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EVALUATION OF EXPERIMENTAL AND DEVELOPMENTAL COMMUNICATION SYSTEMS USED IN UNDERGROUND COAL MINES

PERFORMANCE OF TRAPPED-MINER COMMUNICATION SYSTEM

PREPARED FOR

UNITED STATES DEPARTMENT OF THE INTERIOR

BUREAU OF MINES

BY

ENVIRONMENTAL ACOUSTICS LABORATORY
THE PENNSYLVANIA STATE UNIVERSITY

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EVALUATION OF EXPERIMENTAL AND DEVELOPMENTAL COMMUNICATION SYSTEMS USED IN UNDERGROUND COAL MINES

Performance of Trapped - Miner Communication System

Prepared for:

United States Department of the Interior

Bureau of Mines

By the

Environmental Acoustics Laboratory

of

The Pennsylvania State University

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16. Abstract

Several evaluations were made by Modified Rhyme Test (MRT) word lists transmitted through various simulations of the Trapped-Miner Communication System being developed by the U.S. Bureau of Mines. The tests were performed in a "single blind" situation so that the evaluators would not be biased by knowing the characteristics of the system being simulated. A computer program was developed to provide a simulation of the electrical aspects of the communication system and to provide an evaluation by calculation of the articulation index (AI). The study indicates that communication through the presently proposed system will be somewhat unreliable at depths as great as 300 metre even with the system operating at maximum effeciency.

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No patentable items were produced as a result of this research.

FOREWORD

This report was prepared by The Environmental Acoustics Laboratory of The Pennsylvania State University under USBM Contract Number G 0166120. The contract was initiated under the Coal Mine Health and Safety Program, under the technical direction of PM&SRC, with Mr. H. Kenneth Sacks acting as the Technical Project Officer. Mr. Al Young was the contract administrator for the Bureau of Mines.

This report is a summary of the work recently completed as part of this contract during the period 1 April 1976 to 31 January 1977. This report was submitted by the authors on 7 February 1977.

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Table of Contents

List of Figures

List of Tables

1. Introduction

The U.S. Bureau of Mines is presently developing a baseband communication system for transmitting verbal information from the surface to miners who may become trapped underground.

All of the many communication systems investigated for this purpose involve a limitation on bandwidth, a reduction in naturalness of the transmitted speech, and/or some distortion of the speech signal waveform. Any or all of these factors affect the intelligibility of the speech transmitted through the communication systems. An important part of the evaluation of such a communication system is the measurement of the intelligibility of the speech received through the system.

Clinical methods of measuring speech intelligibility are well established for various purposes. Of these several established tests the modified rhyme test (MRT), which was conceived for testing speech intelligibility in auditoriums and classrooms, was chosen for testing underground coal mine communication systems. This test measures intelligibility in a limited vocabulary situation as would be encountered in coal mining communications.

The rationale behind the choice of this particular test is given by Michael, et al., (1976) and is reproduced as Appendix B of this report for convenience.

An indirect estimate of speech intelligibility may be made by determining the articulation index (AI) from the characteristics of the acoustic output of the communication system. The AI is calculated from the signal-to-noise ratio in each of 20 equal-articulation bands of frequency. The resulting AI can be used to predict approximately the score that should be expected from the use of the modified rhyme test. Thus, the MRT and the AI each give an independent estimate of the intelligibility of speech transmitted through the communication system.

2. Evaluation of Communication Systems by MRT Scores

2.1 Modified Rhyme Tape Format

The details of the Modified Rhyme Test are explained in the Final Report of Grant G 0155032 and in Appendix B of this report. In its original form, the word lists are supplied on six reels of magnetic tape. For the purpose of evaluation of through-the-earth communication systems for USBM, only the male voice recorded at +30 dB s/n was used. These word lists (Table I) were copied with new calibration signals on two reels of low-print tape and have been supplied to the US Bureau of Mines, Pittsburgh Mining and Safety Research Center.

These tapes start with an announcement identifying the particular word list followed by two calibration signals. The first calibration signal is 20 seconds of a 1.000 kHz pure tone recorded at 0 VU level on the tape. This tone is immediately followed by 20 seconds of speech-band noise also recorded at 0 VU ±.5VU. The speech-band noise was generated by passing pink noise through a Krohn-Hite 3750 band pass filter having 24 dB/octave slopes with -3 dB points at 355 Hz and 2828 Hz. The resulting filtered pink noise contains the three principal octaves of the speech band as defined by Fletcher (1953). Following the calibration tones and preceding the word list is a short instructional announcement copied directly from the original word list tapes. (Practice word lists were omitted from these tape copies.)

Each word list contains 50 items. The words are presented in a standard carrier phrase which includes words before and after the test word. The word lists are recorded so that the average level of the test word itself is 0 VU ±2 VU. The signal-to-noise ratio was set by the recording on the original tapes at +30 dB.

The tapes supplied to USBM for use in these evaluations were recorded on a two channel magnetic tape recorder with half-track-width heads. The two tracks each contain the same signal recorded at the same level so that the tapes can be reproduced on either a half-track or a full-track reproducer.

2.2 Reproduction of Word List Tapes

When reproducing the taped word-lists to test a communication system, the 1.000 ± .001 kHz calibration tone is provided to set signal level and to check overall speed accuracy. The noise calibration provides a broad band signal that can be used to determine the bandwidth of the system relative to the speech band by comparing the level of the noise calibration signal with the level of the pure tone calibration signal after they have been transmitted through the system. Allowance must be made in the electronic systems for occasional signal peaks in the word lists which may rise to levels of +4 to +5 VU to avoid problems of waveform distortion.

2.3 Procedures

Tests of through-the-earth communication systems were planned that involved a series of tests at actual coal mine sites by USBM personnel with technical assitance from EAL. Word list tapes supplied by EAL were to be transmitted over the communication system and received underground. The audio signals from the underground receiver were to be recorded in the mine on magnetic tape and then returned to EAL for evaluation. However, in lieu of actual field tests, the program was changed to develop a computer model and to test some PMSRC simulations of the through-the-earth voice communication system.

Tape recordings of the audio output of these simulated communication systems were scored at EAL for intelligibility by a group of 10 trained listeners, all having normal hearing sensitivity and ranging in age from 22-28 years. Listeners were seated in a standard audiometric test room which has a reverberation decay time comparable to that of a coal mine entry. Each subject was seated in approximately the center of the test room one meter from the speaker through which the MRT test words were reproduced. The speaker was placed at 45° azimuth in relation to the subject's head. The speaker and amplifier used are described in section 3 of this report.

Only two of the word lists from any given test recording were scored. To make a statistical comparison of the subjects' performance on the two test lists, the order of the test list presentation was altered so that half the subjects heard one list first and the other half heard the other list first.

The initial calibration of the reproduced sound level was accomplished by using a General Radio Type 1565-A sound level meter with C frequency weighting and slow meter response to set the sound level of the test words to 65 dB at the position of listener's ear with the listener absent. The voltage of the calibration tone at the loudspeaker voice-coil was measured on a Hewlett-Packard Model 403B Voltmeter. This voltage was recorded on the score data sheet. Thereafter, the voltage of the calibration tone was used to set the acoustic signal level for each test presentation.

