case, the packet receiver finds two packet-length estimates of each missing packet, one derived from packets that precede the missing packet, and the other from packets that follow the missing packet. There are two templates, each of duration 2 ms (16 samples). The past template immediately precedes the missing speech and the future template immediately follows it. There are two search windows, each containing 8 ms of speech. The past search window is the half packet received just before the packet that precedes the missing packet. The future search window is the half packet following the packet received after the missing packet. Therefore, the two-sided pattern matching receiver stores the equivalent of three packets (48 ms = 384 samples) of speech.

The minimum absolute distance criterion leads to two 16 ms substitution segments, past and future. The final substitution packet is a weighted sum of the past and future segments. The weight profiles of the two packets are linear functions (that sum to unity) of sample position within the packet. The past substitution segment is an estimate of the speech at the beginning of the missing packet, and the future segment resembles the speech at the end of the missing packet. Therefore, we anticipated that the two-sided technique would be more accurate than other techniques when the missing packet contains a speech transition. On the other hand, two-sided estimation introduces substantially more delay than the other waveform substitution methods.

B. Pitch Waveform Replication

A positive peak detector searches for waveform maxima that occur at the beginning of pitch periods. It records the two time intervals separating the most recent three maxima. Correspondingly, a negative peak detector records two pitch-period estimates based on time intervals between waveform minima. Thus, at any time, there are four pitch-period estimates. Using these numbers, the pitch estimator decides whether the speech just prior to the missing packet is voiced. If the speech is detected as not voiced, the previous packet is taken as the substitution waveform. If the speech prior to the missing packet is deemed to be voiced, the pitch detector computes a single pitch period (T samples) and the substitution waveform consists of successive repetitions of the last T samples of received speech. The speech storage requirement for pitch waveform replication is on the order of the longest detectable pitch period. 20 ms (160 samples) is a reasonable memory budget for this purpose.

In Fig. 3, we have used pitch waveform replication to estimate the two packets missing in Fig. 1. Note that this technique has provided a smooth transition from prior packets to estimated packets. However, there is an abrupt phase discontinuity at the boundary of one of the estimated packets and the packet that follows it.

C. Signal Processing Complexity and Delay

Among the five packet replacement techniques, the number of signal processing operations required for waveform substitution are in the order listed in Table II: packet repetition less than one-sided pattern matching approximately equal to pitch waveform replication less than two-sided pattern matching. Prior to the experiment, it was our hypothesis that output speech quality would have a corresponding rank order.

The inherent delay of two-sided pattern matching is substantially higher than that of the other methods. It requires the reception of one and a half packets after the missing packet before it can derive a missing packet estimate. Because the other techniques obtain missing packet estimates from prior speech material, the only delay they introduce is the processing time to compute a new packet. With a commercial signal processor, this is a small fraction of the packet duration. On the other hand, missing packet detection could introduce a delay equal to the duration of one packet.

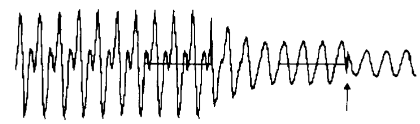

Fig. 3. Pitch waveform replication. The missing packets in Fig. 1 have been replaced by successive replications of the estimated pitch waveform. Discontinuities remain at the boundaries of estimated packets and the received packets that follow them.

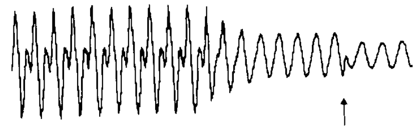

Fig. 4. The discontinuities in Fig. 3 have been smoothed by a packet merging technique.

D. Packet Merging

In our initial studies, we formed the impression that after waveform substitution, most of the residual speech impairment is due to discontinuities, such as the one indicated in Fig. 3, at the boundaries of received packets and substituted packets. To smooth these boundaries, we devised a "merging" technique in which the replacement waveforms are 2 ms longer (extended 1 ms at each end) than the missing packets. When the overlapping portions of the received and estimated packets are combined (by means of a weighted sum), the resultant waveform has relatively smooth transitions at packet boundaries. Fig. 4 shows the effect of the merging procedure on the discontinuity indicated in Fig. 3.

