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ASYNCHRONOUS TIME MULTIPLEXING OF SPEECH SOURCES

by

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ABSTRACT

This report deals with a continued experimental and analytic program aimed at demonstrating the feasibility of asynchronous time multiplexing of multichannel speech. This is the third report in a series dealing with this subject and represents an updated version of the previous work. The major conclusions from the previous work have been repeated herein to make the report self-sufficient. The work reported here includes various experiments which were necessary to formulated a design plan for an asynchronously multiplexed system using a buffer.

Although much of the work herein is applicable to asynchronous operation in general, we tested the feasibility by using extremal sampling on the speech. Single-channel extremal sampling has been tested using both a computer simulation and an analog system. Computer simulation has been used to evaluate the multichannel aspects of the system and, finally, the results of the experimental program have been used to formulate a design plan for a system having the same bandwidth as a 12-channel PCM system.

It has been shown that, for a given bandwidth, the asynchronous multiplexing system can accommodate 3.6 times as many users as the synchronous PCM, with a speech quality which, to the casual listener, is indistinguishable from that resulting from the PCM system.

1. INTRODUCTION

If a number of speech sources are to be multiplexed on a common digital channel (or trunk), a standard method is to use synchronous time multiplexing of the pulse code modulation (PCM) samples. Synchronous operation refers to the fact that each connected source is given a channel slot within a frame; the connected speech source is then sampled regularly at the Nyquist sampling rate.

Synchronous operation has two disadvantages. 1) If the speech sources using the common facility are telephone communicators, each source uses his allotted part of the channel less than one-half of the time, because each is in a two-way conversation. 2) The regular-sampling technique used with synchronous multiplexing does not make efficient use of the redundancy present in all speech.

It is proposed that an appropriate irregular-sampling technique of each <u>active</u> speech source be used, and that each speech source seek access to the channel only when it has a sample to transmit. In other words, since the samples from each source occur irregularly, it is proposed that the samples from all the sources be allowed to enter the channel asynchronously without prior arrangement. Such asynchronous time multiplexing may be regarded as a form of "demand matching (Refs. 1, 2) which both introduces redundancy and smoothens the nonuniform flow of information from the sources.

In an ideal multichannel case, the use of demand matching would enable the channel to handle all the information from all sources. Further, all the information would be statistically stable enough so that the channel could operate at a uniform rate (without a buffer), even though the individual sources created information at a sporadic rate. In practice, one cannot realize this ideal--not all of the information can be transmitted (a situation usually called freezeout), and the information rate of the total ensemble of sources is not perfectly uniform, so that some buffering is required.

Work on an <u>irregular</u> sampling for speech, which permits realization of the asynchronous time multiplexing described here, is reported in Mathews (Ref. 3) and Spogen, et al. (Ref. 4). This sampling method has been termed "extremal coding" or "selective amplitude sampling" (SAS). We shall use the term "extremal sampling" throughout this report. The two major characteristics of extremal sampling are: 1) the samples occur irregularly; and 2) the average rate of occurrence is about one-fifth that of the Nyquist Sampling Rate.

This report represents a culmination of a series of work on this problem. References 5 and 6 present a description of the basic sampling method and an analysis of the buffer operation (amounting to a specialized queueing analysis) conducted by assuming that the rate of occurrence of samples among the multiplexed users is given by a Poisson distribution. The results of a computer Monte Carlo simulation of the buffering operation are described in Ref. 6.

The work is continued in Ref. 7, in which the primary attention was paid to demonstrating the feasibility of the asynchronous multiplexing method. This feasibility demonstration consisted essentially of two parts: (1) single-channel, extremally sampled voice segments were implemented both via a computer method and via an analog method; (2) a system computer simulation was run in which N-1 channels were simulated by a random generator, and one extremally sampled voice channel was used as the test channel.

Recent efforts have focused on performing various experiments needed to formulate a design plan for an asynchronously multiplexed system using a buffer. This report will be directed toward portraying this design plan. Another objective of this report will be to provide a brief review and description of the work and experiments which have been done to show the feasibility of the asynchronously multiplexed system. In many cases reference will be made to the two previous reports in this series (Refs. 5, 7).

In general, this series of work has been directed towards ascertaining the potential and feasibility of using asynchronous time multiplexing as a substitute for synchronous PCM over common trunk facilities. It has been assumed that the operation of the common trunk facility will be digital and synchronous, and that there will be data sources which must be multiplexed with the speech sources. Three methods have been used in doing this work: theoretical analysis, digital computer simulation, and analog experiments. A major reliance has been put on digital computer simulation.

In the following material we will first describe the basic system and its objectives. We will then quickly review those tests and analyses used to evaluate the feasibility of the multichannel aspects of the system. Following this we will describe those tests and analyses used to evaluate the single-channel extremal sampling method. All of this material is directed towards the design plan which follows in Section 5.

2. DESCRIPTION OF ASYNCHRONOUS SYSTEM

The basic objective of asynchronous time multiplexing is to make digital time division multiplexing more efficient when the sources are telephone talkers. The technique used exploits both the fact that a sampling method can be used which is less redundant than regular Nyquist sampling, and the fact that, on the average, telephone talkers use their facility less than half the time. (More than half the time is spent listening or in pauses.) The technique consists of making the sampling a function of the particular talker and dependent on his instantaneous output.

Figure 1 shows a block diagram of the transmitter side of the proposed asynchronous time multiplex system. Irregularly occurring analog samples (i. e., those which occur when a source is active) have access to an M-sample buffer after digital encoding. The irregular sampling allows: 1) a redundancy-reducing sampling; and 2) a more uniform flow of total samples from the N sources. It is assumed that a sample is removed from the buffer every τ seconds (synchronously).

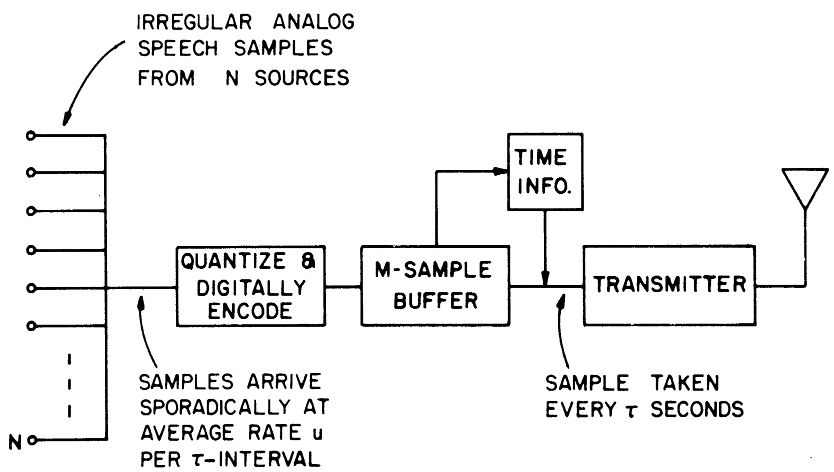


Fig. 1. Transmitter block diagram of asynchronous time multiplex system.

If the buffer is full, an arriving sample will be "frozen out" or rejected. If the buffer is empty throughout a transmission interval τ, the channel will be "idle" for this interval (data that do not require real-time transmission could be inserted in such idle time).

With any such irregular sampling the receiver must know the time position of the samples for its reconstruction. Consequently, if the samples are moved about, such as by buffering, one may have to send time information describing the time spent in the buffer. However, depending on the particular sampling plan and on the buffer length, the maximum buffer time may be short enough so that the jitter incurred by the buffer can be ignored. The design plan will describe both systems requiring time information and those not requiring time information.

In such an asynchronous system, the source of the irregularly occurring samples must be identified since one cannot a priori assign time slots to given sources. This need for source identification must be balanced against the lower number of samples to provide a net advantage.

The feasibility of such a system depends on: (1) an acceptably low total freezeout, (2) moderate buffer length, and (3) an acceptable technique for irregular sampling.

Much of the material of this report, in particular the buffer analysis, is valid irrespective of the particular sampling method used. Most of the assumptions used in conducting the system analysis would be reasonably true for many irregular sampling methods. However, whenever we simulated a system or used operational numbers to evaluate performance, we used the method of extremal sampling, as noted before (Refs. 3, 4). Another possible coding, for example, would be the use of a variable gain clipped speech (Ref. 8). There are undoubtedly other sampling plans which may be used successfully in asynchronous systems of the type being described here.

The basic idea in extremal sampling is shown in Fig. 2. As mentioned, basic work via a computer simulation was done by Mathews (Ref. 3). The essential attempt there was to ascertain the potential for extremal sampling and any permissible technique was used to improve the quality of the speech for a given rate. In the work of Ref. 4 an analog system was constructed and tested with the same sampling plan. In this first construction a basic technique referred to as "boxcarred" reconstruction of the samples was used to effect the analog system.

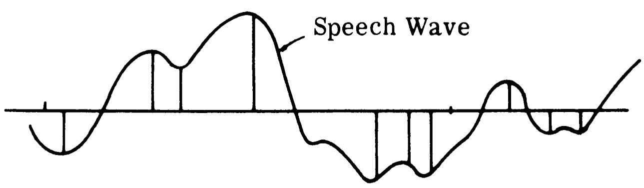


Fig. 2. Depiction of sampling for extremal sampling.

Since then our work reported in Refs. 5, 6 and 7 has contributed to knowledge about this sampling plan since we used the plan to effect the asynchronous system. In addition further work on this plan has been reported in Ref. 9.

3. TESTS AND ANALYSES TO DEMONSTRATE MULTICHANNEL FEASIBILITY ASPECTS

In this section we will briefly state those analyses, simulations and experiments which were used to demonstrate the multichannel aspects of the feasibility of this method. Many of the techniques and results of this section are somewhat irrespective of the particular sampling method and would hold for any sampling method in which the basic multichannel statistics hold. Section 4 discusses tests which are concerned with the single-channel feasibility demonstration, which in this case concerns extremal sampling.

3.1 Buffer Analysis

We will first treat the original buffer analysis, which assumed a Poisson distribution for the occurrence of samples from the multichannel source. Next we will describe the buffer simulation, which corroborated and extended this theoretical buffer work. Following this, multichannel simulation results will be stated. Finally, some recent experiments which complete and round out the multichannel feasibility aspects will be described; these consist of sample freezeout and sample jitter experiments.

An experimental computer analysis (Ref. 7) verified that extremal samples, from N parallel active sources, have a Poisson distribution of arrival of samples. Consequently, in the buffer analysis it was assumed that p(k, u), the probability distribution for k samples, from all N sources, occurring in a transmission time interval τ, is given by the Poisson relation

$$p(k,u) = e^{-u} \frac{u^k}{k!} , \tag{1}$$

where

u = average rate (taken <u>across</u> sources) of occurrence of samples <u>in time</u> τ.

It may be noted that <u>u</u> can be interpreted as the ratio of the average buffer <u>input rate</u> to the operating <u>output rate</u> (since samples are removed every τ seconds, if available). This probability distribution assumes that the time axis of the individual sources is divided into τ intervals, and that a Bernoulli model is used for the ensemble of sources.

Although the buffer action can be solved by use of general queueing theory, this is unnecessarily cumbersome for the task here. Buffer action is more conveniently analyzed by considering the probability distribution of $\underline{k}$ buffer stages being occupied along the time axis of the buffer (see Fig. 3). This "buffer occupancy" probability is periodic with respect to the τ-long transmission intervals. We will concentrate attention on two sets of equal-probability points: those points immediately preceding, and those points immediately following, a sample removal. An iteration procedure can be used to solve for the buffer occupancy probabilities, as follows:

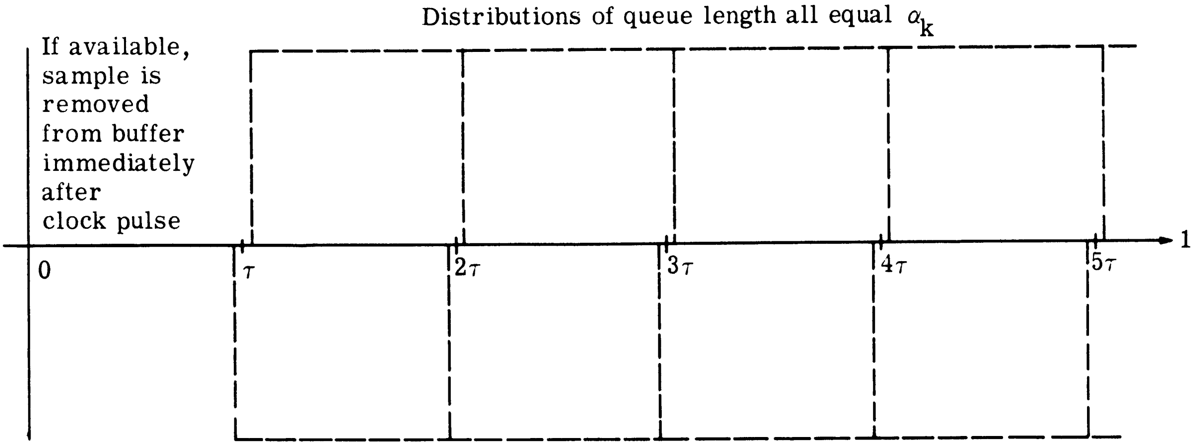


Fig. 3. Depiction of the probabilities α_k and β_k along a time axis.

Referring to Fig. 3, let

α_k = probability that exactly k buffer stages are occupied just after a sample pulse, i. e. , after one sample has been removed, if available;

β_k = probability that exactly k buffer stages are occupied just before a sample pulse;

M = length of buffer.

