Operational Evaluation of a Voice Concentrator Over AUTOVON Interswitch Trunks

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Abstract—**This paper describes results of test and evaluation of a commercial voice concentrator applied to AUTOVON interswitch trunks between Feldberg, Germany and Ft. Detrick, MD. By virtue of time assignment speech interpolation (TASI) techniques, the voice concentrator provided approximately a 2-to-1 compression of voice channels onto trunks, with a configuration of 17 channels onto nine trunks selected for this AUTOVON application. Tests consisted of: 1) signaling performance characterization through the voice concentrator, 2) voice channel characterization of the nine selected trunks before and after cutover of the voice concentrator, 3) performance characterization of data signals operated through voice concentrator channels, 4) traffic data collection and analysis, and 5) user subjective evaluation.**

Test results indicated acceptable performance for the intended application, which was limited primarily to voice signals. Since completion of the testing and subsequent analysis, the Defense Communications Agency (DCA) has accepted the voice concentrator system for operational use with AUTOVON IST's in the Defense Communications System (DCS).

I. INTRODUCTION

A desire to improve the grade of service on AUTOVON interswitch trunks has led to the installation, test and evaluation, and approval of a voice concentrator for AUTOVON IST's operating between Ft. Detrick, MD and Feldberg, Germany. The voice concentrator utilizes time assignment speech interpolation (TASI) to concentrate up to $(2n - 1)$ voice circuits into n trunks, thus providing an improved grade of service on existing trunks or allowing a reduction in the existing number of trunks with attendant cost savings. Several voice grade interswitch trunks were selected from CONUS-Europe AUTOVON IST's as a means of alleviating already saturated communications facilities.

This paper provides results, analyses, and conclusions derived from tests conducted during the period October 1979–February 1980. An operational configuration of 17 channels onto nine trunks was selected for a test between Ft. Detrick and Feldberg, as shown in Fig. 1. Additionally, a mix of media and transmission equipment was selected for routing of the nine trunks. The selection of heavily used AUTOVON trunks, a mix of media with different trunk routing, and a mix of FDM and PCM transmission equipment meant a stringent test environment.

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II. DESCRIPTION OF VOICE CONCENTRATOR

The commercial voice concentrator under test [1] uses microprocessor and digital processing technology to statistically multiplex voice conversations onto trunks in a ratio of $2N - 1$ incoming channels to N outgoing trunks. TASI techniques combined with the use of buffers allow efficient multiplexing of even small trunk group sizes. TASI consists of filling in the silent gaps and pauses occurring in one voice channel with speech from another channel. This technique has previously found application with submarine cable and satellite media for relatively large trunk groups [2].

The voice concentrator transmits and receives voice in an analog mode. Internally, it processes the voice signal in digital format by utilizing standard 64 kbit/s PCM with $\mu(255)$ law companding. Incoming speech on the channel side is detected by speech detectors and simultaneously is stored in a fixed (32 ms) buffer (256 byte memory) assigned to that channel. If a trunk is available, a control symbol is first transmitted to direct the far end to place the speech burst on the correct channel. The delayed speech burst follows the symbol on the same trunk. The fixed buffer delay serves the purpose of allowing time for speech detection and transmission of the symbol without clipping the leading portion of the speech burst. Should an idle trunk not be available, buffers with variable length are used to store speech, providing a delay of up to 3.0 s, until a trunk becomes available. At this point, following the transmission of the symbol, speech is transmitted from the buffer continuously until termination of the speech burst. Hence, the buffer introduces a fixed plus variable delay equal to the time required to detect the initial part of the burst and to locate an available trunk for transmission. When the speech burst terminates, the buffer releases the memory it has acquired until the buffer contents are depleted, at which time the buffer is returned to the pool [3].

Loss of speech segments may result from initial delay while waiting to acquire a buffer and from buffer overflow when all buffer segments are in use. Once a buffer request is made, failure to provide a buffer because all are in use leads to loss of initial speech, called clipping. If the buffers begin to saturate, a speed-up release of segments is started. This is done to release segments faster for new incoming speech.

Under certain conditions of heavy traffic volume, overload may occur which will result in blocking any unseized channels from being seized. Speech activity is monitored and compared to a predetermined threshold for overload. This threshold is

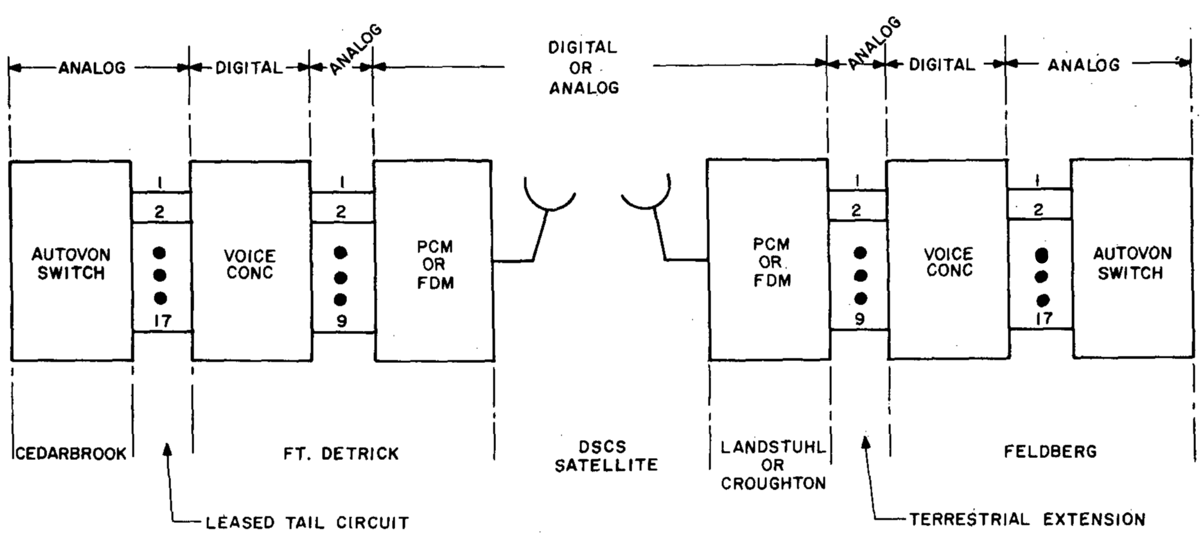

Fig. 1. Operational configuration of voice concentrator.

determined by the total number of speech-available trunks (total number of trunks minus total number of data calls currently active). To come out of overload, average speech activity is continuously compared to this threshold.

E&M leads are provided by the voice concentrator for supervision, but in-band supervision is also possible for answer back. Likewise, address signaling may appear on the E&M leads (dial pulse) or may be accommodated in-band (DTMF or MF). In-band supervision and signaling is treated as speech, although no variable delays are involved once the trunk has been seized since supervision signaling has the highest priority for trunk usage. Signaling appearing as E&M is converted to in-band for transmission and is reconstituted as E&M at the distant end. For E&M supervision signals, a fixed delay of approximately 100 ms is introduced. For dial pulse signaling, a fixed delay of approximately 400 ms is involved with transmission of each dialed digit. For in-band (DTMF or MF) address signaling, a fixed delay of 60 ms is introduced, plus a variable delay from zero to several hundred milliseconds, depending on the load at that instant.