2.4 Results

Test scores for the communication systems tested at PMSRC are given in Tables II, III, IV, V, VI, and VII. The type of signal processing used with tapes RCB-4, 5, 6, 7, 8, and 9 was not revealed to EAL so no evaluation was attempted. Table VIII shows the results of a simulation performed at EAL. The theoretical input filter response curve corresponding to SP-9 is shown in Figure 1. Figure 2 shows the acoustic signal spectrum plotted relative to the normalized noise spectrum at the loudspeaker output for the communication system with signal processing filter SP-9 used. The articulation index for this system was calculated to be .15 but because the curves showed a positive signal-to-noise ratio over a significant portion of the pass band it was decided to obtain MRT scores by re-constructing these spectra as closely as possible. The MRT test score of 68% word intelligibility (AI = .26) is higher than the value of 35% word intelligibility (AI = .15) predicted by the computer program probably because of slight errors in duplicating the required signal and noise spectra.*

The last two tapes analyzed, RCB-10 and 11, (Tables IX and X) were prepared by USBM to determine the effect of compression on the MRT scores. Although the standard deviations of the scores for RCB-11 are slightly greater in general than those for RCB-10, there appears to be no significant difference between compression and no compression as far as word scores are concerned. The major improvement that should be expected from the use of compression is a slight improvement in signal-to-noise ratio that results from a higher average signal energy in the transmit loop while staying within the peak current limit.

3. Low-Power Amplifier

The through-the-earth communication system involves a man-carried amplifier and loudspeaker which is powered by the miner's cap-lamp battery and is worn on his safety belt.

In anticipation of the use of the man-carried loudspeakers to reproduce the test words in our audiometric test suite, a 100 milliwatt (maximum) power amplifier (Figure 3) was built to reduce the possibility of accidental burnout of these small speakers. This amplifier is adequate to produce

*See page 3a.

Note: The AI quoted for SP-9 was obtained with the original computer program in which transmit power was calculated with the formula $P = \frac{E^2}{z}$. This example was included primarily to illustrate the differences that can arise between test word scores predicted from the AI and those actually measured.

Parameter for SP-9 are as follows:

Radius of XMIT Loop	2500 cm
Wire Diameter	0.2053 cm
Inductance	0.000328 henrys
Resistance	0.8184 ohms
Capacitance	10^6 farads

65 dB sound pressure level (conversational level) at one meter distance from the loudspeaker. The man-carried speakers will be used with tapes recorded in actual coal mine situations to provide an overall test of USBM experimental communication systems. Preliminary simulation tapes produced in the laboratory at PMSRC are being reproduced through this amplifier and a Philips 5060M8 wide-range speaker mounted in an infinite baffle.

4. Computer Simulation

4.1 Calculations to be performed

An Articulation Index Computer program was designed to simulate the proposed Bureau of Mines communication system. Based on the 20 band method for the calculation of Articulation Index (AI), this program proceeds through all calculations of speech and noise levels in the 20 separate bands whose mid frequencies are given in Table 8 of ANSI 3.5-1969. All variables listed with a subscript (I) are thus indexed for bands 1 through 20.

The program processes a speech spectrum (Figure 4) and Audio noise spectrum of 20 bands each through a signal processing filter, an Audio Power Amplifier, the "earth attenuation" and the receiver system (Fig. 5). Currents for speech and noise are determined as they appear at the transmit loop. From these currents, the total power for transmission is calculated. At this point, the program checks that the power is less than or equal to 500 watts and that the total peak current is equal to or less than the maximum allowable peak current. Any necessary adjustments are performed. The open circuit voltages induced in the receiving loop are then calculated for speech and noise in each of the 20 bands and attenuated by the earth filter response (Figure 6). (For other power limits, see Note 1, p. 26.)

After electromagnetic noise in the mine (Fig. 7) induced in the receiving loop is added into the noise array, the signals are then combined with the electronic noise of the receiver and amplified. Here, also, the speech and noise arrays are adjusted for the band pass of the receiver (Figure 8). The total power appearing at the loudspeaker is then adjusted to .01 watts simulating the action of the automatic gain control. The 20 band arrays of speech and noise are converted through the speaker response spectrum (Figure 9) to sound pressure levels ref. .0002 n/m^2 at .5 metre. The AI is then calculated as specified in the ANSI standard.

4.2 Description of Program

The initial input data, for the calculation of electrical parameters, i.e., the circumference of the transmit loop (CT), wire resistivity (RHO) and wire diameter (DIM) are entered. The radius of the transmit loop (RA), wire diameter in cms (D), inductance (L) and wire resistance (R) are immediately calculated by the computer* (eqs 1-4, Table XI). If the capacitance (C) is to be taken in account, it is entered in Farads directly into the program after the comment card 'calculate C'. When no resonating capacitor is used, the value $C=10^6$ is entered. This value is

* See Table VII Equations used in Computer Program

presently in the program. RA, D, L, R, and C are printed output. The impedance (Z(I)) is calculated using equation (5) and printed alongside the center frequencies for each of the bands.

For the spectral shape of the speech signal, data from the idealized speech spectrum for male voices (SPHSPM(I)) is taken from Table 8, column (a) of ANSI S3.5-1969. Signal processing (SIGPRO(I)), bandwidths (BNDWTH(I)), acoustic surface noise (ACSFNO(I)), circumference of the receiving loop (CR), earth filter attenuation (ATTNUA(I)), maximum peak signal voltage (PEAKE), and maximum peak signal current (PEAKI) are read into the computer. The Peak and rms values are related to equations 6a and 6b. The earth attenuation is based on a value of 300 metres depth and 0.1 MHOs/metre conductivity but other cases can be used by changing the appropriate data cards. The maximum peak signal current of 19.1 amps and maximum peak signal voltage of 76.4 volts used in the program are based on the Phase Linear amplification specifications. These are also easily changed to allow introduction of loop matching transformers by inserting new data cards.

The ACSFNO presently used in this program is chosen 20 dB below the level of the idealized speech spectrum in each of the 20 bands. This choice is based on the following assumptions:

1. Airborne acoustic surface noise will probably be composed mainly of speech from other persons nearby and, hence, will have the same spectrum characteristic.

2. Data supplied by makers of noise—cancelling microphones (Telex, RCA, Roanwell) indicate that about 20 dB signal to ambient noise can be realized when using these microphones.

3. Airborne acoustic ambient noise of this type will dominate all other system noises in the surface transmitting system.

To determine the signals to be transmitted, SPHSPM(I) and ACSFNO(I) are processed through addition of the signal processing (e.q. 7a, b). The two resulting 20 band arrays are normalized to the maximum value in the resulting speech spectrum, and are then converted to decimal attenuations. (e.q. 8a, b). The transmit currents for speech (XMITSP(I)) and noise (XMITNO(I)) are calculated using equations (9a) and (9b) respectively. After determining the total current from both 20 band arrays (TOTALI), the power through the resistor (POWRES) is calculated from equation (10).