Since one sample is removed from the buffer at each sample pulse (if available), the following relations are valid:

$$\alpha_0 = \beta_0 + \beta_1$$

$$\alpha_k = \beta_{k+1}, \quad 1 \leq k < M-1 \ . \tag{2}$$

Equation 1 specified the probability of k samples occurring in a τ-interval; hence, using Eq. 1 in Eq. 2 yields

$$\beta_0 = e^{-u} \alpha_0$$

$$\beta_1 = u\, e^{-u} \alpha_0 + e^{-u} \alpha_1$$

$$\beta_2 = \frac{u^2}{2} e^{-u} \alpha_0 + u\, e^{-u} \alpha_1 + e^{-u} \alpha_2 \ . \tag{3}$$

The α_k's can be written in terms of α_0, as follows:

$$\beta_1 = \alpha_0 - \beta_0 = (1 - e^{-u})\, \alpha_0 \tag{4}$$

$$\alpha_1 = e^u [\beta_1 - u\, e^{-u} \alpha_0] = e^u\, [\alpha_0 - e^{-u} \alpha_0 - u\, e^{-u} \alpha_0] = \alpha_0 [e^u - u - 1]$$

$$\beta_2 = \frac{u^2}{2} e^{-u} \alpha_0 + u\, e^{-u} \alpha_1 + e^{-u} \alpha_2$$

$$\alpha_2 = e^u \left[\beta_2 - \frac{u^2}{2} e^{-u} \alpha_0 - u\, e^{-u} \alpha_1\right] = [e^u - u]\, \alpha_1 - \frac{u^2}{2} \alpha_0 \ .$$

The general relation can be written as an iteration,

$$\alpha_k = \left[e^u - u - \delta_{1,k}\right] \alpha_{k-1} - \sum_{n=2}^{k} \frac{u^n}{n!} \alpha_{k-n} \qquad 1 \leq k \leq M-1 \tag{5}$$

$$= 0 \qquad M \leq k$$

where

$\delta_{1,k}$ = the Kronecker delta.

If the buffer is M-sample long, then

$$\sum_{k=0}^{M-1} \alpha_k = 1 . \tag{6}$$

Using Eqs. 5 and 6, one can solve for the "probability of k buffer states being occupied."

3.2 Freezeout Fraction (FOF)

The foregoing probability of buffer occupancy permits calculation of the FOF (freezeout fraction), the expected fraction of total samples rejected by the buffer. The FOF can be written as

$$\text{FOF} = 1 - \frac{\text{av. no. of samples emerging from buffer each } \tau\text{-interval}}{\text{av. no. of samples emerging from sources each } \tau\text{-interval}} . \tag{7}$$

In terms of β_0 and u,

$$\text{FOF} = 1 - \frac{(1-\beta_0)}{u} = 1 - \frac{1-e^{-u}\alpha_0}{u} = \frac{u-1+e^{-u}\alpha_0}{u} . \tag{8}$$

The FOF as a function of u and the buffer capacity M is plotted in Fig. 4. From these curves a u of 0.8 appears sensible for FOF's below 0.1 percent with moderate M. With a u of 0.8 a channel will be "idle" 20 percent of the time on the average. As mentioned previously, nonreal time data could be inserted during this idle time.

Note that the curves of Fig. 4, and all the calculations above, assume that the mean number of sources is constant. We will see later that, for low numbers of multiplexed sources, the variance on the active users will be substantial. Therefore any system simulation for relatively low numbers must take this into account. Nevertheless, the values above are valid for any situation where the mean is constant.

Various other operating parameters can be calculated from the basic development of Eqs. 5 and 6, and more detail is available in Refs. 5 and 6.

3.3 Monte Carlo Simulation

A Monte Carlo simulation of a 16 long buffer was run on an IBM 709 computer to ascertain buffer aspects not amenable to analysis, and to corroborate the preceding analysis. The simulation effected a buffer where the input in each τ-interval was determined by a

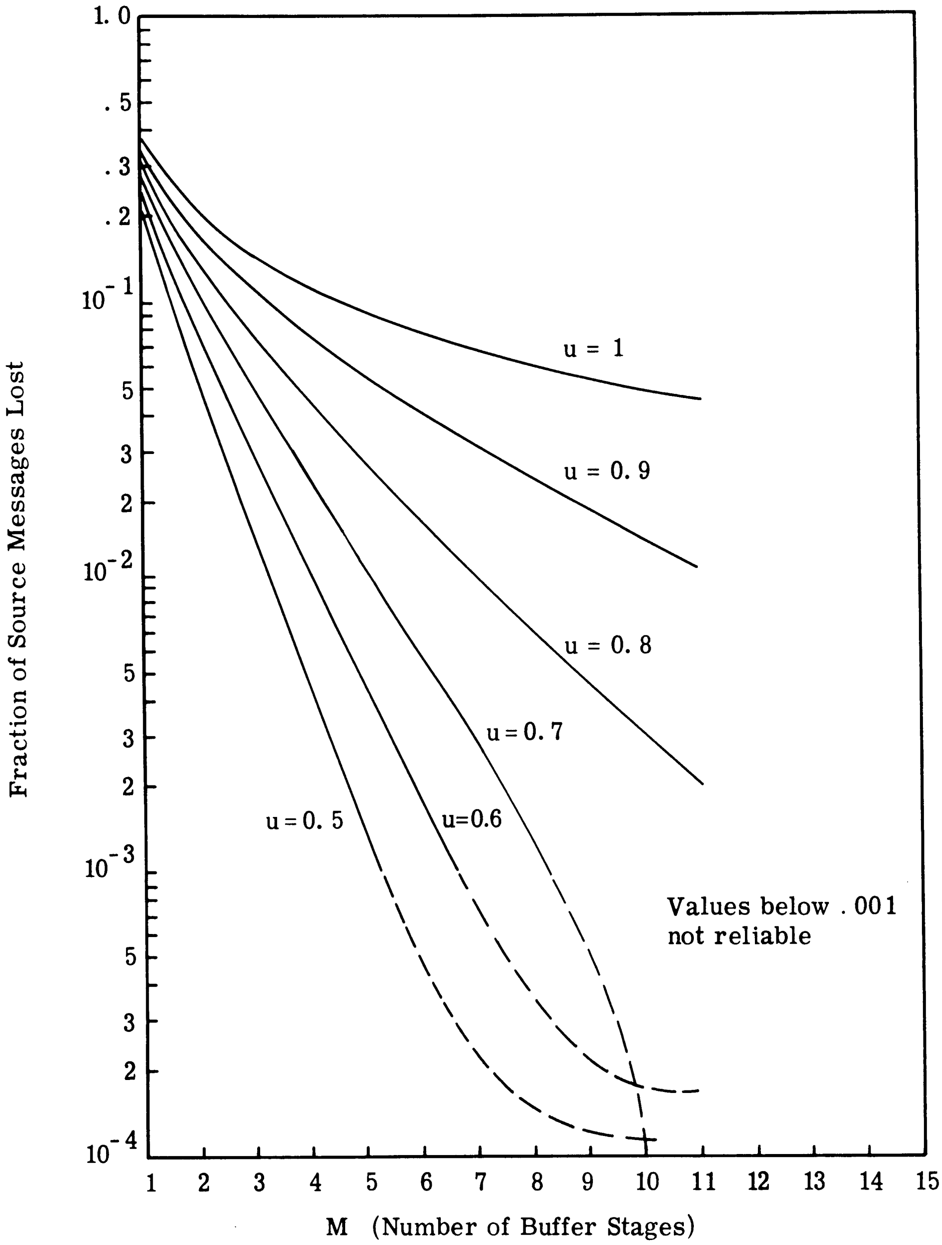


Fig. 4. Freezeout fraction as function of input rate u and storage capacity M.

Poisson random generator, and one sample, if available, was removed at the end of each τ-interval. The Poisson random generator was effected by suitably grouping random digits from an equally-likely random generator. Such an equally-likely generator is available as a standard computer subroutine.

Freezeout conditions were noted by allowing buffer content to exceed buffer length (16) in any τ-interval. After a given occupancy was recorded, samples were limited to 16, one sample was removed, and the next input was called for. The probability density of buffer occupancy (β_k) for u = 0.8 and 0.9 is shown in Fig. 5.

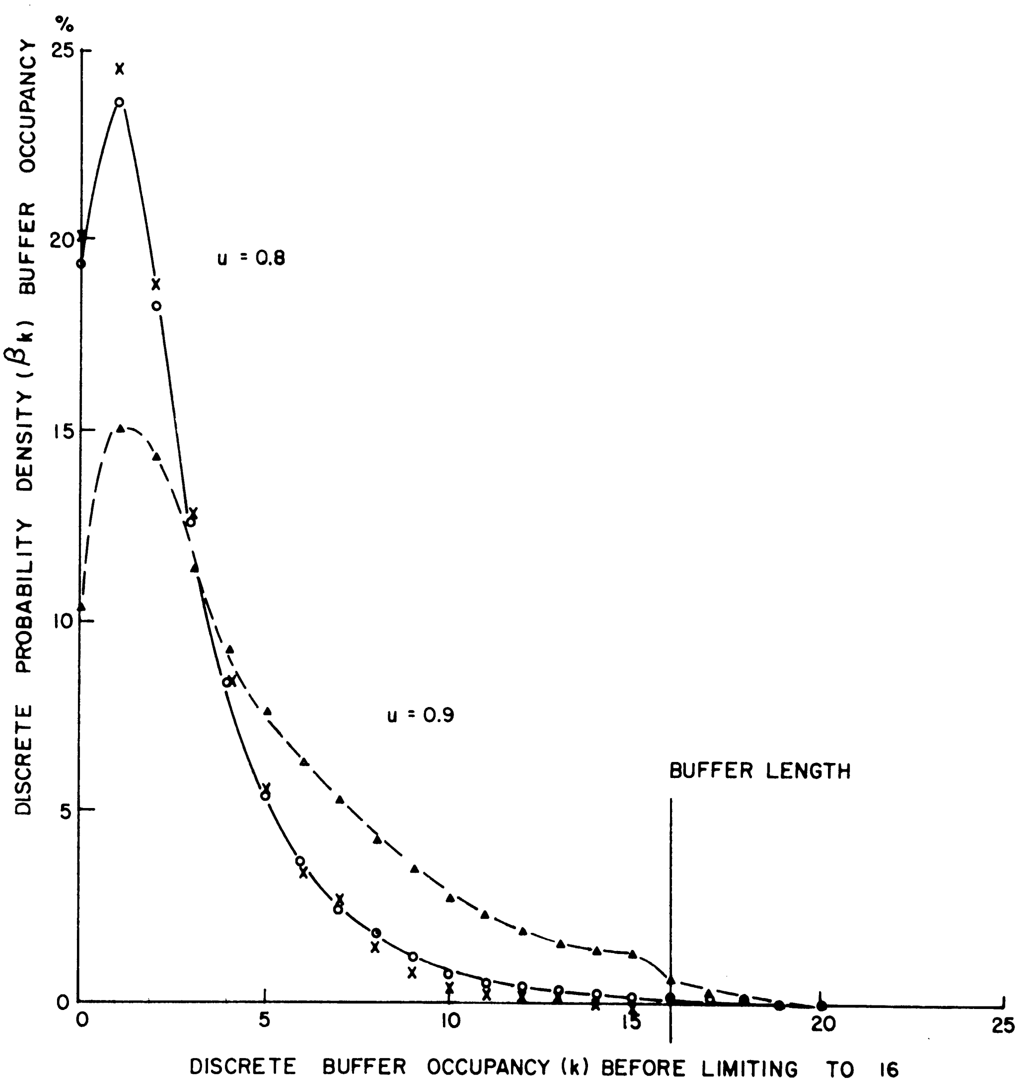


Fig. 5. Discrete probability density of buffer occupancy.

These data resulted from 20 runs of 1000 τ-intervals for each of four initial contents--0, 5, 10 and 15 samples. Since, the average, over 20 histograms of these results were essentially alike for the four different starting conditions, the results of these four were in turn averaged and these averages are plotted in Fig. 5. An important conclusion is that averages over 1000 intervals are "steady-state" averages for u = 0.8 or 0.9. This is true since the averaged experimental values for all four initial contents are close to the theoretical β_k values (shown by x's in Fig. 5). Note that Fig. 5 should be read as a histogram; the points are connected only to aid in separating the two conditions.

These curves show that the idle time is as expected, and that the most likely buffer occupancy is one sample. The computer data also indicated freezeout behavior as predicted by the analysis; however, the freezeout data are not statistically significant since they constitute a negligible percentage ($<$ 0.05 percent).

Another Monte Carlo experiment was run to determine the number of τ-intervals required for the buffer to recover to average occupancy after full load (where freezeout may have occurred). With initial contents of 15, the histogram of occupancy was noted for each of the first 120 τ-intervals. One thousand sample points are used for each histogram. Selected positions from these are shown in Fig. 6, where D is the number of τ-intervals after full load.

It can be seen that, for u = 0.8, the buffer recovers to average conditions at about 80 τ-intervals after full load. These curves depict the transient occupancy conditions after full load.

The Monte Carlo results below, along with the preceding analysis, show that a low freezeout can be obtained for a moderate buffer length if the input is Poisson distributed.