The concentrator automatically bypasses itself because of failure (system or power) by connecting trunk leads directly to corresponding channel leads via internal relays. One channel and trunk (number 1) remain dedicated to the concentrator, which permits intermachine communication and system reinitiation. Channels 2 through n are then available for call processing during bypass, being connected through the relays to trunks 2 through n. Channels $n + 1$ through $2n - 1$ have no corresponding trunks for connection and are busied out by the concentrator.

III. TEST RESULTS

A. Background

The configuration employed with this test provided compression of 17 tail circuits onto nine interswitch trunks. Of the nine trunks, seven were routed through the PCM multiplex of the Defense Satellite Communications System (DSCS) between Ft. Detrick, MD and Landstuhl, Germany. From Landstuhl, these seven trunks were routed to the AUTOVON switch at Feldberg using analog line-of-sight (LOS) links. The remaining two trunks were routed through frequency division multiplex (FDM) of the DSCS between CONUS and Croughton, England. From Croughton, these two trunks were extended to Feldberg via LOS links, although initially a combination of troposcatter and LOS links was used which proved unacceptable because of noise levels which exceeded DCA standards.

B. Performance Characteristics of Signaling

Operation of the voice concentrator through AUTOVON required interface with supervisory, addressing, and control signals [4]. Transparency to AUTOVON precedence and preemption control signals was accomplished without modification of the voice concentrator or use of ancillary equipment. However, interface with other signaling required ancillary equipment at both Ft. Detrick and Feldberg, as indicated in Fig. 2. The access lines from the Cedarbrook AT&T switch were all four-wire with SF in-band signaling. The voice concentrator utilized E&M signaling for interface between the channel side (non-TASI) and the switching equipment. Hence, an SF/E&M converter was required at Ft. Detrick, along with a pulse link repeater (PLR) to provide conversion of the E lead to the M lead and vice versa. Level conversion was provided from 0/0 (transmit/receive) dB levels (relative to transmission level points) at Cedarbrook and Ft. Detrick to the +7/−16 dB voice concentrator levels. Intermachine signaling (between trunk sides of the voice concentrator) is provided in-band using coded tone bursts. This in-band technique was compatible with the PCM multiplexer. However, in the bypass mode of the concentrator, E&M signaling is connected straight through, as shown by the dashed lines of Fig. 2. Since the PCM multiplex required in-band signaling, an SF/E&M converter was required to interface the concentrator, when in the bypass mode, at both Ft. Detrick and Feldberg. Also at Feldberg,

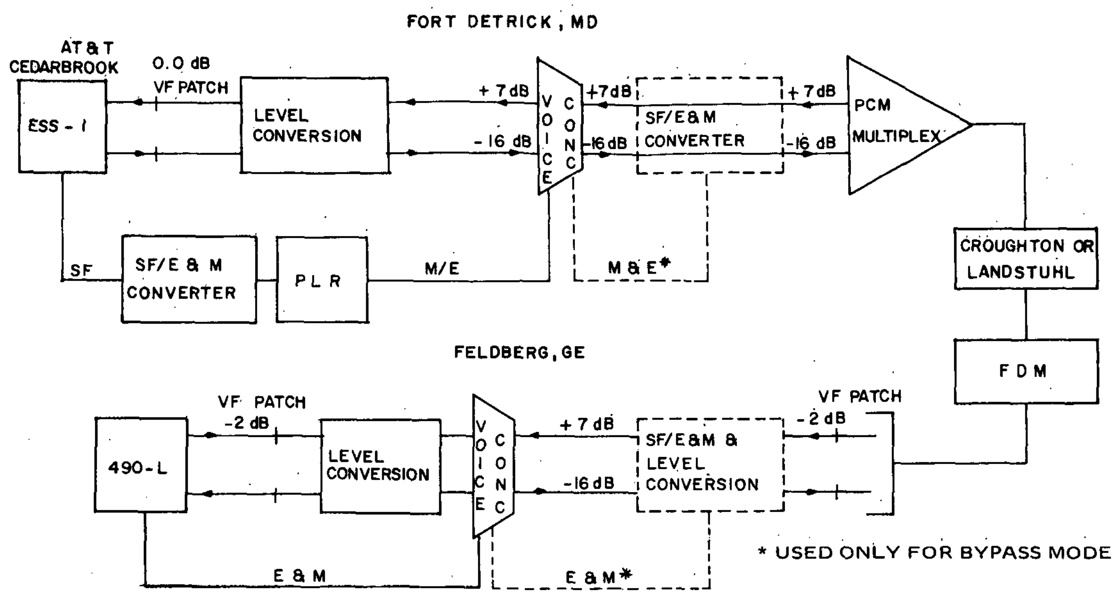

Fig. 2. Voice concentrator level and signaling interfaces.

−2/−2 dB levels are used so that level conversion was required for interface with +7/−16 dB levels.

Once initial signaling compatibility had been demonstrated, operational testing of the concentrator commenced. Calls placed through the concentrator originated from both two-wire phones which were homed on PBX's and from four-wire phones homed directly on the AUTOVON switch. By placing low precedence calls, AUTOVON preemption could be expected, particularly during the busy hours. In this manner, voice concentrator transparency to AUTOVON preemption and precedence signals was successfully demonstrated. This operational configuration presented a stringent test because of the numerous antiquated PBX equipment, homed on the Feldberg AUTOVON switch. Only one signaling problem arose during the test period, in which certain European PBX's were inadvertently generating a 50–70 ms on-hook signal when the distant (U.S.) party answered. The Feldberg voice concentrator stretched this on-hook signal to approximately 200 ms, which was recognized as a true on-hook signal by the Ft. Detrick concentrator and resulted in call termination. Once this problem had been isolated to these PBX's, a simple wiring change in each PBX eliminated the momentary on-hook signal and cleared up the problem.

C. Performance Characteristics of Voice Channels

The effect of TASI on voice channel parameters was determined by means of two baseline tests. First, the nine interswitch trunks (IST's) were tested for voice channel performance before cutover of the voice concentrator. Second, the 17 channels were tested through the voice concentrator using the test configuration of Fig. 3. Tests consisted of a routine quality control (QC) test of those circuit parameters called out in DCA standards [5], [6]. A comparison of trunk versus channel performance then indicated the effect of TASI on voice channel performance. The required level of performance pertained to DCS interswitch voice grade circuits as prescribed by DCA standards [5].

The following circuit parameters were tested for both TASI and non-TASI channels, using the transmission impairment measuring set (TIMS) shown in Fig. 3 and following DCA prescribed procedures [6]:

1) idle channel noise
2) impulse noise
3) frequency response
4) envelope delay
5) signal-to-noise ratio
6) single tone interference
7) frequency translation
8) harmonic distortion
9) net loss variations
10) phase jitter.

A comparison of test results on the nine IST's (without TASI) with DCA requirements [5] indicated that all trunks met all specifications except idle channel noise (ICN). The first seven trunks which were routed through PCM multiplex had ICN measurements in the range 22-29 dBrnC0. Trunks 8 and 9 were routed over FDM links and provided significantly worse ICN in the range 39-48 dBrnC0. Specified as a function of circuit length [5], ICN should be no greater than 47 dBrnC0 for these circuits. Those trunks routed via analog transmission were thus marginally acceptable.