Next, the program goes through a number of check points. If POWRES is greater than 500 watts, the currents are adjusted to bring the power down to 500 watts (e.q. 11, 12a, b). If POWRES is below 500 watts, the program continues to the next check point. Here, the program checks to insure that the total peak current (TOTALI * 1.414) does not exceed PEAKI. When the total current is excessive, the currents for speech and noise are adjusted down (e.q. 13, 14a, b) so that the total peak current equals PEAKI (e.q. 15). POWRES is then recalculated. If the total peak current is within the allowable limit, the program continues.

The open circuit voltage induced in the receiving loop antenna is expressed by equation 16 taken from the Collins Phase I (1975) report.

After substituting known values and simplifying the open circuit voltage is expressed in terms of input data in equation (16b). Equation (16b) is converted to decibel notation as shown in equation (17). The vertical magnetic field component at the receiving loop is expressed in equation (18) in terms of the transmitting loop area A, the transmit loop current I, and the earth attenuation. In decibel notation, equation (18) becomes equation (19). Values for ATTNUA are read in dB either from theoretical calculations (see Figure 6) or from actual measurement.

The transmission of the signal levels in each band (XMITSP(I)) and XMITNO(I)) is accomplished by the use of equation (20) and (21) for the loop-earth atten.-receiver system. (Equations 20 and 21 are particular cases of equation 17 which was derived from equations 16, a and b). The electromagnetic noise in the mine (THATNO(I)) is entered and processed through the receiver system by equation (22). All signals received are modified by the relative receiver response (RERERE) (e.q. 23a, b, c). The electromagnetic noise (ELMGNO(I)) resulting from THATNO(I) is added to the open circuit voltage noise (VOCNOI(I)) by equation (24) resulting in SBTNO1(I) in watts.

The noise due to the amplifer is next calculated and modified by RERERE and added to SBTNO1(I).

The electronic noise voltage is expected to be approximately 10 dB below 20 μ volts at the input to the receiver. It is assumed that this noise is "white" in nature and is therefore constant energy per cycle. Since this electronic noise must be combined with all other noise and signal energies in each of the 20 bands, it is necessary to express each quantity in terms of power so that the power contribution from each source can simply be added together. In doing so, all voltage levels are referenced to 1 volt and an impedance of unity is arbitrarily assumed.

The electronic noise power in band (I), designated ELNOPO(I) in the program is then:

$$\text{ELNOPO(I)} = \frac{(\text{ELNVOC})^2}{\text{R}} \text{ X } \frac{\text{BNDWTH(I)}}{\text{Total Bandwidth}}$$

where: ELNVOC is the equivalent noise voltage at the receiver input.* Since R is assumed to be one ohm for convenience in conversion, and since the total bandwidth of interest is the combined width of all 20 equal articulation bands, the electronic noise power in band (I) can then be calculated:

$$\text{ELNOPO(I)} = \frac{(20 \text{ X } 10^{-6} \text{ X } .3162)^2}{1} \text{ X } \frac{\text{BNDWTH(I)}}{(6000-200)}$$

After simplifying, this equation becomes:

$$\text{ELNOPO(I)} = 6.897 \text{ X } 10^{-15} \text{ X BNDWTH(I)} \tag{25}$$

ELNOPO(I) is then added to SBTNO1(I) (e.q. 26) resulting in SBTNO2(I). POWSPH(I) is also converted to a power for each band (e.q. 27).

*See note 2, page 25.

The speech and noise voltages appearing at the terminals of the receiver loop are then amplified by the 93 dB gain of the receiver (e.q. 28a, b). A correction factor (CORFC2) is then calculated (e.q. 30) and used to adjust the total power delivered to the receiver loudspeaker to .01 watts (e.q. 29-31b). As a check on system operation, this correction factor should be between 1.00 and $.16 \times 10^{-6}$ representing the action of the automatic gain control system. The voltage levels are then re-calculated in dB re 1 volt for the speech and noise spectra (e.q. 32-33). These final voltage levels are then connected (e.q. 34a, b) to acoustic sound pressure levels by using the speaker response data (SPKSPC(I)), see Figure (9). The resulting sound pressure levels for speech and noise in each of the 20 bands are then printed out. For convenience in comparing system performance, these spectra are then normalized to band 3 of the noise spectrum since the noise spectrum is relatively stable as the signal processing is changed. Band 3 is chosen as the reference point because it represents the center frequency of the lowest frequency band passed by the receiver system.

In the Articulation Index procedure, spectrum levels for noise and speech are first calculated (e.q. 35a, b) and a 12 dB peak correction is added to each of the speech spectrum levels (e.q. 36). The noise spectrum levels are used as the effective masking spectrum. Corrected noise or spread of masking adjustments are not necessary as the band sensation levels of the noise received will not approach 80 dB under any reasonable circumstances. Nor will the spread of masking spectrum cause any changes in the received noise spectrum. The AI is then calculated by finding the 20 differences (e.q. 37-39), summing, and dividing by 600 (e.q. 40). Finally, a data sheet is printed at the end of the program, summarizing all of the card input data.

The format of the data cards is shown in Table XII for reference. Table XIII lists and identifies the variables used in the computer program.

5. Interpretation of Articulation Index

A brief discussion of the articulation index is presented in the ANSI Standard S3.5-1969, however, the interpretation must still be made with consideration for the unique situation presented in attempting communication with a trapped miner. Obviously, in this situation the listener is highly motivated to try to understand what is being said.

It is emphasized in the Standard that it is not possible to determine a single AI which would be a true criterion for acceptable communications. The necessary score is dependent on the context and the ultimate goal. In our clinical experience, an AI of .70 gives 98% discrimination of sentences upon first presentation. This score (AI = .70) may be considered the lower limit for desirable conversational conditions.

An AI of .40 is generally accepted as the minimum level for barely intelligible conversations. When the AI is as low as .25, a person can hear that someone is talking but will have considerable difficulty discriminating what is said. In office spaces, this AI of .25 is barely acceptable as a condition for privacy. At an AI of .05 a person may even have difficulty recognizing the fact that someone is talking to him. Such a condition results in very good privacy.

Considering the acoustical conditions of a mine entry and the motivational state of a trapped miner, it would appear than that an AI somewhere between .40 and .25 might be considered as a cut-off level for barely acceptable communication with a trapped miner.

The task of communication with a trapped miner requires a consideration of many aspects which make this situation unique. The motivational state of the miner has already been mentioned. The limited vocabulary involved may be an advantage in some cases although the particular instructions to be transmitted may not represent routine or familiar communications. Message intelligibility is adversely affected in any communication system when the listener has a significant amount of sensori-neural hearing loss, which is indeed the case for many miners. This problem is discussed in section 7 of this report. The intelligibility of the information transmitted is also affected by the pronunciation and inflection of the talker.

While performance of the trapped-miner communication system can be estimated by calculation of the articulation index, intelligibility of information transmitted can best be evaluated by direct measurement. One such direct measurement system, the modified rhyme test, has already been described elsewhere in this report. This test makes use of a limited vocabulary by requiring a forced-choice selection of the words transmitted.