3.4 Poisson Test

Continuing the feasibility demonstrations, we ran an experiment to test the Poisson occurrence of samples from extremal sampled speech (Ref. 7). A speech sample was first digitally sampled at a 40-kc rate, and then all gaps over 50 milliseconds were removed to remove pauses. The resulting single speech sample was then divided into parallel (presumably independent) samples and the numbers of samples occurring within 40-kc τ-interval slots was determined. It was found that the probability for occurrence of extrema

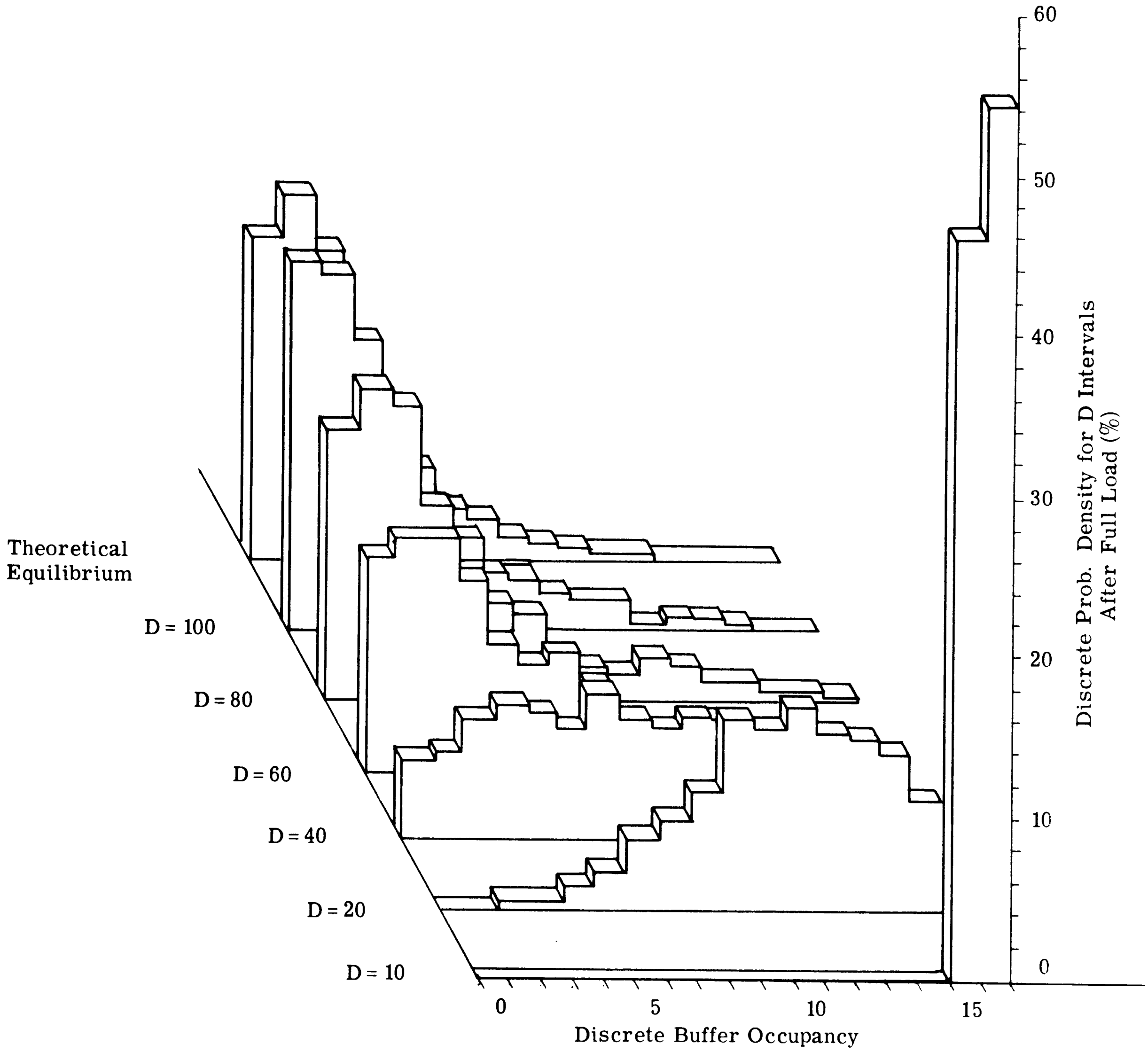


Fig. 6. Probability density of buffer occupancy after full load.

followed the Poisson distribution of Eq. 1 very accurately. Note that this verifies that the occurrence of extrema have a Poisson distribution when the number of "on" channels is constant. The effect of a varying number of active talkers is not expected to change appreciably the essentially Poisson distribution.

3. 5 System Simulation

The next feasibility experiment consisted of performing a full system simulation of an extremal sampled system. In this case a single voice channel was combined with the simulation of N-1 voice channels via random computer simulation (see Fig. 7). The simulation of the N-1 voice channel was carried out in the following manner. First, traffic statistics referring to pauses and length of talk bursts were used from telephone studies in the literature (Ref. 10). The computer was used to generate the various talk bursts and pauses. For each of the N-1 channels then, the digital computer randomly selects pause and talk burst intervals. Based on this information, the computer calculates the number of channels in simultaneous talk burst (N'). This resultant value of N' is then used to obtain the number of extrema in each clock pulse interval by using the Poisson distribution, which was verified in the previous experiments. These extrema are then sent into the buffer with the extrema from single voice sample. The buffer is operated in the usual manner (described above)--a sample, if available, is removed every τ-interval. If the buffer is full, extrema from the voice tape may be frozen out. This freezeout effect is of interest in the results of the actual tests discussed.

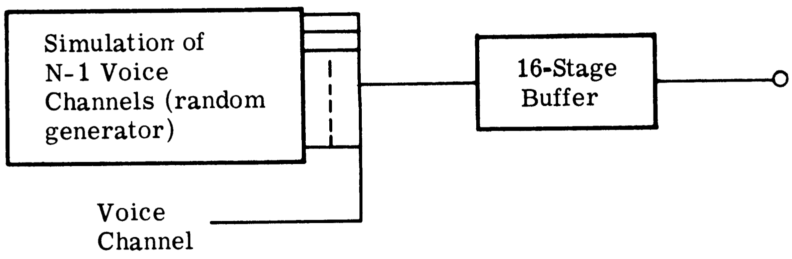


Fig. 7. Depiction of multichannel simulation.

A simulation experiment was run for a total system of 24 channels. The simulation of N-1 channels resulted in the following variation of N' (the number of channels in active talk burst) shown in Fig. 8.

First, this technique was used to verify the freezeout behavior under actual system operation. The test run was under the design conditions of specifying that a "just tolerable freezeout fraction of 1 percent" will be exceeded only 1 percent of the time. This specification called for a τ-interval equal to a 40-kc sample rate (p. 18, Ref. 7). For this reason the speech sample was also sampled at a 40-kc rate. As expected the freezeout incurred with these conservative parameters was unnoticeable.

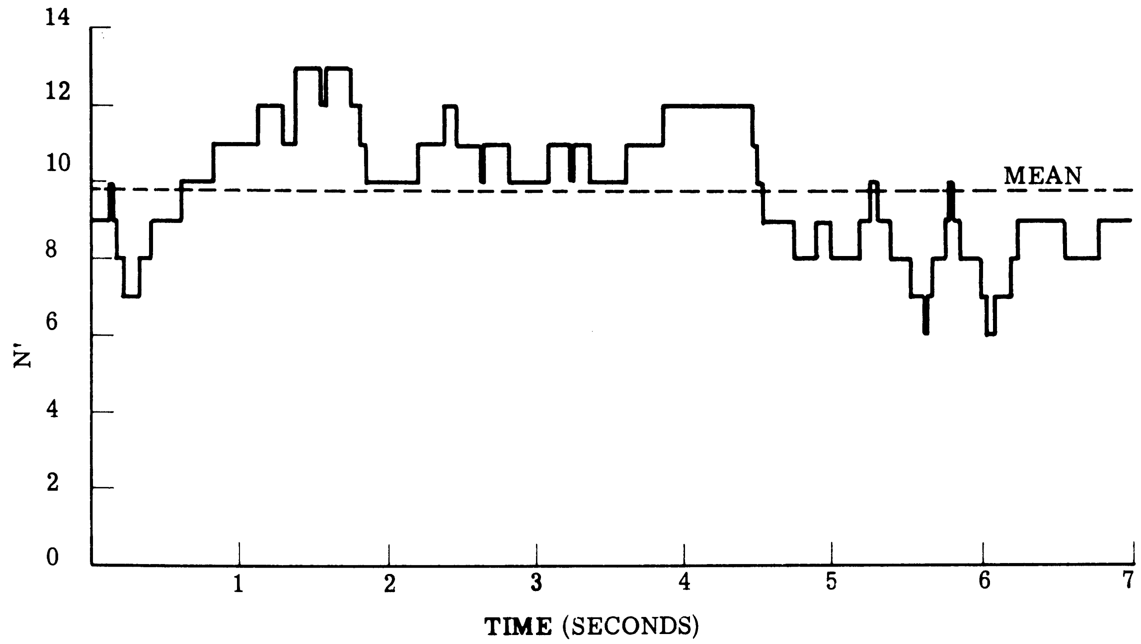


Fig. 8. Channels in simultaneous talk bursts as a function of time.

3.6 Freezeout Test

A test similar to that just described was used to evaluate the effect of freezeout. The system simulation as above (N-1 simulated channels with one voice channel) was used, but this time the N' was held constant. Above, for the 24-channel system, the mean N' was found to be 9.6. As noted, the resulting freezeout was not detectable. With the same τ-interval of 1/40,000, we now let N' take on the values 12, 14, 16, 18, and 20. This experiment is a good indicator of the result when N' seriously exceeds its design value.

Table I shows the resulting numbers of frozen out extrema (for the single voice segment) for the various N'. Note that N' is the number of channels in simultaneous talk burst which are competing with the test voice channel.

In this experiment it was found that the listening quality for N' up to 14 and 16 is very good and the effect of freezeout is barely noticeable. The effect of freezeout is noticeable in the quality for N' = 18 and the quality is seriously deteriorated for N' = 20. It should be noted however that the intelligibility for N' = 20 is still quite high.

Consequently, the overload begins to be serious when N' is about twice its mean value.

Table I. Freezeout test results.

N'	12 (Reference)	14	16	18	20
Total Extrema	20980	20860	20086	18694	16990
Average Extrema Rate (extrema/second)	2432.9	2419.0	2329.3	2167.9	1970.2
Difference from Reference		120	894	2286	3990
Percent Freezeout		0.57	4.26	10.90	19.02

3.7 Jitter Test

The remaining issue which we tested, concerning the multichannel aspects of asynchronous multiplexing, was the effect of jitter on the extremal sample locations. These jitter experiments were done on the digital computer.

In the jitter experiments that we conducted, the general idea was to sample at a rapid rate and then quantize the time positions of the resulting samples into larger and larger digital quantizations. Thus we sampled at a 40-kc rate, and quantized the resulting samples to first a 20-kc time position, then a 10-kc, and finally a 5-kc time position. Assuming that the 40-kc rate is rapid enough to be considered an analog position, this experiment is directly equivalent to operating an analog extremal sampler into a buffer which has the time quantizations corresponding to those stated. It should be noted that as the time quantization increases, one begins to get a small freezeout since more than one extrema can occur within a quantization time interval. For those cases where more than one sample existed in an interval, the computer program selected the first sample and discarded the remainder. This freezeout effect is shown in Table II, where the number of extrema frozen out is appreciable at the 5-kc time quantization. Thus, any results of this experiment are due to a combination of jitter and freezeout. The 40-kc data are also shown in Table II for comparison. The program used to find the extrema was the usual ± 5 constant threshold extremal coding program (Ref. 7, Appendix C).

Table II. Jitter test results.

	40 kc No. of Ext.	20 kc No. of Ext. Discarded	10 kc No. of Ext. Discarded	5 kc No. of Ext. Discarded
	20981	463	917	3411
No. Ext. Used	20981	20518	20064	17570
Approx. Rate/sec	2433. 1	2379. 4	2326. 7	2037. 5

In terms of the listening quality only the 5-kc test showed an appreciable amount of disagreeable noise. It is also noted that this is the only case that had an appreciable number of extrema discarded.

From the above tests it is concluded that a maximum jitter of at least plus or minus one-half the 10-kc rate (± 50 microseconds) is tolerable jitter.

Some work in this area has been reported in Ref. 9 in which it was found that the intelligibility remained high when the speech extrema were quantized to 6-kc intervals. This result was for speech which had been filtered with a passband of 300 to 3000 cycles per second and would indicate that the results given above may be somewhat conservative. However, it should also be noted that the experimental program here has been based on achieving high quality speech which is a more stringent requirement than intelligibility.

4. TEST RESULTS AND IMPLEMENTATION METHODS FOR EXTREMAL SAMPLING

As noted previously, all the tests that were used to experiment with asynchronous multiplexing used extremal sampling as the sampling plan. During the past year the aim has been to complete any computer feasibility experiments using extremal sampling, and to show the feasibility of performing the same functions with analog circuits. Here we will briefly review the computer implementation method and depict the test results. We will then depict the analog implementation method and some brief test results.

The general conclusion drawn from the previous work (Ref. 7) on computer-implemented extremal sampling was that telephone quality is obtainable at extrema rates in the neighborhood of 2100 extrema per second. In general, the digital computer is regarded as an experimental implementation tool which can show the effects of various implementation methods.

With the analog circuit the original purpose was to show the feasibility of reproducing the computer experiments with an analog circuit. In addition to this it was felt that a great many tests could be performed very rapidly with an analog circuit. The analog circuit used in the previous years' work (see Section 4. 2 of Ref. 7) has shown the feasibility of finding the extrema and reproducing an intelligible speech waveform. However, the output was a "boxcar" signal and contained the harsh quality associated with this type of signal. In addition, the original circuit required a high extrema rate in order to achieve acceptable quality. It was felt that this was, at least in part, due to the extreme difficulty in selecting the operating point in this circuit and it was felt that some modification of this circuit would be worthwhile. A discussion of the new circuit is given in Section 4. 2.

4. 1 Extremal Sampling Implemented by Computer

Two reasons for implementing extremal sampling with the computer are: 1) to find the proper parameters for extremal sampling, and 2) to use the computer for a system simulation of an asynchronous time multiplexed system. Although the computer implementation is described in Section 4. 1 of Ref. 10, we will briefly describe the pertinent aspects here.

In the computer method the analog speech segment is first sampled at a regular rate, and the A-D converted samples are recorded on digital tape. The sampling is done at a sufficiently rapid rate so that any extrema may be detected within the necessary accuracy (to be discussed later). The basic block diagram for the computer simulation is shown in Fig. 9. The computer then simulates both the transmitter and the receiver of the extremal sampling process. The transmitter function is simulated by first forming the "derivative" waveform, by subtracting each sample from the next sample. If a particular sample is an extrema (the derivative differs on each side), this sample is recorded and all intervening samples are deleted. The result is a sequence of samples which represent the extrema of the waveform.

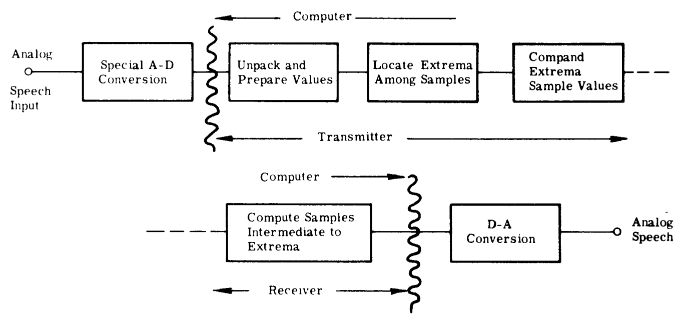


Fig. 9. Basic block diagram of computer implementation.