Results of channel characterization of the TASI trunks are shown in Table I. Note that measurements of idle channel noise and impulse noise were not taken since they are meaningless tests for a TASI system. The absence of any transmitted energy, as prescribed for these two tests [6], will result in the momentary dropping of such a channel by TASI circuitry until such time as the transmit channel again becomes active. During the period of transmit channel inactivity, the far-end receive channel will be quiet, that is, no background noise would exist since the "channel" has been disconnected from the trunk. The test used in place of idle channel and impulse noise was signal-to-C-notched noise ratio, accomplished by measuring test tone plus noise levels and noise

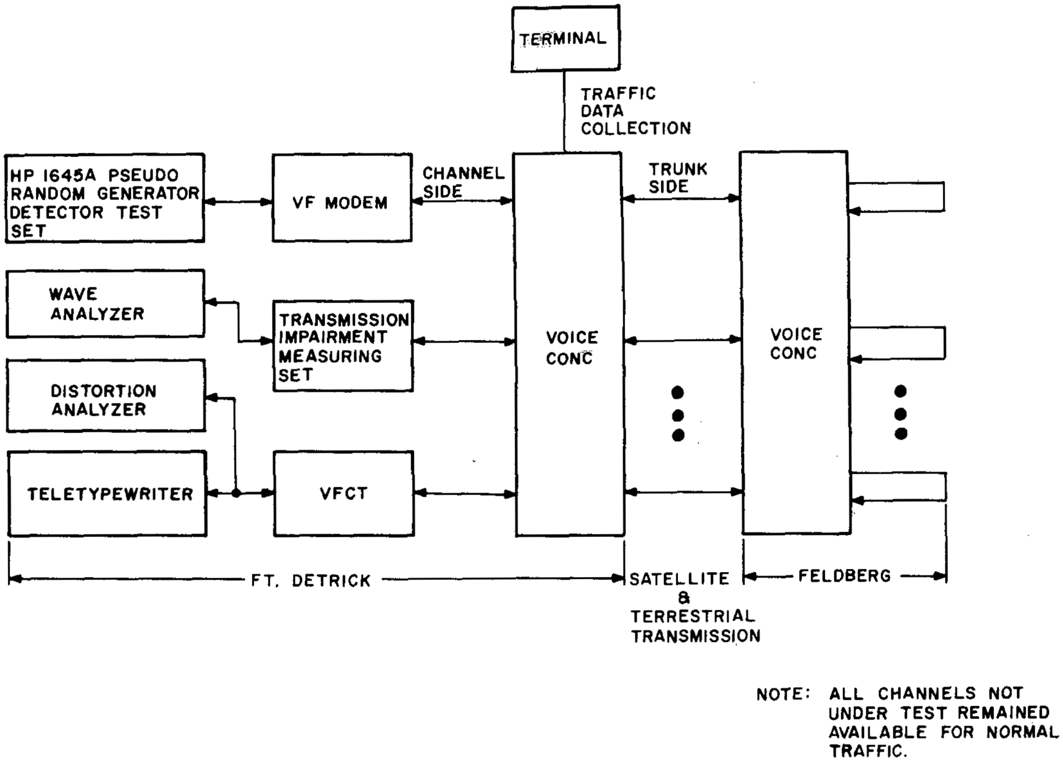

Fig. 3. Test configuration for voice concentrator.

TABLE I
CHANNEL PARAMETER MEASUREMENTS FOR OPERATION THROUGH VOICE CONCENTRATOR (FELDBERG TO FT. DETRICK)

CHANNEL NUMBER	SNR (dB)	FREQUENCY RESPONSE[1] (dB)		ENVELOPE DELAY[2] (MICROSECONDS)		HARMONIC DISTORTION[3] (dBm0)			PEAK TO PEAK PHASE JITTER (DEGREES)
		300 3000 Hz	700 -2300 Hz	500 - 2800 Hz	1000 - 2400 Hz	1400 Hz	2100 Hz	2800 Hz	
0	18	-0.2 to +4.1	-0.2 to +0.2	1788	452	-56	-64	-67	7.2
1	18	-1.5 to +4.6	-1.2 to +0.3	831	263	-54	-64	-65	4
2	18	-1.2 to +1.0	0.0 to -0.4	1318	463	-50	-54	-65	5.8
3	18	-3.8 to +0.5	+0.1 to +0.4	1625	519	-50	-65	-70	4.9
4	18	-0.3 to +5.0	-0.3 to 0.0	881	434	-52	-50	-60	4
5	18	-4.9 to +0.8	0.0 to +0.8	841	250	-55	-63	-70	4
6	19	-0.4 to +1.6	-0.3 to +0.3	1425	350	-50	-65	-63	4
7	19	-0.4 to +2.3	-0.3 to +0.2	1064	336	-43	-62	-65	7
8	18	-0.4 to +1.7	-0.3 to 0.0	903	350	-45	-60	-66	4.1
9	18	-0.1 to +1.8	-0.1 to +0.1	891	316	-43	-50	-65	4
10	18	-0.3 to +3.1	-0.3 to 0.0	950	275	-45	-59	-62	4
11	19	-0.4 to +3.3	-0.2 to -0.4	781	310	-47	-50	-60	4
12	18	-0.3 to +2.2	-0.3 to 0.0	1081	265	-42	-50	-55	4
13	18	-0.3 to +5.7	-0.3 to 1.8	1546	436	-43	-55	-60	8
14	27	-0.7 to +2.0	-0.7 to 0.0	1058	212	-45	-59	-60	4
15	18	-0.4 to 2.0	-0.5 to 0.0	1116	301	-43	-55	-58	4
16	18	-0.9 to +5.4	-0.8 to 1.2	902	379	-42	-60	-65	7

1. Compared to measured loss at 1004 Hz
2. Referenced to minimum delay measured in the specified band
3. Test frequency was 704 Hz at -10 dBm0.

levels with tone notched out, and subtracting the two levels to arrive at the SNR in decibels.

A comparison of non-TASI versus TASI channel performance indicated that all circuit parameters' performance were essentially the same except for frequency response and envelope delay. The TASI channels' maximum and minimum frequency response curves showed 1-2 dB of degradation at the end points, about 300-500 Hz and 2800-3000 Hz, but no noticeable degradation for midband frequencies, as compared to the same curves for non-TASI trunks. The mean value curves for frequency response were essentially identical (within 1 dB across the measured band) for non-TASI and TASI channels. Envelope delay maximum, minimum, and mean curves for TASI channels showed 200-500 μs increase at the end points, about 500 Hz and 2800 Hz. For midband frequencies, between 1000 and 2500 Hz, the maximum, minimum, and mean envelope delay curves for TASI channels showed from 0 to 200 μs increase.

D. Performance Characteristics of Data Signals

Although the AUTOVON IST's under test were not specified for data, these trunks have historically experienced some small percentage of data use. In fact, voice channel tests described earlier established that these trunks, both with and without concentration, met DCA standards for data rates up to 1200 bits/s [3]. It should be noted that because of varying channel noise levels and limited statistical confidence in short-term BER measurements, the results presented here were used to determine go/no-go indications of operating modem and other data signals though the concentrator. More quantitative and conclusive data would require laboratory tests where channel conditions could be controlled.