The actual performance of the trapped-miner communication system ~~will~~, therefore, not be reflected in the AI and indeed for an AI between .25 and .4, many questions still remain as to the actual performance in terms of information transmission. While the modified rhyme test gives a more direct measure of word intelligibility, it does not evaluate information transmission ability. A suitable test could be developed to measure accuracy of information transmission. This test would be based on typical words and phrases that might be transmitted to a trapped miner in a real situation. This type of test would measure the amount of information transmitted by requiring the listener to perform a simple task. A test of this type would provide an accurate evaluation of the communication system and would allow quality comparison of different competitive systems. For accurate evaluation of the trapped-miner communication systems it is recommended that such a test be developed and standardized for use by the U.S. Bureau of Mines.

6. Reduction of Receiver Noise

The receiver bandpass filter as presently conceived has its -3 dB points at 500 Hz and 3000 Hz as shown in Figure 8. Additional attenuation of 60 Hz interference (also shown in Figure 8) is accomplished by the inter-stage coupling capacitors. However, experience with the computer simulation program has indicated that very little energy above 2000 Hz is passed through the communication system because of the attenuation of the earth filter (Figure 6). It therefore seems logical that the receiver bandpass could be narrowed to 500 Hz to 2000 Hz to bring about a reduction in electronic noise (see Figure 10). For this communication system, the loss of information in frequencies above 2000 Hz appears to be negligible compared to the increase in AI brought about by higher signal to-noise ratios at shallow depths. The noise reduction to be expected from this narrower bandwidth is:

$$\text{Noise Reduction in dB} = 10 \log \frac{BWn}{BWw}$$

$$= 10 \log \frac{2000-500}{3000-500} = 10 \log \frac{1500}{2000}$$

$$= -2.2 \text{ dB}$$

7. Compensation for Hearing Loss

Noise-induced hearing loss is quite common among coal miners (Michael, et al., 1972) and it is quite logical to ask if some signal compensation can be included to assist those persons with hearing loss. To answer this question it should first be pointed out that the major amount of hearing loss due to noise exposure occurs at frequencies above 2000 Hz. Although there is always a certain amount of hearing loss below 2000 Hz there is little hope that any kind of compensation can be applied in this frequency region, that would enhance speech intelligibility for the hard-of-hearing. Since frequencies above 2000 Hz are not passed by the communication system, it seems that compensation is not practical.

The problem that will remain for the hard-of-hearing person is the loss in ability to discriminate speech signals in the presence of noise. (This phenomenon is related to recruitment, a loss of tolerance for loud sound.) No satisfactory compensation has yet been found for speech discrimination loss and recruitment although some investigators have had some success with level compression (Villchur, 1974). Since compression has been considered for the communication system for other reasons, it may then serve a dual purpose.

Presently, the most effective way to enhance intelligibility for persons with speech discrimination loss and recruitment is to improve signal-to-noise ratio in the communication system--whether it is a telephone-type system or a hearing aid. The absolute sound pressure level of the signal reaching the person's ears is also quite critical and is best adjusted by the listener himself. In the trapped-miner communication system this will instinctively be adjusted by moving closer or farther away from the loudspeaker. Some persons may even prefer to hold the speaker up to their ear. The design goal of 10 m W average power output should be adequate to allow this range of loudness adjustment for most persons.

Table I

Modified Rhyme Lists Used for Evaluation of Communication Systems

Tape 1		Tape 2	
Form 1		Form 3	
List	A	List	F
	F		C
	B		E
	E		D

Table II

Date June 30, 1976

Evaluation of USBM Tape No. RCB-4-76-6-11 MRT form Form 3

No. of Subjects: 10

Test material: Lists C and F of the Modified Rhyme Test

Calibration Signal: 1,000 Hz. pure tone

Sound Level at Listener's position: 65 dBC_s

Sound level meter: General Radio Type 1565-A

EAL Test room: 3A

Voltmeter: Hewlett-Packard Model 403B

Voltage: .66v on the 1v scale

Speaker position: 45° azimuth in relation to listener's head

Results:

List	Mean	SD
C	68.60	7.72
F	64.60	4.43

Comments:

Although the standard deviation of list C was larger than that of list F, a T-test demonstrated that the subjects' performances on lists C and F were not significantly different at $\alpha = .05$.

Table III

Date 8/16/76

Evaluation of USBM Tape No. RCB-5-76-6-22 MRT form 1 S/N-30

No. of Subjects: 10

Test material: Lists A and F of the Modified Rhyme Test

Calibration Signal: pure tone (1,000 Hz)

Sound Level at Listener's position: 65 dBC_S

Sound level meter: General Radio Type 1565A

EAL Test room: 3A

Voltmeter: Hewlett-Packard Model 403B

Voltage: .251v on the .3v scale

Speaker position: 45 azimuth in relation to subject's head

Results:

List	Mean	SD
A	68.20	4.05
F	68.20	8.509

Comments:

It was noted that the calibration tone had wide level fluctuations. Also some subjects had great difficulty in discriminating any of the test words . When this situation occurred, the subjects were given a practice session during which the tester cued them as to when the test words were being presented.

Table IV

Date September 14, 1976

Evaluation of USBM Tape No. RCB-6-76-8-13 MRT form 1 S/N +30

No. of Subjects: 10

Test material: Lists A and F of the Modified Rhyme Test

Calibration Signal: 1,000 Hz pure tone embedded in noise

Sound Level at Listener's position: 65 dBC_s

Sound level meter: General Radio 1565 B

EAL Test room: 3A

Voltmeter: Hewlett Packard model 403B

Voltage: .36v on the .3v scale

Speaker position: 45° azimuth in relation to lestener's head

Results:

List	Mean	SD
A	70.4	5.23
F	70.2	5.45

Comments:

Subjects commented that this tape seemed more intelligible than those previously played.

Table V

Date 11/12/76

Evaluation of USBM Tape No. RCB-7-76-11-4 MRT form 1, S/B+30

No. of Subjects: 10

Test material: Lists A and F of the Modified Rhyme Test

Calibration Signal: 1,000 Hz. pure tone embedded in noise

Sound Level at
Listener's position: 65 dBC_s

Sound level meter: General Radio 1565A

EAL Test room: 3A

Voltmeter: Hewlett Packard model 403B

Voltage: .12v on the .3v scale

Speaker position: 45 degrees azimuth in relation to listener's head

Results:

List	Mean	SD
A	68.60	6.11
F	70.60	5.89

Comments:

Table VI

Date 11/13/76

Evaluation of USBM Tape No. RCB8-76-11-4 MRT form 1, S/N+30

No. of Subjects: 10

Test material: Lists A and F of the Modified Rhyme Test

Calibration Signal: 1,000Hz. pure tone embedded in noise

Sound Level at
Listener's position: 65 dBC_s

Sound level meter: General Radio model 1565A

EAL Test room: 3A

Voltmeter: Hewlett Packard model 403B

Voltage: .175v on the .3v scale

Speaker position: 45 degrees azimuth in relation to listener's head

Results:

List	Mean	SD
A	73.80	4.47
F	69.80	4.94

Comments:

Table VII

Date 11/15/76

Evaluation of USBM Tape No.RCB-9-76-11-4 MRT form 1, S/N+30

No. of Subjects: 10

Test material: Lists A and F of the Modified Rhyme Test

Calibration Signal: 1,000 Hz. pure tone

Sound Level at Listener's position: 65 dBC_s

Sound level meter: General Radio 1565A

EAL Test room: 3A

Voltmeter: Hewlett Packard model 403B

Voltage: .17v on the .3v scale

Speaker position: 45 degrees azimuth in relation to listener's head

Results:

List	Mean	SD
A	70.40	6.52
F	70.80	5.18

Comments:

Table VIII

Date 11/10/76

Evaluation of SP #9 at 300 meter depth simulation MRT form 1, S/N+30

No. of Subjects: 10

Test material: Lists A and F of the Modified Rhyme Test

Calibration Signal: 1,000 Hz. pure tone

Sound Level at Listener's position: 65 dBC_s

Sound level meter: General Radio 1565A

EAL Test room: 3A

Voltmeter: Hewlett Packard model 403B

Voltage: .67v on the 1v. scale

Speaker position: 45 degrees azimuth in relation to the listerner's head

Results:

List	Mean	SD
A	68.20	6.63
F	67.80	5.12

Comments:

Table IX

Date January 13, 1977

Evaluation of USBM Tape No. RCB-10-76-12-8 MRT form 1

No. of Subjects: 10

Test material: Lists A,F,B,E of the modified Rhyme Test.

Calibration Signal: 1,000 Hz. pure tone

Sound Level at
Listener's position: 65 dBC_s

Sound Level meter: General Radio 1565-A

EAL Test room: 3A

Voltmeter: Hewlett Packard model 403B

Voltage: .6v on the .1v scale.

Speaker position: 45 azimuth in relation to listener's head

Results:

S/N	Mean	SD
Quiet	83.2	4.82
S/N-10	75.0	3.91
S/N-15	75.2	4.34
S/N-20	70.8	7.95

Comments:

Table X

Date January 12, 1977

Evaluation of USBM Tape No. RCB-11-76-12-8 MRT form 1

No. of Subjects: 10

Test material: Lists A,F,B,E of the modified Rhyme Test.

Calibration Signal: 1,000 Hz pure tone

Sound Level at Listener's position: 65 dBC_S

Sound Level meter: General Radio 1565-A

EAL Test room: 3A

Voltmeter: Hewlett Packard model 403B

Voltage: .58v on the .1v scale.

Speaker position: 45 azimuth in relation to listener's head

Results:

S/N	Mean	SD
Quiet	84.6	5.73
S/N-10	71.8	4.66
S/N-15	74.2	8.96
S/N-20	71.0	6.81

Comments:

Table XI

Equations used in Computer Program

* signifies input data

(1) $$Ra = \frac{10^2\ Ct}{(2\pi)\ (3.2808)}$$ [3.2808 ft = 1 metre]

Ra = xmit loop radius in cm.
* Ct = circum. of xmit loop in feet

(2) $$D = (DIM)\cdot(2.54 \times 10^{-3})$$

D = wire diameter in cms (Transmit Loop)
* DIM = wire diameter in mils

(3) $$L = .01257\ Ra\ N^2\ \left[\ln\left(\frac{16\ Ra}{D}\right) - 1.75\right] \cdot 10^{-6}$$

L = inductance in Henrys (Transmit Loop)
N = 1 turn

(4) $$R = \frac{Ct\ \rho}{1000}$$

R = resistance in ohms (Transmit Loop)
* ρ = resistivity in ohms/Kft.

(5) $$Z(I) = \sqrt{R^2 + \left(\omega L - \frac{1}{\omega C}\right)^2}$$

Z(I) = impedance in ohms of band I (Transmit Loop)
ω = 2π (CTRFRQn)
* CTRFRQ(I) = center frequency of band I
* C = capacitance in FARADS

(6) a RMSE = .707 PEAKE
b RMSI = .707 PEAKI

* PEAKE = maximum peak voltage
* PEAKI = maximum peak current
RMSE = maximum rms voltage
RMSI = maximum rms current

Voltage and current limitations due to power supply.

(7)a ATTSP(1) = SPHSPM (I) + SIGPRO(I)

ATTSP(I) = processed speech spectrum of band I
* SPHSPM(I) = spectrum level in band I for speech in dB
* SIGPRO(I) = signal processing function for band I in dB

b ATTNO(I) = ACSFNO(I) + SIGPRO(I)

ATTNO(I) = processed noise spectrum of band I
* ACSFNO(I) = spectrum level in dB in band I for noise

The program here normalizes ATTSP(I) and ATTNO(I) to the maximum value in ATTSP(I)

(8)a $ATTSP(I) = 10^{\left[\frac{ATTSP(I)}{20}\right]}$

b $ATTNO(I) = 10^{\left[\frac{ATTNO(I)}{20}\right]}$

Steps 9 and 10 convert ATTSP and ATTNO to decimal attenuations

(9)a XMITSP(I) = RMSE X ATTSP(I)/Z(I)

XMITSP(I) = transmit current in band I for speech

b XMITNO(I) = RMSE X ATTNO(I)/Z(I)

XMITNO(I) = transmit current in band I for noise

(10) $POWRES = TOTALI^2 \text{ X } R$

POWRES = power through the resistor

$TOTALI = \sum_{n=1}^{20} XMITSPn + \sum_{n=1}^{20} XMITNOn$

(11) $CORFC3 = \sqrt{500/POWRES}$ (See Note 1, p. 26)

CORFC3 = correction factor for currents to prevent POWRES from exceeding 500 watts

(12)a XMITSP(I) = XMITSP(I) X CORFC3

b XMITNO(I) = XMITNO(I) X CORFC3

(13) CORFC4 = PEAKI/(TOTALI X 1.414)

CORFC4 = correction factor to adjust currents to prevent exceeding PEAKI

(14)a XMITSP(I) = XMITSP(I) X CORFC4

b XMITNO(I) = XMITNO(I) X CORFC4

(15) $$\text{PEAKI} \le \left(\sum_{I=1}^{20} \text{X MITSP(I)} + \sum_{I=1}^{20} \text{XMITNO(I)}\right) 1.414$$

(16)a $\text{Voc} = (4\pi \text{ X } 10^{-7}) \cdot N \cdot \omega \text{ Ar Hz } (\mu r)$ (ref: Collins, Phase I, 1975, p. 5)

Voc = open circuit volt at receiving loop
$\omega = 2\pi$ (CTRFRQ)

Where:

$$\text{Ar} = \frac{\text{Cr}^2}{(4\pi)\ (3.2808)^2} = \text{area in (meters)}^2 \text{ of receiving loop}$$