In determining extrema, we pass the derivative waveform (subtracted samples) through a "window" or threshold circuit which prevents minor bumps and bumps generated by noise from causing detection of extrema. Thus whenever the waveform goes through an extrema, the derivative must exceed the threshold on both sides of zero before an extrema will be detected. This threshold setting is one of the parameters which can be adjusted in single-channel coding. A very low threshold setting provides excellent quality; the quality deteriorates as the threshold level for this derivate crossing is raised.

By this means, then, the transmitter is effected via the computer method.

In the computer method the "receiver" is instrumented by placing the extrema at the proper place and fitting a waveform between the extrema samples. The simplest waveform to connect extrema samples would be a straight line (i. e. , hold a given sample until the next one occurs). Such an extrema receiver is called a "boxcar" receiver, and produces a good deal of distortion. Good quality speech can be effected, in a computer simulation, by using a connecting waveform such as the following:

$$F(x) = x^2(3 - 2x) \qquad 0 < x \leq 1 \tag{9}$$

where

$$F(0) = 0, \quad F(1) = 1.$$

This equation assumes that the previous extrema sample is the reference point. A receiver

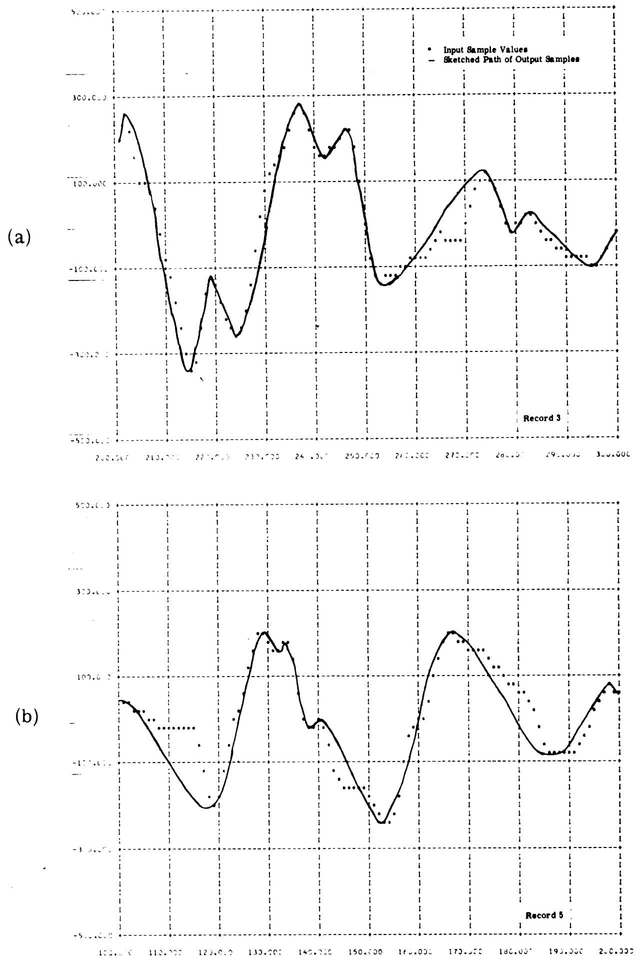


Fig. 10. Time charts of input and output for computer implementation (± 5 constant threshold).

implemented in this fashion is called a "waveform fitted" or "form fitted" receiving method. (See Section 4. 1 of Ref. 7.)

The general result of computer implementation of extremal sampling is indicated in Fig. 10. The dots on this figure represent the input samples as recorded by the computer. The sketched line is drawn between output samples. In other words, the samples are reconstructed between the extrema according to this sketched waveform.

4. 1. 1 Single-Channel Feasibility Tests Using the Computer-Implemented Speech. In the following we will describe a series of relatively independent tests which were used to ascertain various aspects and behavior of extremal sampled speech. All these tests were done on the computer with the computer-simulation method. The general objective was to determine the parameters and general behavior of extremal sampled speech.

The series of tests, its objective, and pertinent remarks are shown in Table III. In the following, the test results will be considered independently. Changes in speaker rate and slight changes in amplitude among the various tests cause somewhat varying extrema rates (for the same threshold) among the various tests.

Table III. Extremal sampling tests.

Purpose	Identification	Specific Objective	Remarks
Tests to Improve Quality of Extremal Sampling	Threshold Tests	Test for optimum threshold	Quality deteriorates fairly linearly with threshold increase
	Zero Modification	Remove a frequent distortion	Improvement not worth the cost in extrema rate
Application Tests	Multiple Speaker	Test sensitivity to different speakers	Process is not sensitive to different speakers
	Hand-set	Behavior with typical Army hand-set	No unusual effect noted
	Replay	Test difficulty of analog switching	Replay with form-fitted receiver is possible

4. 1. 2 Tests to Improve Quality of Extremal Sampling. Two tests were run to ascertain whether the quality of extremal sampling could be improved for a given rate; namely, 1) a threshold test, and 2) a zero modification test.

4. 1. 2. 1 Threshold Test. The threshold here refers to the threshold through which the derivative is passed in determining the extrema. As mentioned, the role of noise exclusion and small extrema exclusion is affected by this threshold setting. Although in the past we tried a threshold that varied with signal amplitude, it was found that the varying threshold curve gave no substantial improvement over a threshold which is constant. Therefore we reverted to the constant threshold (Section 4. 1. 1, Ref. 7).

With the constant threshold a series of tests was run varying the threshold between the limits of ± 3 and ± 10. [1] Previous tests (Ref. 7) were run at threshold settings of either 5 or 10. The objective here was to ascertain, on a smaller quantization basis, whether there was an optimum threshold between these limits. These tests were run and the results are shown in Table IV. The average signal-to-noise ratio was calculated by using:

$$\text{Average } (S/N)_{db} = 10 \log_{10} \frac{\sum_i f_{in}^2(t_i)}{\sum_i [f_{out}(t_i) - f_{in}(t_i)]^2}$$

where

f_{in} = value of input sample at time t_i

f_{out} = value of output sample at time t_i.

The tests were run for constant thresholds of 3 through 10 and the standard three sentences at a constant input level were used for all eight runs. These sentences were:

1. Mary Lou measures her father's nice long eyelashes once every nine years.
2. See which edge young owls could view.
3. Joe took father's shoebench out.

---

[1] Particular threshold setting values are only relative. The effect of a particular setting is partly dependent on the amplitude of the input speech signal.

Table IV. Threshold test results.

Threshold	± 3	± 4	± 5	± 6	± 7	± 8	± 9	± 10
Total Extrema	22218	19051	18051	16555	15500	13778	13282	12709
Extrema Rate	2469. 2	2117. 2	2006. 1	1839. 8	1722. 6	1531. 2	1476. 1	1412. 4
Average of S/N	10. 18	9. 45	9. 17	8. 73	8. 26	7. 55	7. 37	7. 19

It should be noted that the extrema found and the rate for a ± 5 constant threshold are somewhat lower than those reported previously (Ref. 7). In this case the difference may be accounted for entirely by a difference in input amplitude between these tests and the previous tests. The listening quality for these tests is not noticeably reduced until the threshold reaches 7 or 8. The intelligibility is still quite good for the ± 10 threshold.

The dependence of the number of extrema and the average value of the calculated signal-to-noise on the threshold setting is easily seen from Table IV. Based on listening tests, the quality degrades gradually and becomes objectionable only after reaching a threshold level of 7 or 8.

From the series of tests it appears that, for constant volume speech, there is a fairly linear relation between threshold setting and quality. Extrema rates in the neighborhood of 2,000 result in very good quality speech. Upon further comparison of these results with previous results (threshold setting was the same), it is clear that the extrema rate and hence the quality are dependent on the "signal amplitude-to-threshold setting" relation. Consequently this problem must be dealt with in any system. One method of handling this issue in a system would be to use an automatic volume control in each channel of an extremal sampled system. This would require an amplifier in each channel; since an extremal coder is already required in each channel, the addition of an amplifier may not be prohibitive.

Another possibility to account for varying volume is to let the sample amplitude be quantized into a sufficient number of bits (presumably 6) and compand these in the most optimum way to provide for varying volumes. This would have the advantage of requiring equipment only in the central (multiplexer) function, as does PCM. If possible, this would be the desired alternative.

4.1.2.2 Zero-Modification Tests. After examination of the time waveform of computer-implemented sampled speech, it was found that the greatest waveform distortion appeared when the waveform stayed near zero for some period of time. This effect is seen in Fig. 10(a) near the 270 point and in Fig. 10(b) near the 110 - 120 points. This phenomenon is expected, considering the way extremal sampled speech is constructed. Since this phenomenon occurs frequently in conversational speech, it constitutes a potential major source of improvement for a slight increase in sample rate.

The purpose in performing these tests, then, was to determine whether an improvement in the quality-rate ratio could be achieved by forcing the output to follow the input more closely at points where the signal remains near zero for many samples. This was accomplished by modifying the computer program so that an artificial extrema is added if the slope of the waveform stays near zero for a varying number of samples. The test for this was placed on the slope or derivative waveform. Table V shows the results as a function of the number of samples (NPTS) during which a slope must remain inside the threshold before the artificial extrema is added.

Table V. Zero modification results.

	NPTS = 8	NPTS = 4	Previous Test
Threshold	± 10	± 10	± 10
Total Extrema	21834	22765	21145
Average Extrema Rate	2712.1	2827.9	2626.5
Average of S/N	7.65	8.15	7.37

As expected the number of extrema found increases as NPTS is decreased and the S/N increases. It is apparent that one must sacrifice extremal rate for this increase in the S/N and it appears that unless the increase in quality (or S/N) is startling this is probably not worthwhile.

From listening tests it was found that the improvement in quality does not warrant the increased sample rate. Consequently, we conclude that the speech waveform is not very sensitive to this particular type of distortion, and any objectionable characteristics of extremal sampled speech (at the lower extremal sampling rates) are caused by factors other than this particular waveform distortion.

The above tests, then, showed that so far as the threshold is concerned there is no sharp nonlinearity in the quality versus threshold setting. Consequently, one can get the quality desired by continuing to narrow the threshold . Concerning the waveform distortion near zero, it is found that the improvement resulting from correcting this is not worth the increase in extrema rate.

4. 1. 3 Application Tests. We refer here to three tests (see Table III): 1) a multiple-speaker test to explore the sensitivity of extremal sampling to different speakers; 2) a test using an Army TA-43/PT hand-set to assure that no unusual behavior is encountered, and 3) a replay test to explore the effect of multiple, sequential, extremal samplings.

4. 1. 3. 1 Multiple-Speaker Tests. The objective of the multiple-speaker tests was to determine the sensitivity of the extremal sampling process to different speakers. All of the tests run previously used as input speech a master tape recorded by a trained speaker. In addition to having a single speaker, this particular speaker spoke in a radio announcer fashion which led to unnaturally high extrema rates. It was also found that the peak-to-RMS value of this recording is somewhat lower than for the same three sentences recorded by an untrained speaker.

In order to determine the spread in total extrema and extrema rates which might be expected, a series of tests was run using five untrained speakers. The results of these tests are shown in Table VI. For this series of experiments the threshold was adjusted to give a good quality speech output. The same three sentences noted before were used. It should be noted that, for all five of these speakers, the number of records necessary is greater than for the trained speaker used for the experiment described in the previous section.

Previous extrema rates referred to the overall rate, which includes the pauses between the words and sentences in the narrative speech. Table VI shows also the "talk burst" time and "talk burst" rate. This talk burst time is obtained by removing pauses between sentences and between words. The talk burst time for these samples was calculated by recording the output with a Sanborn recorder and from this display determining the actual time of active speech. Using this talk burst time, we could then calculate the talk burst rate.

Table VI. Multiple-speaker results.

Subject	AR	LT	JV	WR	TB
Total Extrema	16909	16167	16189	11764	13009
Extrema Rate (extrema/sec)	1326. 5	1540. 0	1349. 4	1120. 6	1156. 6
Overall S/N	11. 17	11. 39	11. 21	11. 37	11. 03
Time (sec)	11. 9	9. 2	10. 9	9. 3	10. 0
Talk Burst Time (sec)	9. 6	7. 5	8. 8	8. 2	9. 0
Talk Burst Extrema Rate (extrema/sec)	1750	2150	1830	1420	1450

The actual time for the trained speaker (which was used in the previous tests and tables) was 7. 9 seconds, with the talk burst time of about 7. 5 seconds. This would lead to a "talk burst" extrema rate of about 2660 extrema per second. From the above data it can be seen that the overall time for the five speakers ranges from 1. 3 to 4 seconds longer for the same three sentences. The "talk burst" time ranges from 0 to 2. 1 seconds longer so that the untrained speakers had much longer pauses than the trained speaker. The extrema talk burst rates are considerably lower for the untrained speakers and range from 1420 to 2150 extrema per second. It seems apparent that the rate for the trained speaker is indeed high.

The number of tests run is insufficient to determine the spread for general speakers. However, based on these five speakers and the previous data it would appear that an average talk-burst rate in the neighborhood of 2000 samples per second would be an accurate estimate. The talk-burst rates for the two speakers WR and TB are somewhat low while that for the trained speaker is high. The quality for all of the tests of this series was quite high; it can best be described as telephone quality or better.

4. 1. 3. 2 Hand-set Tests. Another application test using the computer implemented method concerned using a typical Army source and ascertaining whether any unusual behavior occurs using extremal sampling with this source.

A Telephone Set TA-43/PT was supplied by USAEL to be used as an input transducer for the extremal coding simulation. The purpose of these tests was to determine the effect of using a standard hand-set in place of a microphone. Two speakers (untrained) were used for this experiment and the threshold was set at ± 5. The results of these experiments are shown in Table VII. The extrema rates are comparable with the rates found previously for ± 5 constant threshold and the quality was very high. The first listening tests were made using an amplifier and speaker rather than the hand-set receiver, and the quality at the speaker of the extremal sampled output was higher than the standard output of the hand-set receiver. It would appear then that the quality when using this type of hand-set is limited by the receiver element in the hand-set. A listening test was made using a hand-set to hand-set link to compare the quality with and without the extremal sampler. It was found that the difference between the quality was slight and amounted to a difference in background noise.

Table VII. Hand-set results.

Subject	AR	TB
Total Extrema	20950	24002
Extrema Rate	1995.6	2133.8
Average of S/N	9.95	7.80

It appears that there is no unusual effect when such a hand-set is used as a source for extremal sampling--rather than the quality microphones which we have used in the past. Since one is already limited in quality by the hand-set receiver, extremal sampling is not the quality limitation.