The voice concentrator provided detection of data calls in two different ways: 1) by sensing energy in the frequency range of 2010-2240 Hz above a threshold of −30 dBm0, and 2) by sensing the total energy above 1100 Hz to be greater than the energy below 1100 Hz by at least 10 dB. In either data detection mode, the voice concentrator immediately seizes a trunk and removes the fixed and variable delay used for TASI operation, since such delays are unacceptable for data transmission. The connection remains dedicated to the trunk selected as long as constant energy is detected.

For continuous signals which do not meet either criterion for a data call, the concentrator will initially connect the signal to a trunk and then hold that signal for a period of several minutes, after which time the signal will be dropped and that channel freed to accept another call. The drop criterion is satisfied if the signal has a frequency less than 1000 Hz and energy greater than approximately −23 dBm0. The reason for terminating such a connection is to minimize the time in which an inadvertent test tone signal would occupy a trunk and reduce the number of trunks available for TASI.

Fig. 3 shows the test configuration utilized for both VF modem and voice frequency carrier telegraph (VFCT) testing. Testing was accomplished by transmitting a test signal into the modem (or VFCT), through the TASI channel, and over satellite and terrestrial links to Feldberg. At the far end, the quasi-analog signal was looped and transmitted back over the satellite and terrestrial links to the near-end site, Ft. Detrick.

TABLE II
MODEL BIT ERROR RATE PERFORMANCE WITH AND WITHOUT VOICE CONCENTRATOR

Modem	Bit Rate	Bit Error Rate With Concentrator	Bit Error Rate Without Concentrator
MD-674	300 bps	8.3×10^{-4}	2.7×10^{-5}
MD-701	1200 bps	4.0×10^{-5}	3.4×10^{-6}
Paradyne	7200 bps 9600 bps	2.8×10^{-5} 3.8×10^{-5}	1.3×10^{-5} 1.8×10^{-5}

NOTE: Nominal -13 dBm Transmit Level used for all tests.

1) VFCT Test Results: Testing of VFCT's indicated that the voice concentrator generally failed to recognize fully loaded VFCT's as a data call and dropped these connections within a few minute period. VFCT's did not meet the criteria for data call recognition because: a) significant energy exists below 1100 Hz, and b) energy levels in the frequency band 2010-2240 Hz are insufficient.

2) Modem Test Results: Testing consisted of operating voice channel modems through the voice concentrator and measuring bit error rates over 10-15 min averaging periods. Modems tested were the Codex LSI 9600, Paradyne LSI 9600, Lenkurt MD 701, and Stelma MD 674. As shown in Fig. 3, bit error rates were measured by using the HP 1645A test set which provided generation and detection of a selectable length, pseudorandom bit pattern. Performance of each modem was characterized for several transmission conditions, as described below.

a) Effects of Voice Concentrator: Table II indicates results of BER testing of each modem, at the bit rate specified, for two test configurations: i) with the modem operating through a TASI-derived trunk as shown in Fig. 3, and ii) with the concentrator removed from the test configuration and the modem operating directly into a PCM derived trunk. For the former test, TASI trunks selected for modem testing were those operating into the PCM voice channels (seven of nine facilities operated through a PCM channel bank). In this way, background noise was constant for all channels, allowing a fairer comparison of modem performance with and without the concentrator. The difference in BER performance is noted to be relatively small for each modem, although a slight degradation in BER was observed for modems operating through the concentrator. This degradation is perhaps explained by the slight increase in envelope delay distortion of the TASI-derived channel compared to a PCM-derived channel. However, even with the test conditions described above, slight variations in channel background noise may have been the major contributor to bit errors.

b) Sensitivity to Transmit Level: In this test, BER performance of each modem operating over the concentrator was measured as a function of modem transmit level. As shown in Fig. 4, the transmit levels were varied from a nominal −13 dBm to 10 dB below nominal. Resulting BER's indicated a slight degradation in BER for a 2-6 dB reduction in transmit level and more significant degradation (approaching one order of magnitude) for 8-10 dB of reduced level. This same test

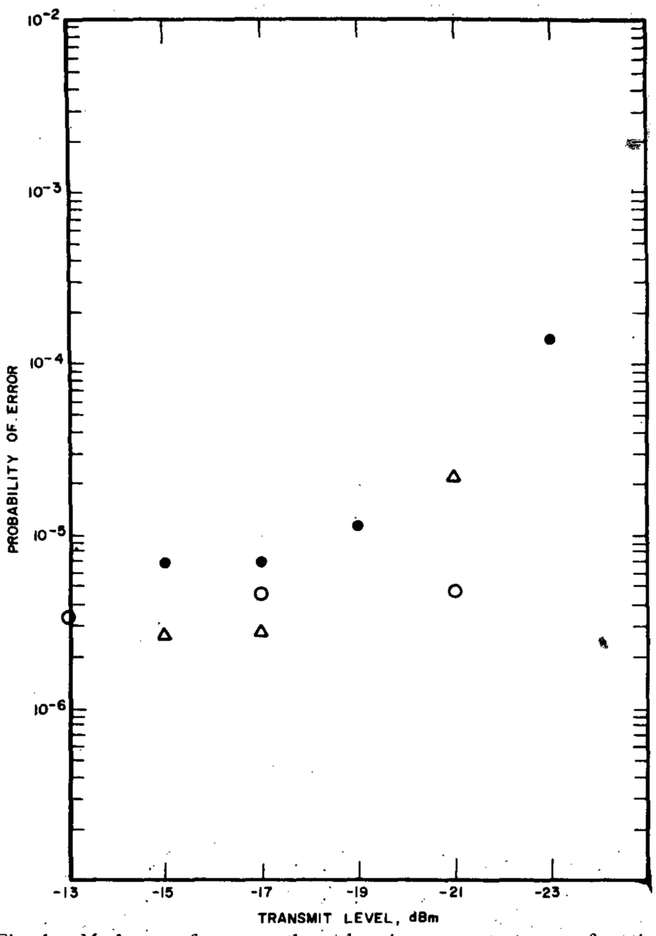

Fig. 4. Modem performance through voice concentrator as a function of transmit level. ● Paradyne 9600 bits/s, △ Codex 2400 bits/s, ○ Lenkurt 1200 bits/s.

conducted without the concentrator resulted in the same relative degradation in BER with reduced transmit levels, but with a shift in BER as predicted by Table II.

c) Sensitivity to Frequency: As a final test, an oscillator was used to input a frequency varying signal to determine at what frequencies the concentrator would recognize a data call. Starting at a test tone of 1 kHz at −16 dBm, the frequency was slowly increased and the following was noted.

i) At 1060 Hz, data call criterion 2 (energy above 1100 Hz) was recognized.

ii) At 1930 Hz, data call criterion 1 (energy in 2010-2240 Hz range) was recognized.

Then, starting at a test tone of 3 kHz at −16 dBm, the frequency was slowly decreased and the following was noted.

iii) At 3000 Hz, criterion 2 was recognized.

iv) At 2300 Hz, criterion 1 was recognized.

The above data indicate the upper and lower frequencies at which the data call criteria were satisfied.

E. Traffic Data Analysis

This section describes results of statistical analysis applied to traffic data collected during a 30 day test period in October 1979 (October is one of the busiest months for AUTOVON usage). Traffic data for the 17 channels were collected via the voice concentrator automatic reporting system, which was connected to a terminal for direct printout and key-in capability. This reporting system automatically provided speech call, data call, and total call statistics on an hourly basis.