* Cr = circum. receiving loop in feet
$\mu r = 1$

b $$\text{Voc} = \frac{(2\pi \text{ X } 10^{-7})(\text{CTRFRQ})\ \text{Cr}^2\ \text{Hz}}{(3.2808)^2}$$

(17) $$20 \log \text{Voc} = 20 \log \left[\frac{2\pi \text{ X } 10^{-7}}{(3.2808)^2}\right] + 20 \log \text{CTRFRQ} + 40 \log \text{Cr} + 20 \log \text{Hz}$$

(18) Hz = A x I x att(d,f)

Hz = signal magnetic field
att(d,f) = earth attenuation; a function of depth, frequency, and conductivity

(18) continued

where:

$$A = \frac{Ct^2}{4\pi(3.2808)^2} = \text{area } (m^2) \text{ for xmit loop}$$

* Ct = circum. xmit loop in feet

and:

I(I) = E/Z(I)

(19) $20 \log Hz = 20 \log A + 20 \log I + 20 \log att(d,f)$

$= 40 \log Ct - 20 \log [4\pi(3.2808)^2)]$

$+ 20 \log I$

$+ ATTNUA$

[ATTNUA = earth attenuation (See Table XII)]

(20) $20 \log VOCSPH(I) = -144.6756 + 20 \log CTRFRQ(I) + 40 \log Cr$

$+40 \log Ct - 42.6234 + 20 \log XMITSP(I)$

$+ ATTNUA(I)$

VOCSPH(I) = open circuit volt. in band I at receiver for speech signal

(21) $20 \log VOCNOI(I) = -144.6756 + 20 \log CTRFRQ(I) + 40 \log Cr$

$+40 \log Ct - 42.6234 + 20 \log XMITNO(I)$

$+ ATTNUA(I)$

VOCNOI(I) = open circuit volt. in band I at receiver for noise from surface input
process electromagnetic noise in the mine through receiving loop

(22) $20 \log ELMGNO(I) = -144.6756 + 20 \log CTRFRQ(I) + 40 \log Cr$

$+ THATNO(I)$

ELMGNO(I) = electromagnetic noise in band I at receiver
* THATNO(I) = level in dB of electromagnetic noise in the mine in band I.

(23)a VOCNOI(I) = VOCNOI(I) + RERERE(I)

b VOCSPH(I) = VOCSPH(I) + RERERE(I)

c ELMGNO(I) = ELMGNO(I) + RERERE(I)

RERERE(I) = relative receiver response in band I

(24) $$\text{SBTNO1(I)} = 10^{\left[\frac{20 \log \text{ELMGNO(I)}}{10}\right]} + 10^{\left[\frac{20 \log \text{VOCNOI(I)}}{10}\right]}$$

SBTNO1(I) = subtotal 1 of noises in band I in watts

(25) $\text{ELNOPO(I)} = (6.987 \times 10^{-15}) (\text{BNDWTH(I)})$ (See Note 2, p. 26)

ELNOPO(I) = electronic noise in band I in watts

(26) SBTNO2(I) = SBTNO1(I) + ELNOPO(I)

SBTNO2(I) = subtotal 2 of noises in band I in watts

(27) $$\text{POWSPH(I)} = 10^{\left[\frac{20 \log \text{VOCSPH(I)}}{10}\right]}$$

POWSPH(I) = power of speech signal in band I in watts
add 93 dB (receiver gain) to each band

(28)a $\text{POWSPH(I)} = (\text{POWSPH(I)}) (2 \times 10^9)$

b $\text{POWNOI(I)} = (\text{POWNOI(I)}) (2 \times 10^9)$

(29) $$\text{SUMPO2} = \sum_{I=1}^{20} \text{POWSPH(I)} + \sum_{I=1}^{20} \text{POWNOI(I)}$$

SUMPO2 = total power after 93 dB gain

(30) $$\text{CORFC2} = \frac{.01}{\text{SUMPO2}}$$

CORFC2 = correction factor to adjust SUMPO2 to .01 watts
(automatic gain control function)

(31)a POWSPH(I) = POWSPH(I) X CORFC2

b POWNOI(I) = POWNOI(I) X CORFC2

at this point:

$$.01 = \sum_{I=1}^{20} \text{POWSPH(I)} + \sum_{I=1}^{20} \text{POWNOI(I)}$$

(32)a $\text{VOSPH2(I)} = \sqrt{\text{POWSPH(I)} \cdot 8}$

b $\text{VONOI2(I)} = \sqrt{\text{POWNOI(I)} \cdot 8}$

VOSPH2(I) = voltage in each band for speech
VONOI2(I) = voltage in each band for noise

conversion in dB:

(33)a VOSPH2(I) = 20 log VOSPH2(I)

b VONOI2(I) = 20 log VONOI2(I)

now, VOSPH2(I) = voltage in band I for speech in dBV
VONOI2(I) = voltage in band I for noise in dBV

(34)a SPLSPH(I) = VOSPH2(I) + SPKSPC(I)

b SPLNOI(I) = VONOI2(I) + SPKSPC(I)

SPLSPH(I) = sound pressure level for speech in band I @ .5m
SPLNOI(I) = sound pressure level for noise in band I @ .5m
* SPKSPC(I) = speaker response characteristics for 1 v input

correct for bandwidth to get Spectrum Level:

(35)a SPLSPH(I) = SPLSPH(I) - 10 log BNDWTH(I)

b SPLNOI(I) = SPLNOI(I) - 10 log BNDWTH(I)

(36) SPLSPH(I) = SPLSPH(I) + 12

calculate Articulation Index:

(37) If $(\text{SPLSPH(I)} - \text{SPLNOI(I)}) \leq 0$ Diff(I) = 0

(38) If $(SPLSPH(I) - SPLNOI(I)) \leq 30$ Diff(I) = 30

(39) If $0 < (SPLSPH(I) - SPLNOI(I)) < 30$ Diff(I) = Diff(I)

(40) $$A.I. = \frac{\sum_{I=1}^{20} \text{Diff}(I)}{600}$$

AI = articulation index

Note 1: If the limit of 500 watts is to be changed, statements 75, 76, and 77 of the program must be altered. Simply change 500 to the desired power limit.

Note 2: ELNVOC is assumed to be 10 dB below 20 microvolts at the receiver input. For other electronic noises, insert the appropriate figures for ELNVOC and recalculate equation 25. The revised equation must then be substituted for statement No. 140 in the program.