4.1.3.3 Replay Experiment. In another applications test we ran one replay experiment with the computer system. The reason for this is that, in a given application, it may be necessary to pass through an analog switchboard. In such a case it may be necessary to restore the signal to analog form for switching, and then resample for further transmission. Therefore, it is of interest to determine the effect of repeated extremal sampling of speech.

Generally speaking with a technique such as the form-fitted computer implementation, where the extrema positions are retained one should not experience great difficulty in replaying. We will see later, when looking at the analog implementation, that things are somewhat different there. In any case, for the computer-implemented type here the source of difficulty in replay would be the accumulated buffer jitter. That is, the samples used in an asynchronous system are always jittered somewhat because of the τ-interval quantization. This accumulated jitter will be the eventual source of degradation for replayed speech.

As a start in ascertaining the effect of replay, a single test was run in which the output of the ± 5 test from the threshold tests (Table IV) was used as the input to the extremal sampling simulation program. The resulting data are shown in Table VIII. For the ± 5 test the total number of extrema was 18051 and therefore approximately 3000 extrema were lost in the replay process. This loss in extrema is possibly due in part to a variation in amplitude between the two inputs to the A-D equipment. The quality has not been seriously affected by the replay. We feel this test is somewhat inconclusive; however it does indicate that the replay with a computer form-fitted implementation will not cause major problems. This should be tested further with a closer control on the signal levels in order to obtain more exact results.

Table VIII. Replay experiment results.

Total Extrema	15028
Extrema Rate	1824. 4
Overall S/N	11. 85

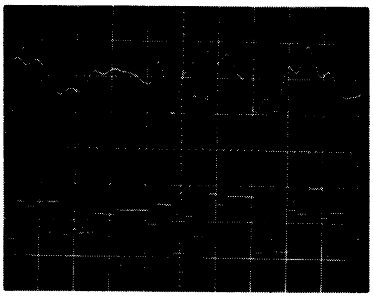
4. 2 Analog Implementation of Extremal Sampling

In addition to the computer implementation and the associated feasibility tests, we have developed an analog implementation for the purposes of both experimenting and of demonstrating feasibility of construction. We will first review the previous single-channel method, which was developed in Ref. 7, and then describe a more recent dual-channel prototype method.

4. 2. 1 Single-Channel Method. The simplest method of extremal sampling reception was the "boxcar" output. Extrema were held at the value detected until the next extrema occurred at which time the new value was held (see Fig. 11). This "boxcar" output was then filtered. The final output was quite objectionable and could best be described as having a significant amount of rasp.

Audio

Boxcar

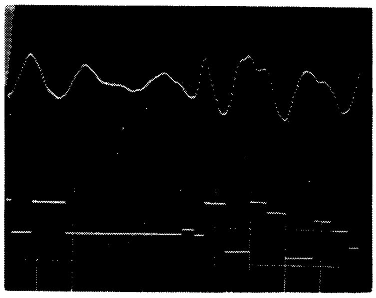


1 ms/cm

a) Small Threshold

Audio

Boxcar



1 ms/cm

b) Large Threshold

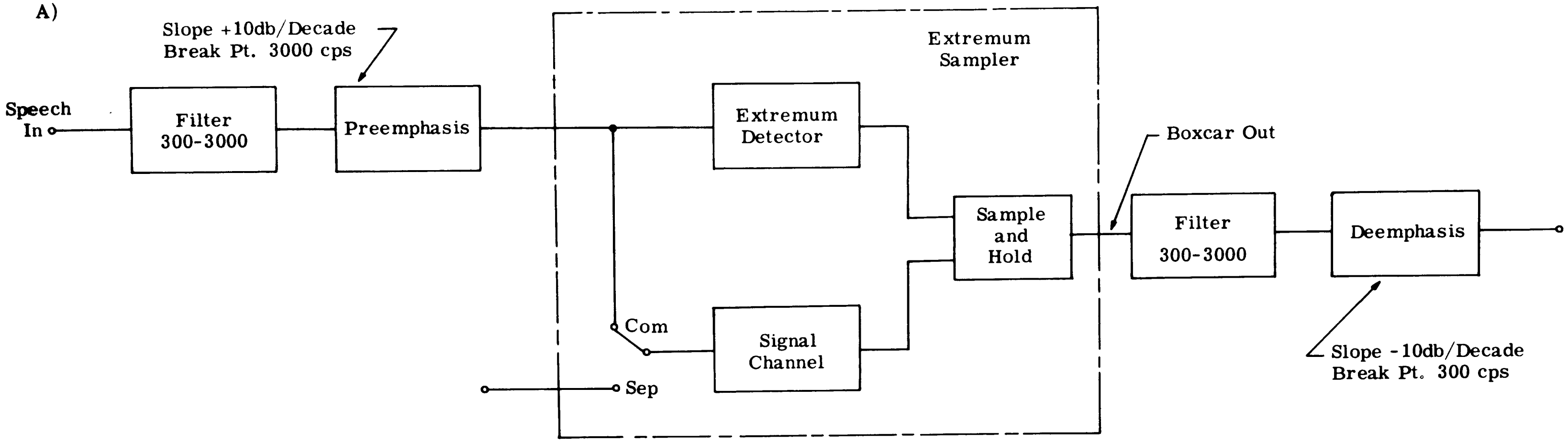
Fig. 11. Extremum sampling.

One source of the rasp is the harmonic distortion that remains within the speech passband. In an effort to reduce the distortion, certain methods of pre-emphasis and de-emphasis were tried.

One such method is shown in Fig. 12(a). The high end of the passband is emphasized using a 6 db-per-octave slope. The operating dynamic range of the extremal sampler is not affected since the speech spectrum rolls off at about a 6 db-per-octave slope with a turnover at 800 cps. The de-emphasis network which equalizes the signal now attenuates the harmonics with respect to their previous amplitude. However, since the highs are emphasized at the input of the extremum detector, the sample rate is increased for a given threshold setting and any increase in quality was more than offset by the increased sample rate. That is to say, the quality for a given sample rate is not improved and the low end of the speech spectrum is especially deteriorated.

In an effort to lower the sample rate but derive the benefits of the above mentioned, the method shown in Fig. 12(b) was tried. As expected, however, because of the phase difference between the extremum detector and the signal channel, there was a null at the lower end accompanied by amplitude distortion above and below the null. In order to compensate partially for this, at least above the null, a 20 db-per-decade de-emphasis network was used. The quality of the resulting speech was unsatisfactory.

A)



B)

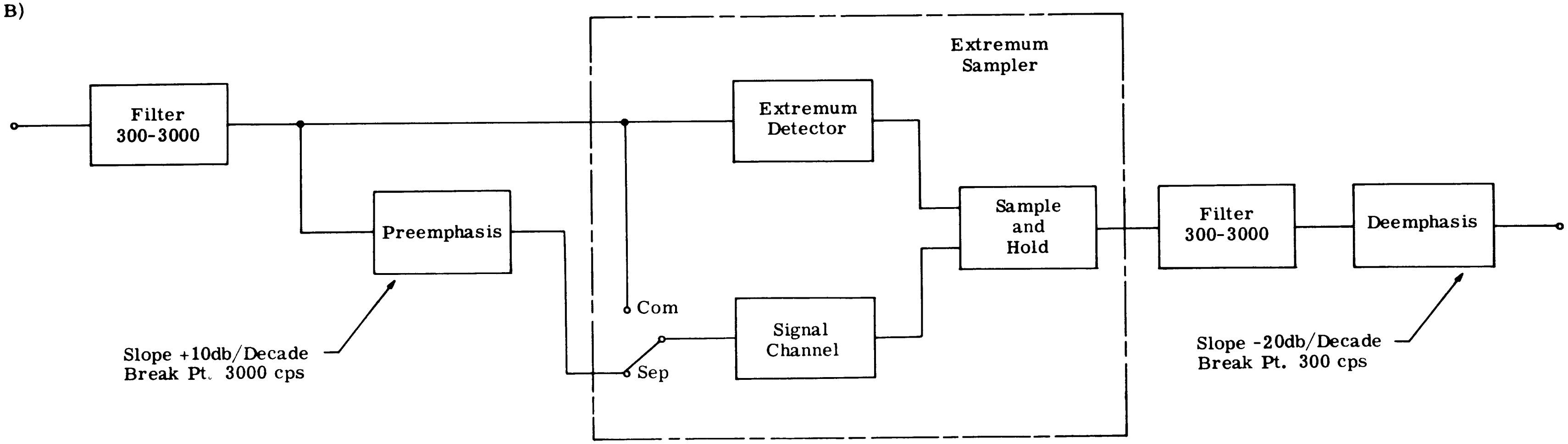


Fig. 12. Single-channel block diagrams

4.2.2 Dual Channel Method. A method which is successful in reducing the harmonic distortion is shown in Fig. 13. The particular case shown indicates that the third harmonic distortion of each passband falls outside that passband. After the third harmonic is filtered out of each band they are added back together. There is some increase in overall sample rate for a given threshold setting, but we believe that the increase in quality (from that of boxcarred) far outweighs this increase making it possible to use a much lower sample rate while maintaining an acceptable quality.

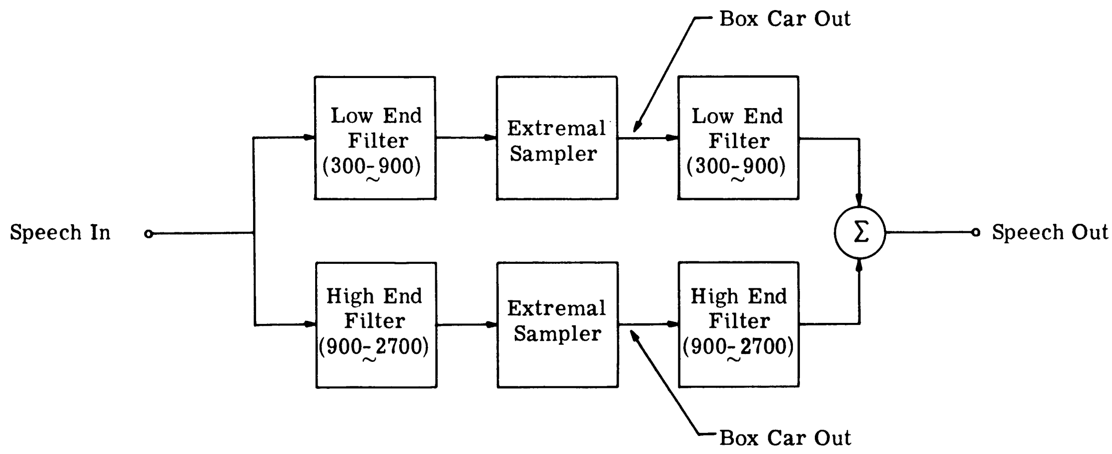


Fig. 13. Block diagram of dual channel method.

The dual-channel method has a further added advantage. When the threshold is increased in a single-channel case, eventually a level will be reached where large pieces of audio will drop out (see Fig. 11). This is true because the threshold is placed symmetrically about the derivative of the signal used in detecting extrema. Thus, with large thresholds, the low frequencies, where few or no high frequency components are present, are not sampled or are attenuated. In the dual channel method,the low end extremal sampler can be set with a low threshold to assure no drop out, more or less independently from the overall sample rate.

The effectiveness of the dual channel method rests on balancing the reduced third harmonic distortion plus the individual threshold control against the addition of one bit per sample and the need for two extremal samplers per channel. This will be affected both by the degree of quality increase and the possibility of using microelectronics to diminish the complexity problem.

Table IX illustrates the effectiveness of being able to select threshold settings independently in the dual channel method. Acceptable quality should not be confused with intelligibility; all runs shown were highly intelligible. Further, the judgment is a subjective one and based on telephone quality. Table IX refers to some preliminary listening tests which were made with various threshold settings and illustrates this advantage. In runs 1 through 3 the threshold was steadily increased. Runs 1 and 2 sounded normal in average intensity and were quite satisfactory. On run 3 there was a decided booming effect as lows dropped out in large pieces and bursts of lows reappeared. On runs 4 and 5 this booming effect was not present since, even though the overall sample rate was low (run 5 was lower than run 3), the low band threshold was maintained constant at a low value.

Table IX. Sample rates for various test runs.

Approximate Average Low End (Samples/Sec)		Approximate Average High End (Samples/Sec)	Approximate Average Overall (Samples/Sec)	Quality
Same Threshold Settings on Both Channels	400	1920	2320	Acceptable
	390	1700	2090	Acceptable
	300	1220	1520	Not Acceptable
	460	1200	1660	Acceptable
	460	950	1410	Acceptable

It has been found that the ability to vary the two threshold settings independently is an advantage in transmitting extremal sampled speech. Although it is quite straightforward to build two encoders and receivers as shown, the above tests used a single encoder-receiver and a dual-channel tape recorder.

Based on subjective listening tests, we believe that this dual channel system affords telephone quality speech at rates of about 1500 extrema per second.

4.2.2.1 Instrumentation. The instrumentation of the extremal samplers represents a unique method of obtaining and deriving the extremum detection. The extremal sampler basically consists of two parts (see Figs. 14 and 15); namely, an extremum detector and a signal channel. Since the two parts operate independently, one can sample with a steady-state sine wave at twice the highest frequency to give PCM. Thus, this circuit is useful for immediate comparison to PCM.

In order to detect extrema the signal is first differentiated using a standard RC differentiator. Any zero-crossing phase error is kept small across the speech spectrum. The differentiated output is amplifier and emitter followered to a high impedance driver which drives a half-wave bridge. The high impedance driver in conjunction with the bridge are used to detect zero crossings accurately and at the same time provide a convenient method of applying a threshold about the derivative. The high output impedance of the bridge driver (a few megohms) reduces error caused by the nonlinearity of the diodes close to the knee; in other words, acts as a good current driver.