To determine the effectiveness of packet merging, we tested each replacement scheme (except silence substitution) with and without packet merging. Packet merging can be used with silence substitution. (It amounts to a gradual attenuation to zero of the end of the packet before a missing packet and a gradual amplification from zero of the following packet.) However, we view silence substitution as a baseline, no-cost approach to the missing packet problem, and therefore we tested it only without packet merging.

E. Missing Packets

There were four missing packet rates in the experiment: 2 percent, 4 percent, 8 percent, and 16 percent. Each presentation consisted of three sentences recited as a continuous sequence by one speaker. (The speech library is described in detail in Section III-A.) To produce a listening condition, a computer divided a speech sequence into 16 ms packets and distributed at random the locations of missing packets. For each listening condition, there was a different pattern of missing packets. The accuracy of an estimate depends on the speech content of the missing packet. We changed the missing packet locations from condition to condition in order to obtain average quality measures, rather than measures that are characteristic of specific missing packet distributions.

We estimate that the average speech activity of all of the material in the subjective test is on the order of 80 percent, which means that up to one-fifth of the "missing packets" contained no speech material. Our missing packet rates of 2, 4, 8, and 16 percent correspond, therefore, to rates of 1.6, 3.2, 6.4, and 12.8 percent in a packet communications system with speech activity detection.

F. Reference Conditions

In order to relate the data obtained in this test to the results of other tests, we included speech impaired by modulated white noise. Produced on the computer, the magnitude of each additive noise sample is in a fixed ratio to the magnitude to the corresponding speech sample. The polarity of the noise sample is positive or negative with probability 0.5. Subjectively, this noise is similar to that produced by a modulated noise reference unit [9].

While this noise is similar perceptually to the quantization noise of certain digital coders, it is quite different in character from the speech distortion due to missing packets. The modulated noise conditions were included in this test as a calibration of the range of listener responses rather than as reference distortions for the speech impairments of current interest. The signal-to-noise ratios of the four reference conditions were 6, 14, 22, and 30 dB.

G. Catalog of Conditions

In all, there are 40 speech processing conditions in the experiment, 36 packet communication conditions, and four reference noise conditions, all listed in Table I. The packet conditions comprise all combinations of four missing packet rates and nine packet replacement techniques: silence substitution without packet merging, packet repetition with and without packet merging, one-sided pattern matching with and without merging, two-side pattern matching with and without merging, and pitch waveform replication with and without merging.

III. Subjective Test Format

A. Speech Library

Each of the 40 signal processing conditions was tested with eight different speech sequences to create, in all, 320 test conditions. Each speech sequence consists of three consecutive Harvard sentences, recited as a continuous utterance by one speaker. There were two sets of sentences and four speakers, two male and two female. Each utterance lasted on the order of 9 s and was divided into 500–600 packets. The number of missing packets per presentation therefore ranged from about ten (two percent rate) to approximately 80. Of these, about one-fifth contained only silence. As stated earlier, the missing packet positions were different for each presentation, so that the entire test contained a comprehensive set of distortions.

The speech was processed by a 3.2 kHz low-pass filter, sampled at 8 kHz, and digitized with a 16 bit linear quantizer. Finally, the recorded speech levels were adjusted to produce equal active speech power (measured with a British Telecom SV6 speech voltmeter) over the set of eight utterances.