1) CCS Statistics: Fig. 5 is a histogram of total CCS (hundred call seconds) for all 17 channels averaged on a per-hour basis for the 30 day test period. The busy hours were as expected, between 1200 and 1700 Zulu, where office hours overlapped between CONUS and Europe. With a maximum CCS per hour of (17 channels) × (36 CCS/channel) = 612 CCS, it is readily apparent that these 17 channels were heavily used, especially during busy hours. For those working days of the 30 day test, the busiest hour was 1500-1600 Zulu, with a an average CCS of 553. Total data call CCS statistics were also collected for each hour. As expected, data call activity through these 17 channels was quite small, since these circuits are not normally used for data calls. Maximum data call activity occurred during the busy hours of approximately 1200-1600 Zulu, with a range of 3-6 CCS per hour.

2) Speech Loss Statistics: Speech loss will occur in any TASI device when speech activity per channel becomes excessively high and when the number of channels simultaneously having this high speech activity becomes excessive, where

$$\text{percent speech activity} = \frac{\text{speech CCS}}{\text{total CCS}} \times 100. \qquad (1)$$

When this condition arises, speech loss can occur due to the dropping of speech samples or the lack of buffer space for storage of speech samples. Analysis of this speech loss information first required the summing of buffer and speech sample loss in seconds to arrive at the total speech loss for all 17 channels for each hour of the 30 day test period. The total speech loss per hour was then divided by total speech seconds for that hour to arrive at percentage of total speech lost:

$$\text{percent speech loss} = \frac{\text{speech loss in seconds}}{\text{speech CCS} \times 100} \times 100. \qquad (2)$$

The first presentation of these data is the histogram of Fig. 6, which shows the total speech loss for all channels averaged on a per hour basis for the 30 day period. Note that, as expected, the maximum speech loss occurred during the busy hours 1200-1700 Zulu. The second presentation of these data is given in Fig. 7, which gives the distribution of speech loss percentage, again for all channels averaged on a per-hour basis over the 30 day test period. Note that 99.2 percent of all hours had a speech loss percentage of less than 1.0 percent and 74 percent of all hours had no speech loss.

3) Blocking Statistics: Blocking of incoming channels occurs in a TASI system when existing loading indicates that additional channels, if accepted, would experience speech loss greater than the preset threshold. Calls are prevented from accessing the voice concentrator by indicating a busy tone on the channel(s) to be blocked. The actual speech activity, number of data calls, and number of trunks out of service are all used to determine when blocking (overload) conditions exist. The activation of blocking in no way affects existing calls, but rather acts to maintain voice quality on existing connections.

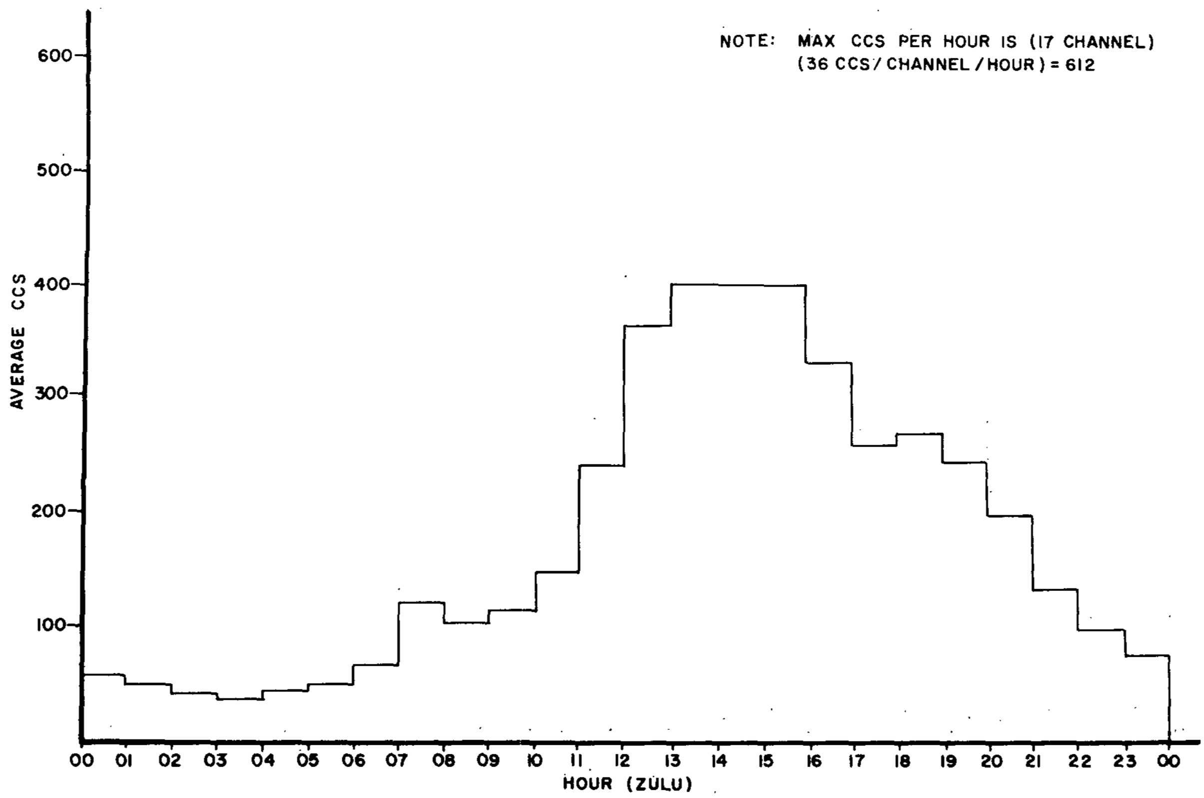

Fig. 5. Histogram of average CCS per hour.