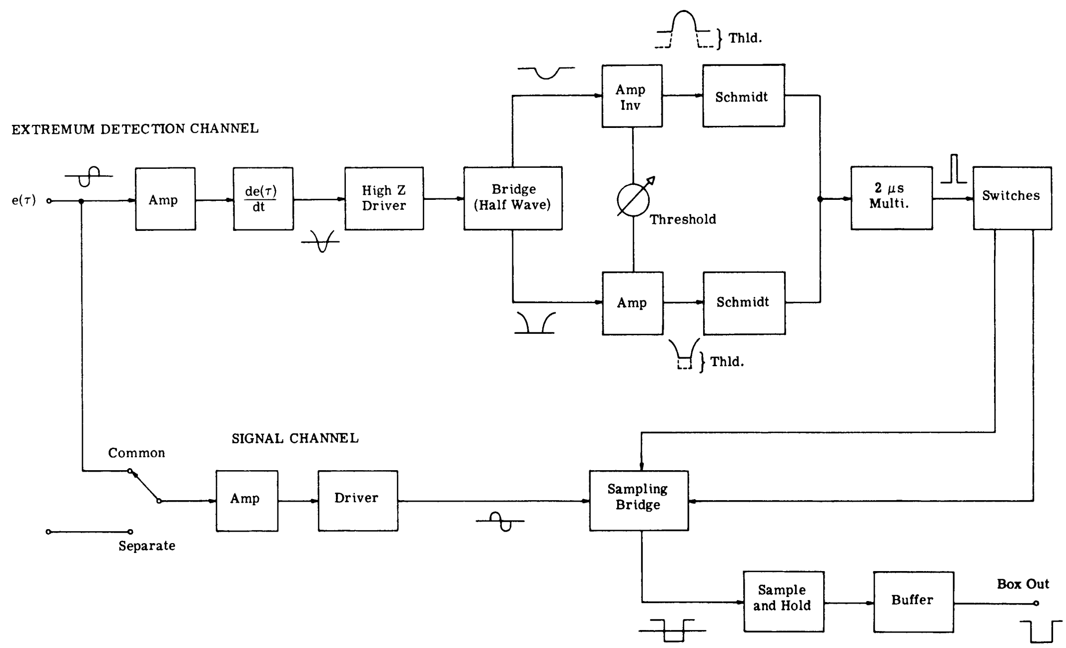


Fig. 14. Extremal sampler block diagram.

To accomplish the current driver, two matched complementary transistors Q4 and Q5 were stacked collector-to-collector. Operating point drift is further reduced by making the dc generator impedance at the bases high and the ac generator impedances at the

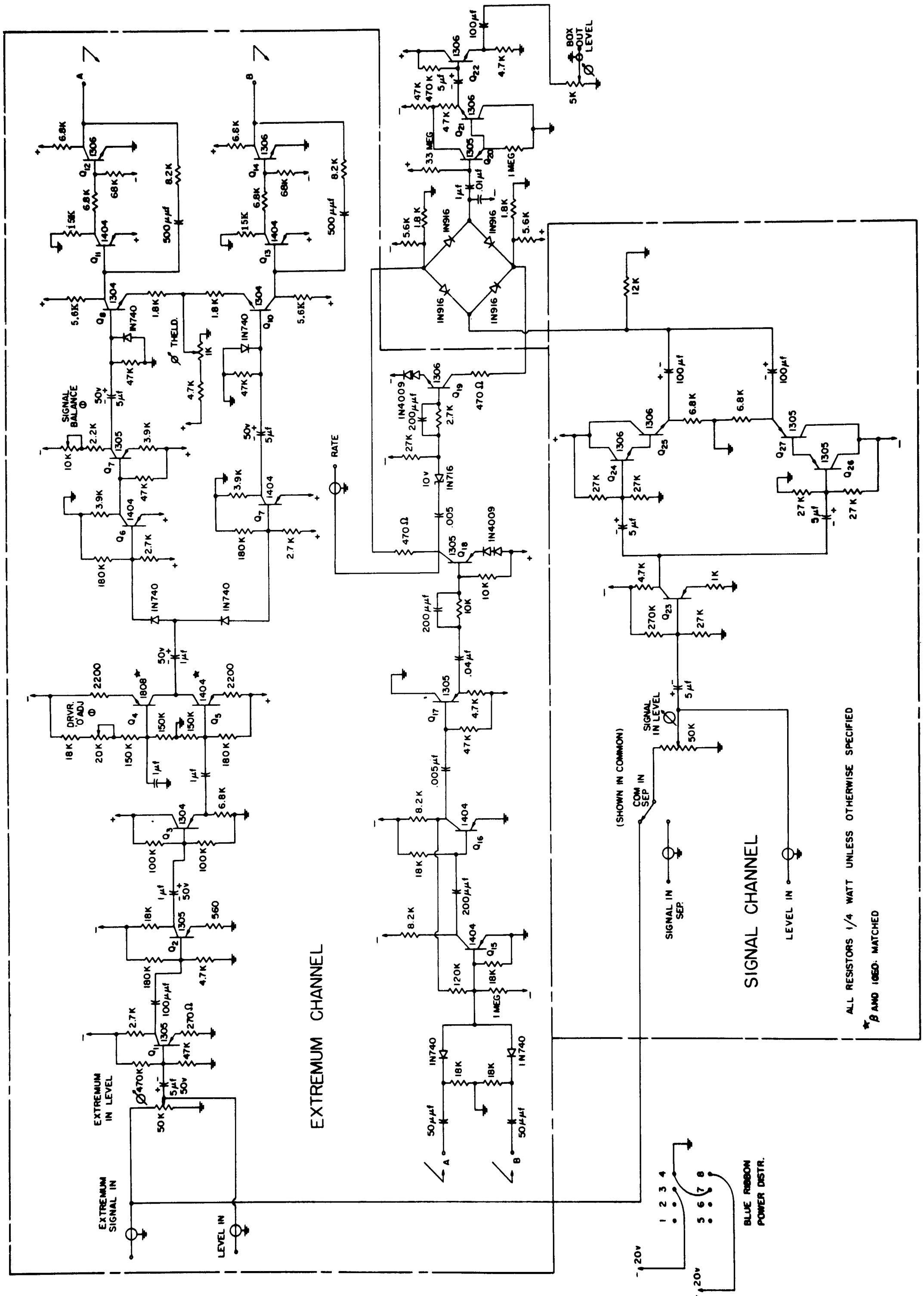


Fig. 15. Extremum sampler.

bases low, thereby accomplishing a high output impedance for signal and a low output impedance for drift components. One of the complementary parts serves only as a load while the other serves as a driver with the signal.

Each half-wave rectified output of the bridge is amplified and one output is inverted. These two signals now drive two stages that have their bases clamped to ground and are biased at or below cutoff with a common bias pot in their emitters. This bias pot provides a symmetrical threshold that the derivative must exceed (when going positive or negative) before either stage can fire its associated Schmitt-type trigger.

Either leading edge of the Schmitts is used to fire a 2-μsec multivibrator which in turn is buffered and drives two switches which drive a sampling bridge. The switches (Q18 and Q19) drive the sampling bridge in a complementary fashion. Initially, opposite parallel legs of the bridge begin to conduct. The particular pair that conducts is determined by whether the storage capacitor is at a higher or lower potential than the signal to be stored. As the storage capacitor charges towards B+ or B-, eventually the signal level will be reached at which time the other legs of the bridge begin to conduct holding the storage capacitor at signal level. The forward drop across the diodes between the signal input to the bridge and the storage capacitor cancel. At the end of the sampling time the switches are turned off and voltage dividers at the switch inputs to the bridge back-bias the bridge, isolating the storage capacitor. The fast sampling time is primarily due to the fact that the storage capacitor charges towards B+ or B- through a relatively low impedance.

The signal to be encoded is buffered and amplified and fed to a push-pull driver of Darlington configuration which, in turn, drives the sampling bridge in the manner just discussed. The push-pull driver provides a very low output impedance to insure fast sampling.

The signal on the storage capacitor at the output of the sampling bridge drives a high impedance circuit consisting of a collector feedback emitter follower whose output is buffered. The output of this buffer is the desired result--a waveform sampled at its extrema.

This extremal sampler is then used in the manner depicted in Fig. 14. As noted before, two such encoders are required for each speech channel.

As mentioned previously, the separate handling of the contiguous frequency bands suppresses much of the distortion from the lower band that would normally fall within the upper band. The experiments described before have shown this technique to be very effective in raising the quality of extremally coded speech from that of single-channel boxcarred speech. The quality of this dual-channel method competes very favorably with that of the more complicated "form-fitting" method previously mentioned. Furthermore, the quality of this dual-channel method has been judged to be superior to the single-channel filtered boxcar even when the sample rate of the dual channel receiver has been reduced to roughly one-half of that of the single-channel boxcar receiver.

4.2.3 Analog Tests. In the time available after construction of this circuit we were able to conduct some listening tests, which were described above. We generally concluded from this that the analog implementation affords telephone quality speech at about 1500 samples per second.

A replay experiment was also attempted with this analog implementation. For this replay experiment the output of the two channels was added as shown in Fig. 13 and was then fed back into the input. The threshold was set at approximately the same level as that for the first pass through the sampler and the speech out was fed into an amplifier for listening tests. There was some decrease in extrema rate as also noted in the computer experiment. There was very little degradation in listening quality. It is therefore concluded that the necessity for restoring the signal to analog form for switching and then resampling will not present a serious problem for the dual-channel analog technique. We would expect the same general behavior for a modified boxcar implementation.

5. SYSTEM DESIGN PLAN

In this design plan we will use the results of all the work described above and specify an asynchronous time multiplexed system which we believe best utilizes the aspects and parameters. We will do this by considering two separate aspects: (1) the system parameters, (2) the system block diagram implementation.

5. 1 System Parameters

The objective in this section is to give the best estimate, based upon all the foregoing work, of a workable asynchronous time multiplexed system which uses extremal sampling. These results represent an updating of the previous estimates presented in Refs. 5 and 7.

We will do this by first considering a system which has the same video bandwidth available as a 12-channel synchronous time multiplexed PCM system. We will choose the system parameters based upon the work described before. Following this we will note the changes necessary if one designs first a smaller asynchronous system, and then a larger one.

Table X shows a comparison of a 12-channel PCM system, with an asynchronous system in which the same video bandwidth (288 kc) is used. As seen in the table the asynchronous system permits approximately a factor of 3. 6 more "off-hook" users than does the synchronous PCM system. In forming this table we have used the information from the feasibility tests and analyses depicted earlier. We will now briefly review the basis for choice of the various parameters shown in this table.

The average sample rate of 1500 samples per second implies the use of either the dual-channel, extremal-sample implementation described in Section 4. 2 or an improved boxcar implementation. It is believed that telephone quality is available at this sample rate.

The next issue concerns the bits per sample. In the asynchronous system we require address bits and possibly bits relating to the buffer time for each sample. We have chosen the following bit construction per sample:

Amplitude bits:	6
Dual channel indication:	1 bit
Buffer time information:	1 bit
Address bits ($\leq$ 64 channels):	6

Thus there are a total of 14 bits per sample in the asynchronous system.

Table X. Comparison of PCM system and asynchronous system.

	12-Channel Synchronous Time-Division Regular PCM	Asynchronous and Buffered Time Division Multiplex Extremal Sampling
Average Sample Rate	8 kc	1.5 kc
No. of Active Users (in talk burst)	12	460.8/1.5 x 14 ≈ 21.9
No. of Connected Subscribers Off-Hook	12	≈ 44
Sample Amplitude Bits	6	6
Address Bits	0	6 + 1 = 7
Time Information Bits	0	1
Total Bits per Sample	6	14
Pulse Width	1.736 μsec	1.736 μsec
τ-interval (channel sample time)	10.42 μsec	14 x 1.736 = 24.3 μsec
Frame Time	125 μsec	- -
Maximum Expected Bit Rate	8 x 6 x 12 = 576 k	576 kc
Average Bit "Occupancy" Rate	576 k bits/sec	0.80 x 576 = 460.8 k
Bandwidth	288 kc	288 kc
$\mu = \frac{\text{Buffer Input Rate}}{\text{Buffer Output Clock Rate}}$	- -	0.80
Buffer Length	- -	6 stage = 146.0 μsec
Freezeout Fraction (Fig. 30, Ref. 3)	- -	0.018
Comments	A standard 12-channel PCM multiplex system	Accommodates about 3.6 times the number of connected subscribers

The use of 6 bits for amplitude information allows the possibility of using companded 64-level bit quantization; this should permit operation with varying volume speech as does 6-bit companded PCM. The six address bits permit encoding up to 64 source numbers. We will only need 44 of these in the system here. Regarding buffer time information, we will see below that the maximum buffer delay time is about 146 microseconds. Since this value exceeds the maximum tolerance for high quality speech (Section 4. 1), we divide the total buffer time by two so that the maximum sample jitter will be about 73 microseconds. [1]

The remaining important parameter is the buffer length. We have used the curves of Fig. 4 to estimate the freezeout. Note that this assumes a constant "u", which in turn assumes a constant number of simultaneous talkers in talk burst. On the average, the number of talkers in simultaneous talk burst (active users) is taken to be one-half the number of subscribers "off-hook." Actually, the number of active users will vary about a lower mean with some variance, as was described in Section 3. 5. Also, we have found from the freezeout experiments (Section 3. 6) that the freezeout tolerance is quite lax; therefore one need not tightly specify the freezeout. On the constant "u" basic, we have chosen a 6-sample buffer, which gives a freezeout fraction of about 1. 8 percent for a u of 0. 8. From computer experiments it has been found that this freezeout fraction is not noticeable with casual listening. Further, it is expected that freezeout with the dual-channel analog implementation is reasonably represented by the computer simulation case.

It may be noted that in previous work (Refs. 5, 6 and 7) used a buffer length of 16 samples. From the freezeout and jitter experiments which we have performed (Section 3. 6), it is apparent that such a long buffer is unnecessarily conservative.

The other item that requires comment with this system is the expected quality. Based on subjective listening tests, we believe that the system provides quality comparable with telephone quality. Actual speech reproduction from each implementation will of course differ somewhat, but the pleasantness, naturalness, etc., if the dual-channel, extremal sampled system is used is believed to be comparable with telephone quality speech.

As noted, then,this asynchronous system would allow about 3. 6 times as many off-hook subscribers as would a synchronous PCM system of the same bandwidth.

If a smaller asynchronous system is designed, the variance of the simultaneous talkers will become a more important factor. Also, the maximum possible jitter for a given sample-size buffer increases. However, it is predicted that a bandwidth saving of about 3. 0 can be achieved (over synchronous PCM) with a 12-channel (off-hook) system.