TABLE I
MEAN OPINION SCORES

Technique	Merging	Missing-Packet Ratio 0.02	0.04	0.08	0.16
silence substitution	no	3.41	2.62	1.85	1.27
packet repetition	no	4.01	3.67	2.81	2.22
	yes	4.14	3.66	2.95	2.39
one-sided pattern matching	no	4.22	3.92	3.23	2.45
	yes	4.33	3.76	3.40	2.51
pitch waveform replication	no	4.31	4.10	3.65	2.86
	yes	4.36	4.09	3.80	2.91
two-sided pattern matching	no	4.24	3.83	3.33	2.48
	yes	4.25	4.00	3.40	2.41
		Signal-to-Noise Ratio (dB) 30.0	22.0	14.0	6.0
noise reference		4.00	2.73	1.69	1.04

B. Test Environment

The subjects sat in partitioned cubicles in a room with acoustic absorbing material on the walls. The background noise level was 35 dBA. They listened to the test conditions over calibrated *G*-type telephone handsets. There was no sidetone and the noise in the handsets was on the order of 15 dBrnC. The listening level of the speech was 80 dBspl.

3 s of silence preceded each presentation, and the subjects were given 5 s to respond. They expressed an opinion by pushing one of five buttons labeled "Excellent," "Good," "Fair," "Poor," and "Unsatisfactory." Colored lights indicated when listeners should "WAIT," "LISTEN," or "VOTE."

At the beginning of a test, an experimenter read the test instructions to the subjects. Then the subjects responded to 12 practice conditions, chosen to span the full quality range of the test material. Next, the subjects responded to half of the experimental trials. After a break, the subjects completed the second half of the test.

C. Test Sessions

We administered the test twice on the same day, to a total of 21 female subjects, 11 in the morning and 10 in the afternoon. All of the subjects were housewives, and most of them had participated in other subjective tests. Their ages range from 20 to 60 years.

We presented the test conditions in one random order in the morning and in a different order in the afternoon.

In each of the 288 packet transmission conditions, the locations of missing packets were determined by a sequence of pseudorandom numbers.The sequence was chosen independently for each condition. For a given condition, there were two different patterns of missing packet locations: one in the morning session and another in the afternoon session.

IV. Test Results

In all, the test contained 6720 presentations (320 test conditions presented to 21 listeners). Except for ten missing votes, each presentation elicited a quality opinion on a scale from one to five, corresponding to the categories: unsatisfactory (1), poor (2), fair (3), good (4), excellent (5). To create a complete database, we replaced each missing vote with the mean value of the (nine or ten) votes received for this specific presentation.

A. Mean Opinion Scores

Among the causes of variability in listener opinions, we are primarily interested in the effects of signal processing and missing packet rate. We included other variables, such as the 576 patterns of missing packets and the eight utterances to obtain average measures based on a wide range of speech material and missing packet contexts. Our data analysis begins with the computation of two mean opinion scores for each communication condition. One is the mean opinion score taken over the 88 votes per condition (11 listeners, eight speakers) in the morning session. The other mean opinion score is obtained from the 80 votes per condition in the afternoon session.

In a preliminary statistical analysis, we found a close correspondence between the morning and afternoon mean opinion scores for the 40 test conditions. F tests confirmed homogeneity of variance; t tests of the difference between two means supported the null hypothesis that the votes obtained for a particular condition in the morning session were drawn from the same statistical distribution as votes obtained in the afternoon session. This led us to aggregate all 168 votes per condition in calculations of mean opinion scores. The results are displayed in Table I.

B. Reference Conditions

Modulated noise mean opinion scores in Table I are lower than those obtained for similar conditions in other tests [10]. This may be due to the fact that listeners in our specific test environment in general provide lower votes than other panels in other environments. Or it may be due to a difference between the computer-generated noise in our test and the noise generated by a modulated noise reference unit [9] used in previous tests.

C. Packet Merging

Except for silence substitution, we obtained two sets of quality measures for each of the packet replacement techniques. In one set of conditions, the packet merging method described in Section II-C was applied. In the other

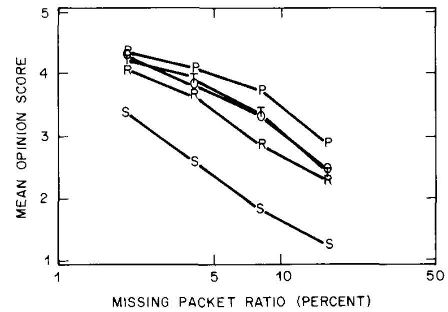

Fig. 5. Mean opinion score as a function of missing packet percentage for five packet replacement schemes. Results with and without packet merging have been aggregated. S: silence substitution, R: packet repetition, O: one-sided pattern matching, T: two-sided pattern matching, P: pitch waveform replication.