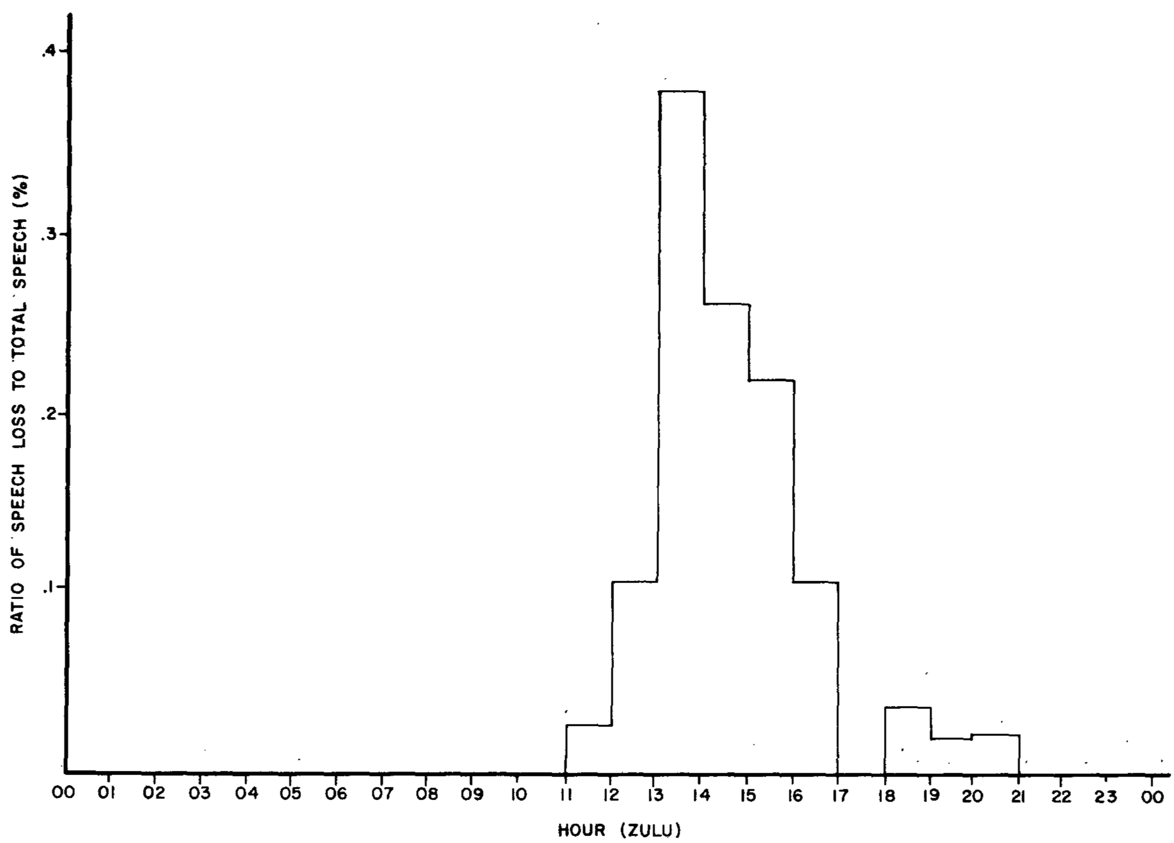

Fig. 6. Histogram of average speech loss per hour.

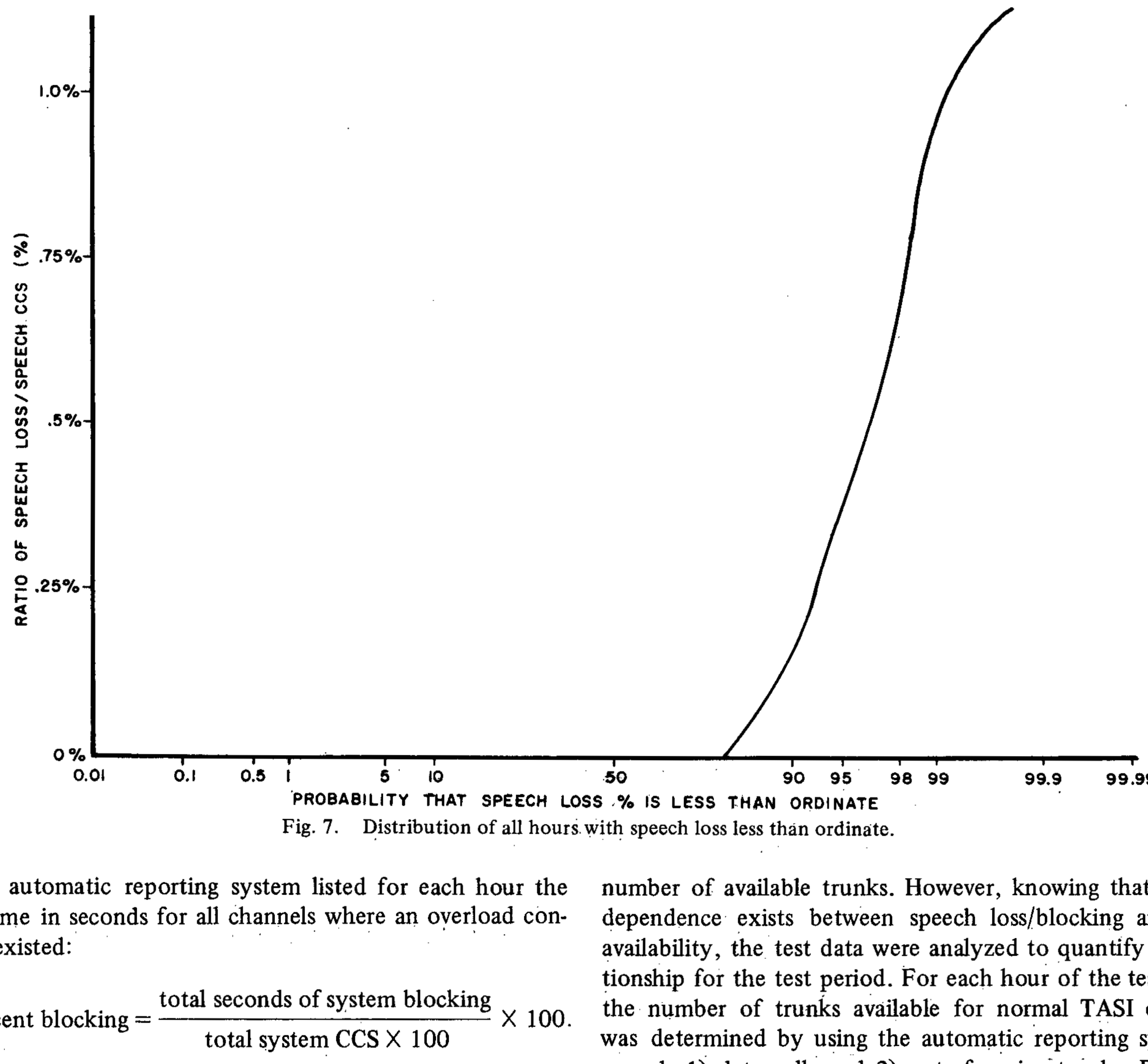

Fig. 7. Distribution of all hours with speech loss less than ordinate.

The automatic reporting system listed for each hour the total time in seconds for all channels where an overload condition existed:

$$\text{percent blocking} = \frac{\text{total seconds of system blocking}}{\text{total system CCS} \times 100} \times 100. \tag{3}$$

Fig. 8 is a histogram of percentage of total blocking time for all channels averaged over each hour of the 30 day test period. The only significant blocking is indicated over the three hour period 1200–1500 Zulu, which coincides with the busy hours on these CONUS-Europe trunks. Fig. 9 provides the distribution of blocking percentages for all channels averaged over each hour of the 30 day test period. This curve indicates that the blocking percentage is less than 10 percent for 90 percent of all hours tested, and that no blocking occurred for 75 percent of all hours. Because of three hours which showed excessive blocking (greater than 50 percent), the curve shows a steep slope beyond the 99 percent point.

4) Effect of Lost Trunks on Blocking and Speech Loss: Another means of characterizing blocking and speech loss performance is as a function of the number of total trunks available for processing calls. Data calls and out-of-service trunks will reduce the number of trunks available for TASI operation. As the number of trunks is reduced, an increase in blocking can be expected, and an increase in speech loss is also suspected. The overall algorithm used to drop speech and block channels is complex and depends on more than just the number of available trunks. However, knowing that a direct dependence exists between speech loss/blocking and trunk availability, the test data were analyzed to quantify this relationship for the test period. For each hour of the test period, the number of trunks available for normal TASI operation was determined by using the automatic reporting system to record: 1) data calls and 2) out-of-service trunks. Because a trunk is dedicated full period for each data call, the number of trunks lost due to data calls was immediately known. It should be noted that both routine operational data calls and modem/VFCT testing conducted under this test program reduced the number of available trunks. Out-of-service trunks were recorded by an alarm report, which provided a printout of various alarm conditions including failed trunks. Out-of-service conditions occurred due to both equipment failure and high noise on a particular trunk(s). For each trunk outage, the automatic reporting system provided the time of outage, type failure or degradation, and time of trunk restoration.