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[1] It may be possible to eliminate the buffer time bit without any degradation of quality; however, this must be determined experimentally.

If a larger system is designed, the jitter and freezeout problems diminish, due to the added natural smoothing of the irregular samples from the higher number of sources. Consequently, buffer time information can probably be ignored for such larger systems.

5.2 System Implementation

A functional block diagram implementation of an asynchronous system is shown in Figs. 16 and 17. Figure 16 depicts the transmitter which consists mainly of the extremal detector and the buffer. For the extremal detector we would propose to use either the dual-channel extremal sampler described above in Section 4.2 or an improved boxcar receiver implementation. Each of these techniques implements an extremal sampler (and receiver) in a relatively simple and straightforward way. Consequently, the circuit of Fig. 15 can be considered as a potential prototype system. Although the dual channel method requires a filtering action, this could probably be accomplished with microelectronics by using active RC circuit techniques.

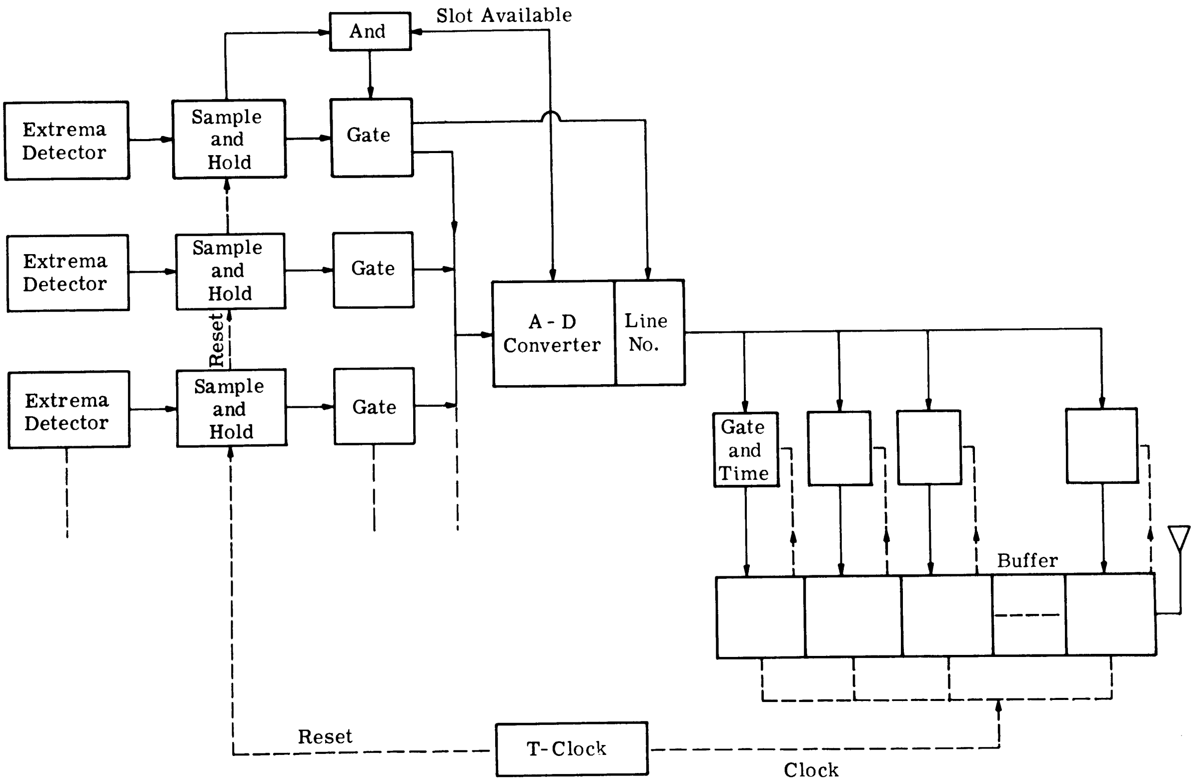

Fig. 16. Component diagram of asynchronous multiplexing transmitter.

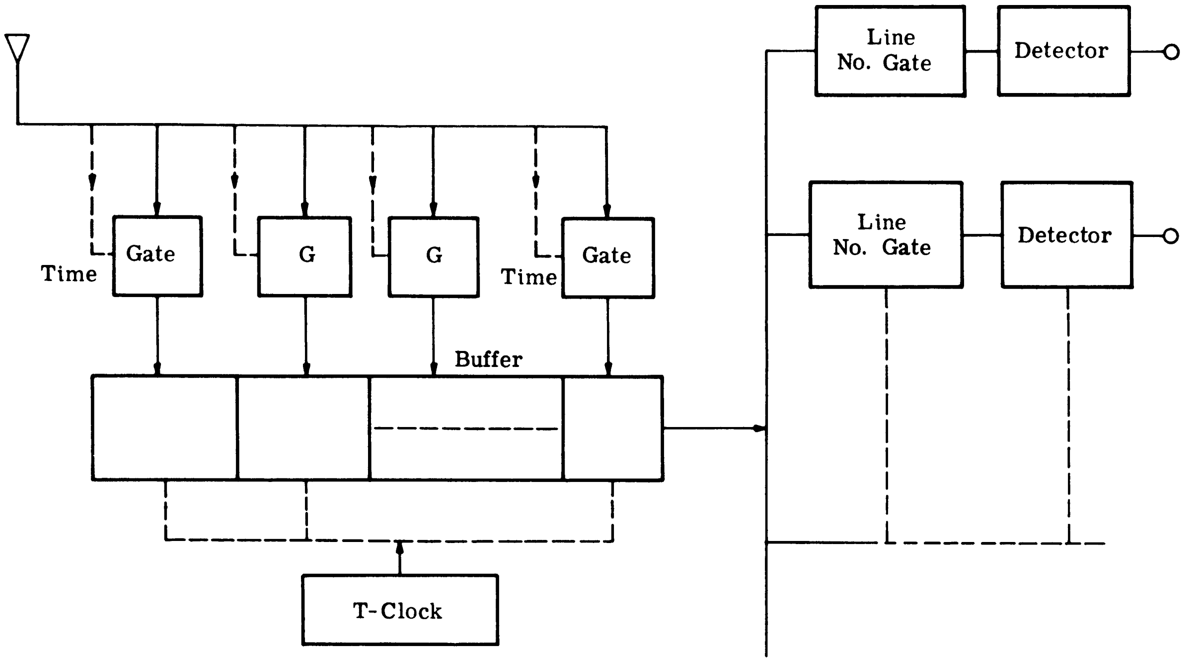

Fig. 17. Component diagram of asynchronous receiver.

The general idea in the transmitter (Fig. 16) is to put any extrema which occur within a given τ-interval into the earliest available spot in the buffer. Further, it is desirable to convert the samples from analog to digital in a central place, rather than in each channel. Hence the A-D converter is expected to be placed in the central facility.

Starting with the extrema detector, then, assume that analog samples can appear at any random time. First a sample-and-hold circuit would be set, and this would be reset at the end of each τ-interval. An AND gate would be required to control the entrance of the sample into the A-D converter. The AND gate would require that the A-D converter be free and the sample and hold circuit have a stored value. Under these conditions the gate permits the sample to go into the A-D converter. The sample is converted and a source number is added. The output then is the sample containing the amplitude information and the source number.

The next step is to enter the transmitter buffer at the earliest (nearest to the output) available position. This can be accomplished by gating the buffer input; the status of the gate is determined by the occupancy of the buffer. Also, the necessary buffer time information can be inserted at this gate. The result is a total sample placed in the buffer which contains the amplitude, the address, and the buffer time information.

A system receiver block diagram implementation corresponding to the transmitter implementation is shown in Fig. 17. The total sample would enter the buffer at the commensurate gate, depending on the buffer time information. Then, depending on the address, the samples would be dispersed to the proper lines through a line-number gate. Following this, one would go through the detection process, as described in Section 4. 2.

We now consider the implementation requirements. All of the equipment to the left of the dashed line in Fig. 16 must occur in each channel. Therefore, every function here must be kept to the simplest possible form. As mentioned, it is expected that microelectronic techniques could accomplish these functions in a small physical dimension. For the system described in Table X, the sample-and-hold circuit and the gate must operate at a rate of about 50 kc (24. 3-μsecond intervals).

Only a single A-D converter and buffer is required for the asynchronous system. Consequently, more complexity would be permitted here. For the system that we have shown the A-D converter should be capable of converting 6 samples in the 24. 3-microsecond sample time. This is the upper limit that would occur if the buffer were originally empty and 6 simultaneous samples occurred within the next τ-interval.

Assuming that we wish to account for this upper limit speed[1] the converter would have to convert six 14-bit samples in 24. 3 microseconds. This means that the bit rate for the A-D converter must be 3. 5 megacycles, comprised of 14-bit samples at the rate of 250, 000 per second. Such an A-D converter is within the state-of-the-art.

There should be no speed problem in the buffer since it is receiving the samples serially. With a 6-stage buffer one needs 84 flip-flops of storage to account for six (6) 14-bit samples. Such a buffer unit would not be large, especially if microelectronic implementation were used.

Based on the above discussion it is apparent that there is no instrumentation requirement which is extremely difficult to realize. It must be remembered that the system depicted here should be compared to that equipment which is necessary with synchronous PCM.

---

[1]One need not necessarily account for this joint event "buffer empty"--and "6 simultaneous events." However, it appears this speed is within the state-of-the-art.

6. CONCLUSIONS AND RECOMMENDATIONS

The work described in this report has shown that use of an asynchronous time multiplexing system can increase the number of users for a given synchronous PCM system by a factor of about 3.6. This factor is sufficiently high to be exploited whenever a given digital (multiplexed) facility is expensive to duplicate. Examples of this are tactical Army trunk links, oceanic cables, and communication satellite links.

In general, the chief advantage of the asynchronous time multiplexed method is the saving in bandwidth (or power) for a given number of users, or the equivalent increase in possible users for a given constant bandwidth. The chief disadvantage of the system is the need for equipment additional to that required for PCM. The major addition is the extremal detector in each channel; PCM requires only a sampler. However, microelectronic techniques make such equipment complexity a matter of economics rather than of size and reliability.

In addition to the straightforward advantage of multiplexing voice channels, there are many ancillary benefits of using the asynchronous time multiplexed system. First, one can intersperse nonreal time data at the rate of 20 percent of total capacity. Furthermore, the system has the elastic overload feature which is common to all asynchronous systems. Thus a certain system design could tolerate a system overload at the expense of added freeze-out averaged over all channels. If a system is designed for 44 off-hook subscribers, one could permit 60 at the expense of deterioration of each channel. Such gradual degradation is not possible with a synchronous system.

To assure system operation under actual conditions, a parallel computer and field study should be implemented. The complete system should be simulated on a computer, assuming traffic parameters are the starting point. In parallel with this computer study, the system should be tried, with actual hardware, in a given trunk application.

Further work should investigate the effect of sequential extremal sampling (and restoring) for the dual channel implementation method. Also, the basic aspects of performing digital switching for the extremal sampled speech should be investigated.

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APPENDIX A

EQUIPMENT

During the past year a new set of data conversion equipment was acquired by this laboratory. This equipment is an interface between electronic experiments and the digital computer. The new equipment has increased flexibility and speed and should provide a more reliable system. Essentially, the equipment prepares and reads digital tapes. The purpose of this section is to compare the new equipment with the original system.

The original system was limited to a 5-kc sampling rate and used only the low density (200 bpi) on the digital tape. Since however the number of words per record was not precisely controlled, record length varied from record to record. This equipment could be operated in either a continuous record (i. e., no record gaps) or record with gaps and with either 6-bit or 10-bit accuracy.

The characteristics of the new system are shown below.

The highest sampling rate for this equipment is 41, 667 samples per second, with a 6-bit amplitude description (64 levels). The amplitude levels can be increased to 10 bits (1024 levels); for any number of bits greater than 6 the maximum sampling rate is 20, 556 samples per second. The present equipment will allow multiplexing up to 4 channels. In the future we will be able to add up to 24 multiplexed channels by purchasing additional electronic cards.

There are three relatively independent constraints in the operation of such equipment: 1) the maximum sampling rate for a given sample length (in bits), 2) the highest analog frequency of a full-scale input signal, considering only the accuracy at the sample points, and 3) the highest analog frequency of a full-scale input signal considering also the reconstructable accuracy between sample points.

The sampling rate is controlled by two possible tape recording densities, and by the fact that we can combine two tape characters into a single sample. The following reference sample rates are possible:

Although the following table lists only 6-bit and 10-bit samples, it is clear that intervening sample lengths may be accomplished by having the computer ignore lower order bits. Also, sampling rates other than the reference IBM rates listed are permissible.

Sampling Rate (samples/sec)	Bits	Tape Density (bits/inch)
41,667	6	556
20,833	10	556
15,000	6	200
7,500	10	200

The second constraint, highest analog frequency considering the sample point, is affected by, 1) the aperture time of the input (i. e., the input signal may not change during a sample "look"), and 2) the amplitude precision (sample length). This constraint applies if it does not need to confine precisely the error between samples (if the waveform were reconstructed either in the computer or by a filter). This is the usual case if one is going from an analog signal to the computer and not doing the reverse. In such cases the amplitude precision is determined by the type of computer processing and the dynamic range of the signal.

For this case the equipment here offers the highest (full-scale) frequencies shown below, versus sample length. The criterion is used that the full-scale highest frequency should not move more than one-half of the amplitude quantization level during the aperture time. Although not precisely tested, the effective aperture time of this equipment is estimated to be 75 nanoseconds.

In these terms, then, the highest full scale frequencies for the various sample lengths, are:

Bits	Highest Full Frequency for Aperture Error (cps)
6	33,000
7	16,500
8	8,250
9	4,000
10	2,000

If the analog waveform is to be reconstructed, either in the computer or by a filter, the error between samples (called sampling rate error) must be included when determining highest analog frequency. (This is the third case mentioned above.)