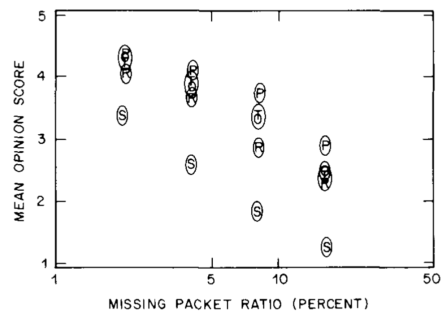

Fig. 6. The same information as Fig. 5 with ellipses enclosing data points that are not significantly different at the 0.01 level.

set, it was omitted. In Table I, the *merge* and *no merge* mean opinion scores are nearly equal for all relevant conditions. This suggests, and statistical tests confirm, that this particular packet merging technique does not contribute to speech quality. Because we cannot reject the null hypothesis that they come from the same distribution, we have aggregated in Fig. 5 the two mean opinion scores for each missing packet condition.

D. Packet Replacement Techniques

Fig. 5 displays a clear ranking of the effectiveness of the five packet replacement techniques. As expected, all of the waveform substitution methods provide much higher speech quality than silence substitution over the entire range of missing packet rates. While packet repetition is substantially better than silence substitution, it is less effective than the three methods that involved missing packet estimation. Unexpectedly, one-sided pattern matching and two-sided pattern matching are essentially equivalent. The most effective method is pitch waveform replication.

Fig. 6 contains the same information as Fig. 5 and also shows the results of statistical evaluation of differences between mean opinion scores. For a given missing packet ratio, the mean opinion scores of points that are in different ellipses are significantly different at a confidence level of 0.01. When two points are in the same ellipse, we cannot reject (at level 0.01) the hypothesis that they come from the same distribution. Fig. 6 shows that the ranking *pitch waveform replication better than packet repetition better than silence substitution* is statistically significant for all missing packet rates. One-sided and two-sided pattern matching are always equivalent to each other. With two percent of the packets missing, they are effectively as good as pitch waveform replication. With 16 percent missing, the two-sided scheme is no better than packet repetition.

V. Discussion

The goal of the subjective test was to evaluate the relative merits of several methods of reproducing speech signals with missing packets. It was our hypothesis that perceived speech quality would increase with the complexity of the signal processing technique. Table II contains two orderings of five approaches to estimating the contents of missing packets. The first list indicates relative complexity as measured by speech storage and signal processing requirements. The other list indicates relative quality. Note the important finding that pitch waveform replication, although less complicated than two-sided pattern matching and no more complicated than one-sided pattern matching, produces higher quality speech than both of them. Our test indicates that waveform segments corresponding to pitch intervals are better building blocks than packet-sized segments, at least for the 16 ms packets we evaluated.

We also hypothesized, on the basis of our own impressions prior to the experiment, that a technique for smoothing the boundaries between received and estimated packets would significantly enhance speech quality. This hypothesis was not confirmed by the experiment. Even though waveforms such as Fig. 4 appear smoother on an oscilloscope than waveforms similar to Fig. 3, naive listeners report no difference in their quality. A more effective technique of producing phase continuity at packet boundaries is called for.

The practical implication of the experiment is that waveform substitution can extend the acceptable packet loss level of a communication system. For a target mean opinion score of 3.5, packet repetition can extend the maximum packet loss rate from two percent (the limit with silence substitution) to five percent. It requires that the receiving terminal distinguish missing packets from silent intervals in the transmitted speech, and that the terminal store the contents of one prior received packet. Pitch waveform replication requires slightly more storage, on the order of 160 speech samples in all, and the implementation of a fairly simple pitch estimation algorithm. It extends the packet loss limit by a further factor of two to ten percent.