Figs. 10 and 11 indicate the effect of the number of trunks lost on total channel blocking and speech loss percentages, respectively. As indicated in each figure, there were four different data points observed during the test period, corresponding to the following.

Data Point	Number Trunks Lost	Number Hours Observed
1	1	27
2	2	9
3	3	3
4	7	3

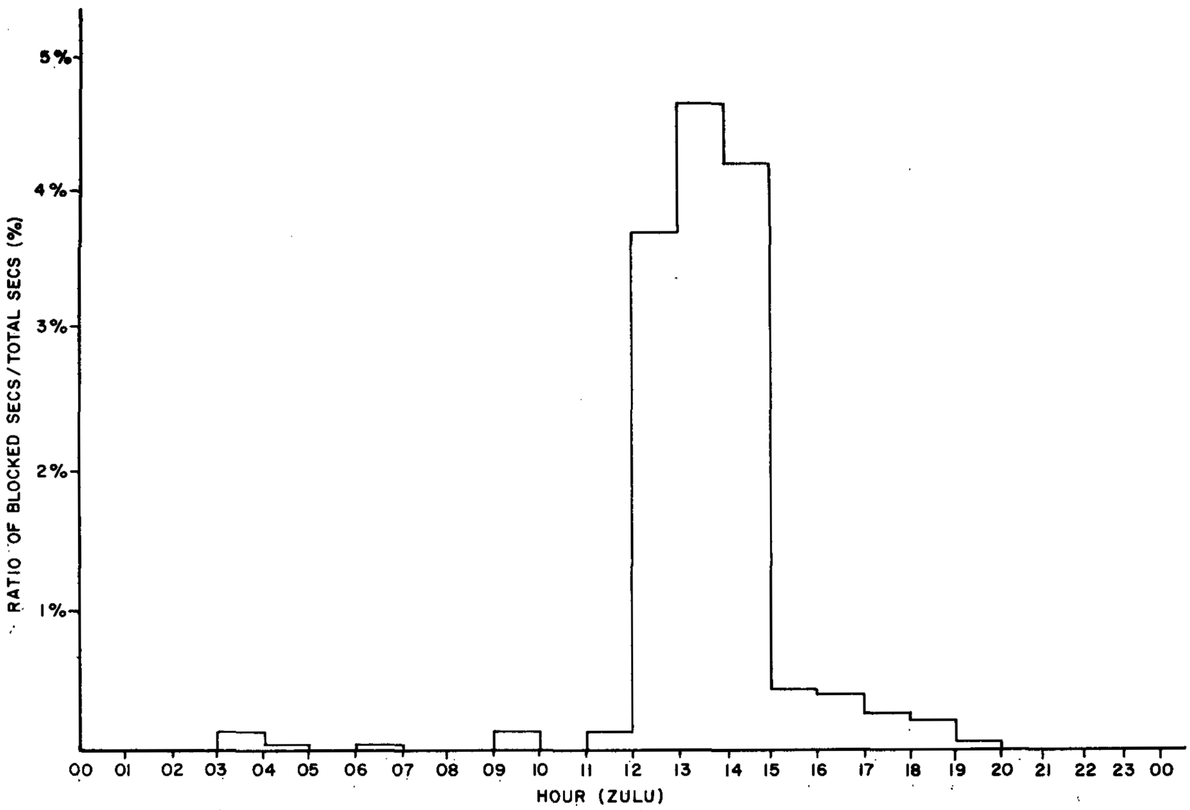

Fig. 8. Histogram of average blocking per hour.

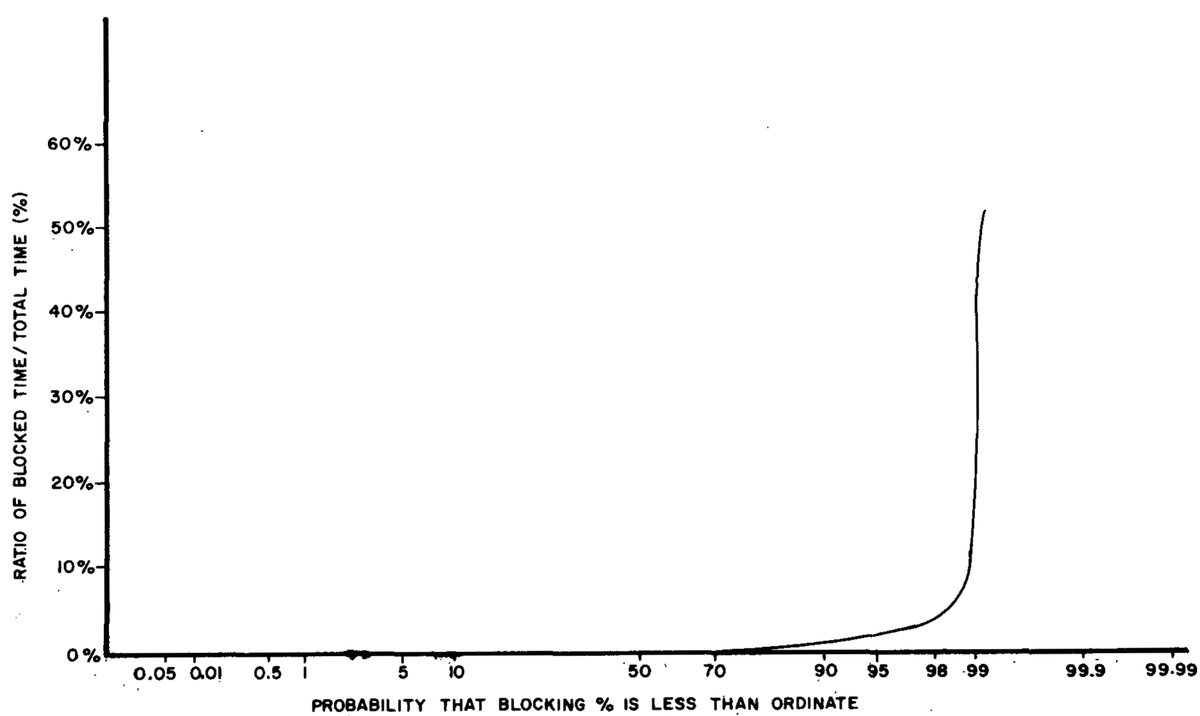

Fig. 9. Distribution of all hours with blocking percentage less than ordinate.