It is well known that whenever the analog waveform is to be reconstructed for use in a quantitative manner, one must in practice always sample at a rate greater than twice the highest frequency (20,000 for a 41,000-sample rate). The rate depends greatly on the reconstruction method that is used. For some data processing applications one may wish the ratio of sample frequency to highest frequency to be as high as 70 (for 10-bit accuracy). Such special applications would require the direction of special attention to each case. Here we will portray an overall "engineering" result. For reasonable quantitative reproduction we will assume that the ratio of sampling frequency to highest signal frequency should be at least 4 to 5. On this basis the following frequencies versus amplitude accuracy are possible:

Bits	Highest Frequency (cps)	Sampling Rate
6	10,000	41,667
7	5,000	20,833
8	5,000	20,833
9	4,000	20,833
10	2,000	7,500

Note that often the highest frequency can be altered by recording the analog information on an analog magnetic tape and changing the speed by a precise factor. For this the signal-to-noise ratio must be kept high and the speed must be precisely controlled (i. e., a good quality tape recorder must be used).

APPENDIX B

ANALOG OPERATION

Signal level adjustments and monitor points are provided in both channels. The level into the extremum detection channel is usually set for 1-volt peak maximum to utilize the full dynamic range. Since the sampling bridge has a slight hysteresis, best results are obtained with the level into the signal channel set at its rated input of 1-volt peak. This, however, is far from a limiting factor. A boxcar output level adjustment is also provided and serves primarily as a volume control. The threshold adjustment is based in conjunction with the sample rate output, which provides a 2-μsec pulse from -5 to +18 volts at the time each sample is taken, driving a counter in order to set the desired sample rate. The sample rate is, of course, a function of the level input to the extremum channel and so the level should be set before the threshold is set. In rare cases the input level to the extremum channel may be varied to control the sample rate.

Specifications

Input impedance	Extremum Detection channel--min. 20 K Signal Channel--min. 20 K
Boxcar Output Impedance	1 K
Rated Input	Both channels 1-volt peak max.
Rated Output	2-volt peak (boxcar)
Overall Response	See Fig. 18
Dynamic range	25 db at 400 cps to detect extrema accurately

Alignment

In addition to the aforementioned front panel adjustments, there are two screw driver adjustments. The pot in the base of Q_4 should be adjusted for 0 volts ±2 at the collector junction of Q_4 and Q_5. This puts the high impedance driver in the center of its operating range. The pot in the collector of Q_7 is adjusted so that the positive and negative threshold crossings are symmetrical about zero. In order to do this place com. sep switch in com. and connect a sine-wave generator to the extremum signal input. Set the generator at 400 cps and adjust input levels for the 0. 3-volt peak and the boxcar out level for maximum. Observe the boxcar output on a scope. Adjust the pot until the square wave on the scope remains symmetrical as the threshold is varied. The amplitude of the square wave will decrease as the threshold is increased, but the threshold should not be increased to the point where the square wave breaks up or disappears.

These adjustments are not critical and once made will seldom, if ever, need readjustment.

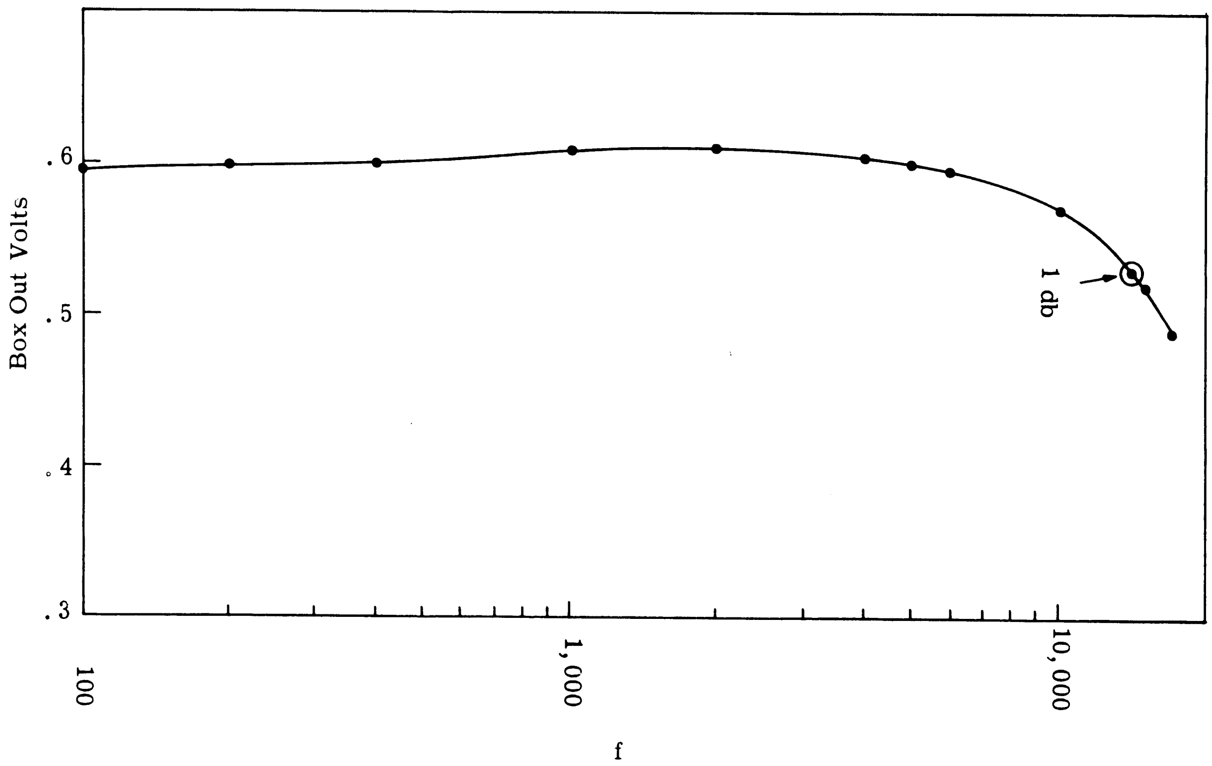

Fig. 18. Frequency response of extremum sampler boxcar output rated signal in.

APPENDIX C

COMPUTER PROGRAM MODIFICATION

The computer programs which have been used for the simulation studies during the past year are essentially the same as those used in Ref. 7. The purpose of this section will be to discuss briefly those modifications and additions which were made.

The extremal sampling program was changed slightly in order to make use of the new equipment. It is now possible to use more than one file of input data and to write more than one file of output. The clip level has been made part of the data for the program so that several runs with different clip levels can be made a part of one problem. The one addition to the program was for the "Zero Level" tests described in Section 4. 1. 2. 2. For this test the number of points which fall within the threshold are counted and when this number exceeds a predetermined number (NPTS, fed in as data), an extrema is found whether the threshold is crossed or not.

The jitter program was a combination of the extremal coding program and a special program. The input was speech-sampled at 40 kc and the extrema were located in the usual manner (using the extremal sampling program). The value of the extrema and the time of occurrence are then written on an intermediate tape which is then used as input to the second program (Jitout). In the second program the time slots are quantized to 50 μsec, 100 μsec, and 200 μsec. If more than one extremum occurs in the time slot, the earliest extremum is used and the remainder are discarded. The occurrence of the two samples in a time slot is noted and the total number of times this occurs is printed out as part of the output data.

The freezeout program starts in the same manner as jitter; i. e., a 40-kc input is used to find extrema which, with time of occurrence, are written on an intermediate tape. This tape is then fed into another program which freezes out extrema depending on the value of N' used. The output of this program is another intermediate tape with the remaining extrema which is then fed into the final program which performs the receiver operations (curve fits the data).

APPENDIX D

TEST DATA

The purpose of this appendix is to show the record-by-record data for the extremal tests discussed in Section 4.

Table XI. Threshold test results.

Threshold	± 3		± 4		± 5		± 6		± 7		± 8		± 9		± 10	
Record No.	No. of Ext.	S/N	No. of Ext.	S/N	No. of Ext.	S/N	No. of Ext.	S/N	No. of Ext.	S/N	No. of Ext.	S/N	No. of Ext.	S/N	No. of Ext.	S/N
1	716	10.75	646	9.61	632	9.55	595	8.94	583	8.89	535	7.99	529	7.84	513	7.64
2	2410	11.62	2144	10.81	2032	10.32	1929	9.97	1853	9.53	1671	8.97	1619	8.76	1567	8.64
3	2452	10.22	2204	9.88	2106	9.52	1962	9.37	1862	9.07	1694	8.54	1634	8.40	1576	8.32
4	2490	10.54	2226	10.07	2128	9.87	2004	9.26	1906	8.56	1730	7.99	1678	7.91	1606	7.78
5	1934	10.25	1651	9.39	1545	8.64	1385	8.26	1235	7.79	1013	6.19	955	6.23	899	5.94
6	2371	11.24	1965	10.46	1865	10.24	1665	9.83	1556	8.97	1408	8.11	1360	8.03	1309	7.91
7	2158	12.19	1832	11.53	1726	11.46	1580	11.12	1454	10.30	1296	10.09	1249	9.97	1185	9.59
8	1944	10.97	1696	10.12	1605	9.75	1503	9.32	1403	9.00	1238	7.91	1200	7.78	1138	7.65
9	1366	10.11	1095	9.23	1011	8.96	898	8.23	818	7.69	706	7.54	678	7.38	646	6.96
10	2127	10.65	1743	10.21	1637	10.10	1465	9.75	1367	9.32	1202	8.28	1147	7.97	1095	7.92
11	1839	11.38	1598	10.18	1535	9.78	1425	9.07	1329	8.61	1183	7.71	1141	7.15	1097	7.04
12	411	2.24	251	1.85	229	1.85	144	1.58	134	1.35	102	1.31	92	1.07	78	0.91
Total Ext.	22218		19051		18051		16555		15500		13778		13282		12709	
Rate (Extrema/sec)	2469.2		2117.2		2006.1		1839.8		1722.6		1531.2		1476.1		1412.4	
Avg. of S/N		10.18		9.45		9.17		8.73		8.26		7.55		7.37		7.19

Table XII. Zero modification results.

	NPTS = 8		NPTS = 4		Previous Test	
Threshold	± 10		± 10		± 10	
Record	Rate	S/N	Rate	S/N	Rate	S/N
2	All	Noise	All	Noise	All	Noise
3	1983	8. 68	2174	9. 60	1937	8. 34
4	3466	7. 23	3587	7. 54	3396	7. 10
5	3513	4. 83	3667	5. 41	3445	4. 61
6	3363	1. 07	3474	1. 55	3322	1. 01
7	2626	10. 54	2744	10. 91	2507	10. 17
8	3226	10. 99	3360	11. 60	3117	10. 81
9	2814	11. 51	2907	11. 81	2666	11. 15
10	2959	7. 41	3090	7. 99	2884	7. 09
11	2111	8. 84	2223	9. 53	1989	8. 28
12	1470	10. 47	1531	11. 34	1380	10. 05
13	3286	-0. 14	3390	0. 08	3204	-0. 314
14	1728	10. 34	1788	10. 49	1671	10. 19
Average	2712. 1		2827. 9		2626. 5	
Total	21834		22765		21145	
Avg. of S/N		7. 65		8. 15		7. 37

Table XIII. Multiple-speaker results.

Speaker	AR		LT		JV		WR		TB	
Record No.	No. of Ext.	S/N	No. of Ext.	S/N	No. of Ext.	S/N	No. of Ext.	S/N	No. of Ext.	S/N
1	743	5.53	566	5.16	347	1.78	505	12.65	84	-2.19
2	1246	11.99	1294	11.62	1049	10.50	1188	11.35	1055	10.66
3	1289	12.78	1232	12.60	1318	10.43	952	11.58	1511	11.93
4	1156	9.07	1215	11.23	1121	11.54	952	10.22	987	11.21
5	1518	8.56	1444	12.28	1052	11.45	1120	10.31	1318	9.89
6	928	8.62	1427	10.27	964	11.54	742	9.48	651	10.56
7	1146	8.91	976	9.04	1013	9.82	345	9.91	1372	11.66
8	316	2.87	1492	12.85	466	2.11	943	12.50	615	12.15
9	918	11.87	1457	10.53	1427	12.65	929	11.67	928	11.72
10	1611	14.24	1079	12.94	1366	9.70	888	12.74	999	12.61
11	1186	11.00	671	7.57	1140	12.23	284	10.99	748	11.23
12	647	9.76	1028	10.32	897	11.10	916	10.60	280	5.67
13	109	-0.04	1635	10.50	572	2.53	1423	11.85	828	10.59
14	1161	12.39	651	10.31	999	-4.17	577	12.26	1197	9.41
15	1383	12.04	--	--	1431	8.74	--	--	436	12.50
16	1076	11.31	--	--	1027	12.65	--	--	--	
17	476	8.11	--	--	--	--	--	--	--	
Total	16909		16167		16189		11764		13009	
Rate	1326.5		1540.0		1349.4		1120.6		1156.6	
Overall S/N		11.17		11.39		11.21		11.37		11.03
Time (sec)	11.9		9.2		10.9		9.3		10.0	
Talk Burst Time (sec)	9.6		7.5		8.8		8.2		9.0	
Talk Burst Rate	1750		2150		1830		1420		1450	

Table XIV. Hand-set results.

Record No.	No. of Ext.	S/N	No. of Ext.	S/N
1	26	-0.51	5	0.02
2	1005	9.60	1299	5.56
3	2338	10.92	2548	8.50
4	1923	9.50	2197	12.63
5	1707	9.94	2356	10.99
6	1496	9.69	1682	9.40
7	1508	11.05	2229	12.08
8	1825	12.22	657	1.12
9	1656	11.78	1575	10.94
10	2185	12.05	2370	11.18
11	1435	11.77	1417	7.36
12	614	11.41	542	5.05
13	1946	9.26	1289	8.47
14	1286	9.58	2372	8.31
15	--	--	1464	5.46
Total Ext.	20950		24002	
Rate (extrema/sec)	1995.6		2133.8	
Avg. of S/N ratios		9.95		7.80

Table XV. Replay experiment results.

Record No.	No. of Ext.	S/N
1	510	12. 56
2	1811	9. 37
3	1786	6. 40
4	1853	11. 61
5	1130	11. 18
6	1617	11. 73
7	1411	4. 54
8	1280	2. 98
9	824	11. 20
10	1690	10. 24
11	1116	12. 02
Total Ext.	15028	
Rate (extrema/sec)	1824. 4	
Overall S/N		11. 85

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