Table XII

Data Card Format

Data Card	Format	Data Variable	
1	F10.2	Ct	circum. of transmit loop (ft)
2	F10.2	RHO	resistivity (ohms/Kft) (Transmit Loop)
3	F10.2	DIM	wire diameter (mils) (Transmit Loop)
4	10F 5.2	CTRFRQ	center freqs. of bands 1-10 (Hz)
5	10F 5.2		center freqs. of bands 11-20
6	10F 5.2	SPHSPM	ideal speech spectrum bands 1-20 ref. .0002 N/m^2
7	10F 5.2		ideal speech spectrum bands 11-20
8	10F 5.2	SIGPRO	signal processing bands 1-10 (dB)
9	10F 5.2		signal processing bands 11-20
10	10F 5.2	BNDWTH	bandwidths bands 1-10 (Hz)
11	10F 5.2		bandwidths bands 11-20
12	10F 5.2	ACSFNO	acoustic surface noise bands 1-10 (dB) ref. .0002 N/m^2
13	10F 5.2		acoustic surface noise bands 11-20
14	F10.2	Cr	circum. of receiving loop (ft)
15	10F 5.2	ATTNUA	earth attenuation bands 1-10 (dB)
16	10F 5.2		earth attenuation bands 11-20
17	F10.2	PEAKE	maximum peak voltage (VOLTS)
18	F10.2	PEAKI	maximum peak current (AMPS)
19	10F 5.2	THATNO	E-M noise in the mine bands 1-10 (dB) ref. 1A/m
20	10F 5.2		E-M noise in the mine bands 11-20
21	10F 5.2	RERERE	relative receiver response bands 1-10 (dB)
22	10F 5.2		relative receiver response bands 11-20
23	10F 5.2	SPKSPC	speaker response spectrum bands 1-10 (dB) ref. .0002 N/m^2 @ .5m*
24	10F 5.2		speaker response spectrum bands 11-20*

*for 1 volt rms speaker input in the band

Note: ATTNUA (I) = the vertical magnetic field in the mine in DB ref 1A/M that would be produced with a vertical axis loop antenna on the surface with loop area = 1 $metre^2$ and loop current = 1 ampere at a frequency = CTRFRQ(I).

THATNO = Vertical Magnetic Field Intensity noise in the mine due to mine generated noise, noise from distant electrical storms attenuated through the earth, etc.

Table XIII

List of Variables

Ct	- circumference of transmit loop	ft	*
RHO	- resistivity	ohms/Kft	*
DIM	- wire diameter	mils	*
CTRFRQ(I)	- center frequency of band I	Hz	*
RA	- loop radius	cms	
D	- wire diameter	cms	
L	- inductance	Henrys	
R	- resistance	ohms	
C	- capacitance	FARADS	
Z(I)	- impedance of band I	ohms	
SPHSPM(I)	- speech spectrum level of band I	dBV	*
SIGPRO(I)	- signal processing of band I	dBV	*
BNDWTH(I)	- bandwidth of band I	Hz	*
ACSFNO(I)	- acoustic surface noise of band I	dBV	*
Cr	- circumference of receiving loop	ft	*
ATTNUA(I)	- earth attenuation of band I	dB	*
PEAKE	- maximum peak signal voltage	VOLT	*
PEAKI	- maximum peak signal current	AMPS	*
RMSE	- maximum RMS voltage	VOLT	
RMSI	- maximum RMS current	AMPS	
ATTSP(I)	- speech attenuation fraction of band I		
ATTNO(I)	- noise attenuation fraction of band I		
XMITSP(I)	- transmit current for speech of band I	AMPS	
XMITNO(I)	- transmit current for noise of band I	AMPS	
TOTSPI	- total speech current rms	AMPS	
TOTNOI	- total noise current rms	AMPS	
TOTALI	- total (speech and noise) current rms	AMPS	
POWRES	- power through resistor	WATTS	
CORFC3	- correction factor to adjust currents preventing POWRES from exceeding 500 W		
CORFC4	- correction factor to adjust currents preventing TOTALI X 1.414 from exceeding PEAKI		
VOCSPH(I)	- open circuit voltage for speech band I at receiving loop	dBV	
VOCNOI(I)	- open circuit voltage for noise band I at receiving loop	dBV	
THATNO(I)	- electromagnetic noise in the mine in band I	dB re 1A/m	*
ELMGNO(I)	- electromagnetic noise of band I	dBV	
SBTNO1(I)	- sum of ELMGNO and VOCNOI of band I	WATTS	
ELNOPO(I)	- electronic noise power of band I	WATTS	
SBTNO2(I)	- sum of SBTNO1 and ELNOPO of band I	WATTS	
POWSPH(I)	- power of speech signal of band I	WATTS	
POWNOI(I)	- power of noise of band I	WATTS	
TOPS2	- total power of speech bands 1-20 after 93 dB gain	WATTS	
TOPN2	- total power of noise bands 1-20 after 93 dB gain	WATTS	
SUMPO2	- sum of TOPS2 and TOPN2	WATTS	

Table XIII (continued)

CORFC2	- correction factor to adjust power to .01	WATTS
VOSPH2(I)	- voltage for speech signal of band I	dBV
VONOI2(I)	- voltage for noise of band I	dBV
SPKSPC(I)	- speaker response of band I	dB
SPLSPH(I)	- sound pressure level for speech of band I	dB re .0002 N/m^2 @.5m
SPLNOI(I)	- sound pressure level for noise of band I	dB re .0002 N/m^2 @.5m
SPLSNZ(I)	- SPL speech normalized to band 3 of noise	dB
SPLNNZ(I)	- SPL noise normalized to band 3 of noise	dB

* signifies card input data

8. List of References

a) ANSI S3.5-1969 "Methods for the Calculation of the Articulation Index," (Approved January 16, 1969), American National Standards Institute, Inc.

b) Collins Radio Group, "Phase I Study of Various Approaches, Emergency Downlink Receiver," Rockwell, International, Cedar Rapids, Iowa (14 January 1975).

c) Fletcher, H., "Speech and Hearing in Communication," New York: Van Nostrand (1929 rev. 1953).

d) Michael, P.L., L. W. Saperstein, J. H. Prout, R. L. Kerlin, W. W. Kaufman, G. R. Bienvenue, "Aspects of Noise Generation and Hearing Protection in Underground Coal Mines," Final Report Grant G 0122004 (November 20, 1972) USBM OFR 17-73.

e) Michael, P.L., J. H. Prout, R. L. Kerlin, G. R. Bienvenue, S. Singer, G. Kreik, A. Kohut, "Evaluation of Speech Processing Systems; Evaluation of Electronic/Active Hearing Protectors for use in Underground Coal Mines," Final Report, Grant G 0155032 (May, 1976).

f) Villchur, Edgar, "Simulation of the Effects of Recruitment on Loudness Relationships in Speech," J. Acoust. Soc. Am., 56, 5, November 1974.

g) Wait, J.R., and K. P. Spies, "Sub-Surface Electromagnetic Fields of a Circular Loop of Current Located Above Ground," Institute for Telecommunication Sciences, Office of Telecommunications, U.S. Department of Commerce, Boulder, Colorado; Informal Report to the USBM on Task 5, March 15, 1972.

1. Introduction

The U.S. Bureau of Mines is presently developing a baseband communication system for transmitting verbal information from the surface to miners who may become trapped underground.

All of the many communication systems investigated for this purpose involve a limitation on bandwidth, a reduction in naturalness of the transmitted speech, and/or some distortion of the speech signal waveform. Any or all of these factors affect the intelligibility of the speech transmitted through the communication systems. An important part of the evaluation of such a communication system is the measurement of the intelligibility of the speech received through the system.