We remind readers that in our investigation, missing packets were distributed randomly in time. If, in anticipated application, there are other patterns of packet loss, the effectiveness of waveform substitution must be reevaluated. The number of consecutive missing packets has a strong influence on the estimation techniques. If, in practice, this number is higher than in a random distribution, speech quality will be lower than our tests indicate. Conversely, if the likelihood of consecutive missing packets can be reduced, higher quality will result.

TABLE II
Complexity, Delay, and Quality

increasing complexity	increasing delay	increasing quality
silence substitution	silence substitution	silence substitution
packet repetition	packet repetition	packet repetition
pitch waveform replication	pitch waveform replication	one-sided pattern matching
one-sided pattern matching	one-sided pattern matching	two-sided pattern matching
two-sided pattern matching	two-sided pattern matching	pitch waveform replication

Acknowledgment

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Ondria J. Wasem (S'84) received the B.S. and M.S. degrees in electrical engineering and computer science from the Massachusetts Institute of Technology, Cambridge, in 1986.

During the summer of 1983 she worked as an intern at AT&T Laboratories, Murray Hill, NJ, in the Robotics Principles Research Department. During the summer of 1984 and from June 1985 through January 1986, she worked again as an intern at AT&T Bell Laboratories, Holmdel, NJ, this time in the Communication Methods Research

Department. There she completed her Master's thesis on reconstructing missing packets of PCM and ADPCM encoded speech. She is currently in the Laboratory for Information and Decision Systems of the Department of Electrical Engineering and Computer Science (EECS), M.I.T., working on the Ph.D. degree under a National Science Foundation Fellowship.

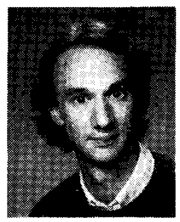
Mrs. Wasem chaired the M.I.T. Student Chapter of the IEEE and the M.I.T. EECS Student Faculty Committee from September 1983 through May 1984. She is also a member of Eta Kappa Nu, Tau Beta Pi, and Sigma Xi.

David J. Goodman received the Bachelor's degree from Rensselaer Polytechnic Institute, Troy, NY, in 1960, the Master's degree from New York University, New York, NY, in 1962, and the Doctorate degree from Imperial College, University of London, England, in 1967, all in electrical engineering.

Since 1967 he has been at AT&T Bell Laboratories. He his currently a Department Head in the Communications Systems Research Laboratory. He is also a Visiting Professor in the Department of Electrical Engineering, Imperial College, London, and in the Department of Electronics and Computer Science, University of Southampton, England. He has done research in several aspects of speech coding, digital signal processing, and communications networks. Recently, he has been studying short-range networks, including cellular mobile radio and indoor wireless communications.

Charles A. Dvorak (M'71) received the B.E.E., M.E.E., and Ph.D. degrees from the University of Delaware, Newark, in 1971, 1975, and 1978, respectively.

He has been with the Network Performance Center of AT&T Bell Laboratories, Holmdel, NJ, since 1982. Since 1984 he has managed the Performance Objectives Studies Group of the Network Performance Planning Department, whose activities include evaluating the voice transmission quality of new communications technologies and services. His responsibilities also include participation in domestic and international standards organizations. Prior to joining Bell Laboratories, he spent four years doing imaging systems evaluation research with Xerox at the J. C. Wilson Center for Technology, Rochester, NY.



Howard G. Page (S'79–M'82) received the B.S. degree in computer science from the University of Colorado, Boulder, in 1982 and the M.S. degree in computer science from Stevens Institute of Technology, Hoboken, NJ, in 1987.

He is currently a member of the Technical Staff at AT&T Bell Laboratories, Holmdel, NJ, in the Voice Performance Planning Group in the Network Performance Planning Department working on the development of a new generation of software tools for network performance modeling.