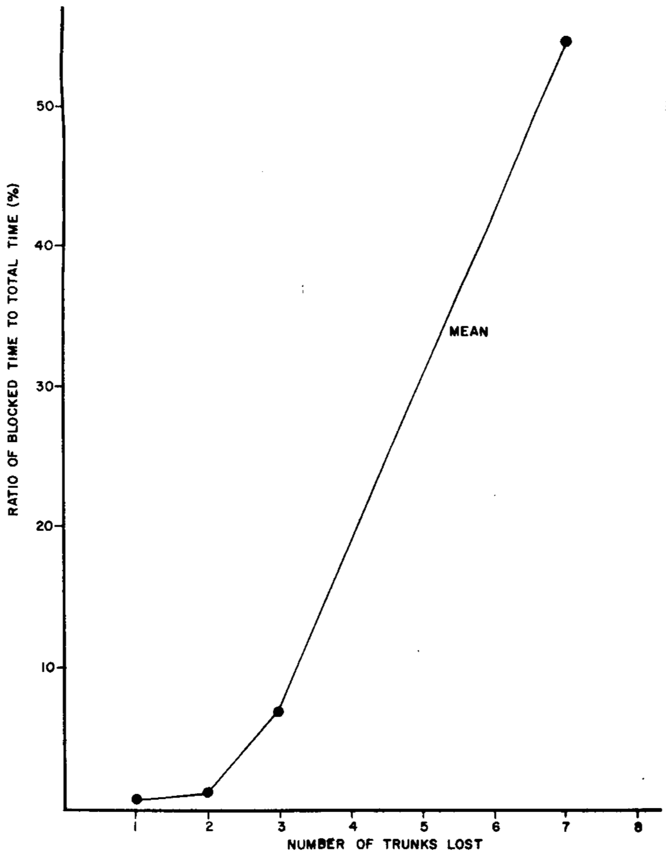

Fig. 10. Effect of number of trunks lost (out of nine) on blocking probabilities.

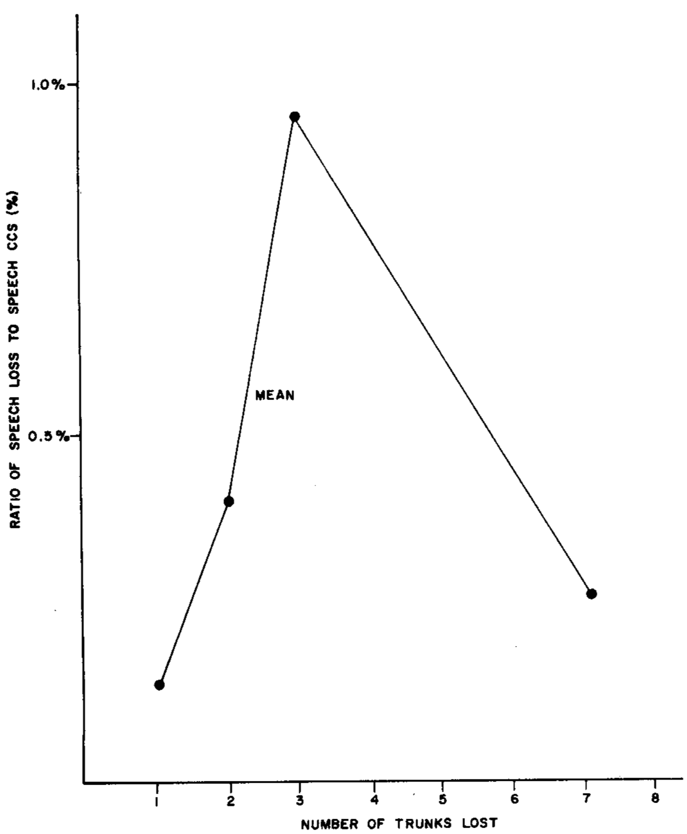

Fig. 11. Effect of number of trunks lost (out of nine) on speech loss.

Operational Test

AUTOVON USER REPORT SHEET

Using Area Code 319 will trunk your CONUS AUTOVON calls over a special trunk group designed to improve AUTOVON service. This special trunk group is under evaluation. Please list and evaluate each 319 call.

Date: Time:

Loud Intense	( )	( )	( )	( )	( )	Soft Mild
Continuous Sustained	( )	( )	( )	( )	( )	Intermittent Clipped
Natural Familiar	( )	( )	( )	( )	( )	Unnatural Foreign
Pleasant Pleasing	( )	( )	( )	( )	( )	Annoying Irritating
Intelligible Clear	( )	( )	( )	( )	( )	Unintelligible Hazy
Steady Stable	( )	( )	( )	( )	( )	Fluttering Unstable
No Interference	( )	( )	( )	( )	( )	Interference Crosstalk
No Echo Instantaneous	( )	( )	( )	( )	( )	Echo
Active Brisk	( )	( )	( )	( )	( )	Passive Dragging

How would you rate this system on a 100 point scale of overall acceptability? .... ______

Fig. 12. User subjective evaluation form.

For each data point, the mean was calculated as shown in the appropriate figure. A straight line approximation has been drawn between the third and fourth data points. As expected, there existed a direct relationship between blocking and the number of lost trunks. With seven trunks lost, there existed only two trunks available for TASI, and the mean blocking percentage exceeded 50 percent. With two trunks lost (seven trunks available), the mean blocking percentage was less than 2 percent. As shown in Fig. 10, the relationship between speech loss and number of lost trunks was not as direct. This may be explained by the fact that these data do not take into account the number of active channels or the speech activity factor during periods of lost trunks, leading one to conclude that speech loss is more dependent on total channel activity than on the number of available trunks.

F. User Subjective Evaluation

This test was designed to subjectively measure users' reactions to calls made through the voice concentrator. A special NYX code was programmed at European AUTOVON switches that permitted callers to select transoceanic circuits using the voice concentrator. Voice users were provided an evaluation form, shown in Fig. 12, especially designed to determine the following:

1) intelligibility
2) quality (compared to normal AUTOVON service)
3) effect of blocking and clipping
4) effect of fixed and variable processing delay.

An analysis was made of over a thousand report sheets. Based on a 100 point scale of overall acceptability, subscribers on the average rated the service as 95.

IV. CONCLUSIONS

In summary, voice channel performance, established via quality control tests, indicated that all channels met performance standards specified by DCA for AUTOVON interswitch trunks. In comparing results before and after cutover of the voice concentrator, negligible degradation of voice channel performance was observed. Tests of VFCT's indicated that the voice concentrator in general failed to properly recognize these signals as data and preempted these connections after a short period. Various modems tested provided marginally acceptable performance, both before and after cutover of the voice concentrator. However, it should be pointed out that these particular AUTOVON trunks are not specified for data-carrying capability, and that these data tests should be repeated over data grade trunks. Traffic data collection and analysis indicated that the 17 channels were heavily used as expected on those already saturated trunks. Percentages of speech loss and blocking, two potential problems with any TASI system, were quite low and judged to be acceptable. Finally, user subjective evaluations collected for over a thousand calls indicated user acceptance of the system.

Although approval has been given for AUTOVON IST's, application of the voice concentrator to other DCS networks, such as AUTOSEVOCOM or AUTOVON access lines, will require additional testing. Additionally, certain tests could not be practically accomplished in an operational configuration. A list of tests yet to be conducted but required to prove out TASI applications in general is given below.

1) Sensitivity to Channel Degradation: The effect of channel degradation on voice concentration performance would be determined by varying channel error rate for PCM transmission and signal-to-noise for FDM transmission.

2) Multiple TASI: This would determine the effect of multiple TASI operations on voice and data signal performance.

3) Speech Recognition Sensitivity: This test would determine the ability of a voice concentrator to recognize speech as a function of input level, speech burst duration, and noise level.

4) Speech Loss and Blocking: Speech loss and blocking would be determined as a function of a number of busy channels, speech activity factor per channel, and number of available trunks.

5) Signaling Characteristics: Voice concentrator transparency to other types of signaling such as DTMF or MF would be determined, with emphasis on the effects of delay introduced by buffered TASI.

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