Clinical methods of measuring speech intelligibility are well established for various purposes. Of these several established tests the modified rhyme test (MRT), which was conceived for testing speech intelligibility in auditoriums and classrooms, was chosen for testing underground coal mine communication systems. This test measures intelligibility in a limited vocabulary situation as would be encountered in coal mining communications.

The rationale behind the choice of this particular test is given by Michael, et al., (1976) and is reproduced as Appendix B of this report for convenience.

An indirect estimate of speech intelligibility may be made by determining the articulation index (AI) from the characteristics of the acoustic output of the communication system. The AI is calculated from the signal-to-noise ratio in each of 20 equal-articulation bands of frequency. The resulting AI can be used to predict approximately the score that should be expected from the use of the modified rhyme test. Thus, the MRT and the AI each give an independent estimate of the intelligibility of speech transmitted through the communication system.

2. Evaluation of Communication Systems by MRT Scores

2.1 Modified Rhyme Tape Format

The details of the Modified Rhyme Test are explained in the Final Report of Grant G 0155032 and in Appendix B of this report. In its original form, the word lists are supplied on six reels of magnetic tape. For the purpose of evaluation of through-the-earth communication systems for USBM, only the male voice recorded at +30 dB s/n was used. These word lists (Table I) were copied with new calibration signals on two reels of low-print tape and have been supplied to the US Bureau of Mines, Pittsburgh Mining and Safety Research Center.

These tapes start with an announcement identifying the particular word list followed by two calibration signals. The first calibration signal is 20 seconds of a 1.000 kHz pure tone recorded at 0 VU level on the tape. This tone is immediately followed by 20 seconds of speech-band noise also recorded at 0 VU ±.5VU. The speech-band noise was generated by passing pink noise through a Krohn-Hite 3750 band pass filter having 24 dB/octave slopes with -3 dB points at 355 Hz and 2828 Hz. The resulting filtered pink noise contains the three principal octaves of the speech band as defined by Fletcher (1953). Following the calibration tones and preceding the word list is a short instructional announcement copied directly from the original word list tapes. (Practice word lists were omitted from these tape copies.)

Each word list contains 50 items. The words are presented in a standard carrier phrase which includes words before and after the test word. The word lists are recorded so that the average level of the test word itself is 0 VU ±2 VU. The signal-to-noise ratio was set by the recording on the original tapes at +30 dB.

The tapes supplied to USBM for use in these evaluations were recorded on a two channel magnetic tape recorder with half-track-width heads. The two tracks each contain the same signal recorded at the same level so that the tapes can be reproduced on either a half-track or a full-track reproducer.

2.2 Reproduction of Word List Tapes

When reproducing the taped word-lists to test a communication system, the 1.000 ± .001 kHz calibration tone is provided to set signal level and to check overall speed accuracy. The noise calibration provides a broad band signal that can be used to determine the bandwidth of the system relative to the speech band by comparing the level of the noise calibration signal with the level of the pure tone calibration signal after they have been transmitted through the system. Allowance must be made in the electronic systems for occasional signal peaks in the word lists which may rise to levels of +4 to +5 VU to avoid problems of waveform distortion.

2.3 Procedures

Tests of through-the-earth communication systems were planned that involved a series of tests at actual coal mine sites by USBM personnel with technical assitance from EAL. Word list tapes supplied by EAL were to be transmitted over the communication system and received underground. The audio signals from the underground receiver were to be recorded in the mine on magnetic tape and then returned to EAL for evaluation. However, in lieu of actual field tests, the program was changed to develop a computer model and to test some PMSRC simulations of the through-the-earth voice communication system.

Tape recordings of the audio output of these simulated communication systems were scored at EAL for intelligibility by a group of 10 trained listeners, all having normal hearing sensitivity and ranging in age from 22-28 years. Listeners were seated in a standard audiometric test room which has a reverberation decay time comparable to that of a coal mine entry. Each subject was seated in approximately the center of the test room one meter from the speaker through which the MRT test words were reproduced. The speaker was placed at 45° azimuth in relation to the subject's head. The speaker and amplifier used are described in section 3 of this report.

Only two of the word lists from any given test recording were scored. To make a statistical comparison of the subjects' performance on the two test lists, the order of the test list presentation was altered so that half the subjects heard one list first and the other half heard the other list first.

The initial calibration of the reproduced sound level was accomplished by using a General Radio Type 1565-A sound level meter with C frequency weighting and slow meter response to set the sound level of the test words to 65 dB at the position of listener's ear with the listener absent. The voltage of the calibration tone at the loudspeaker voice-coil was measured on a Hewlett-Packard Model 403B Voltmeter. This voltage was recorded on the score data sheet. Thereafter, the voltage of the calibration tone was used to set the acoustic signal level for each test presentation.

2.4 Results

Test scores for the communication systems tested at PMSRC are given in Tables II, III, IV, V, VI, and VII. The type of signal processing used with tapes RCB-4, 5, 6, 7, 8, and 9 was not revealed to EAL so no evaluation was attempted. Table VIII shows the results of a simulation performed at EAL. The theoretical input filter response curve corresponding to SP-9 is shown in Figure 1. Figure 2 shows the acoustic signal spectrum plotted relative to the normalized noise spectrum at the loudspeaker output for the communication system with signal processing filter SP-9 used. The articulation index for this system was calculated to be .15 but because the curves showed a positive signal-to-noise ratio over a significant portion of the pass band it was decided to obtain MRT scores by re-constructing these spectra as closely as possible. The MRT test score of 68% word intelligibility (AI = .26) is higher than the value of 35% word intelligibility (AI = .15) predicted by the computer program probably because of slight errors in duplicating the required signal and noise spectra.*

The last two tapes analyzed, RCB-10 and 11, (Tables IX and X) were prepared by USBM to determine the effect of compression on the MRT scores. Although the standard deviations of the scores for RCB-11 are slightly greater in general than those for RCB-10, there appears to be no significant difference between compression and no compression as far as word scores are concerned. The major improvement that should be expected from the use of compression is a slight improvement in signal-to-noise ratio that results from a higher average signal energy in the transmit loop while staying within the peak current limit.

3. Low-Power Amplifier

The through-the-earth communication system involves a man-carried amplifier and loudspeaker which is powered by the miner's cap-lamp battery and is worn on his safety belt.

In anticipation of the use of the man-carried loudspeakers to reproduce the test words in our audiometric test suite, a 100 milliwatt (maximum) power amplifier (Figure 3) was built to reduce the possibility of accidental burnout of these small speakers. This amplifier is adequate to produce

*See page 3a.

Only two of the word lists from any given test recording were scored. To make a statistical comparison of the subjects' performance on the two test lists, the order of the test list presentation was altered so that half the subjects heard one list first and the other half heard the other list first.

The initial calibration of the reproduced sound level was accomplished by using a General Radio Type 1565-A sound level meter with C frequency weighting and slow meter response to set the sound level of the test words to 65 dB at the position of listener's ear with the listener absent. The voltage of the calibration tone at the loudspeaker voice-coil was measured on a Hewlett-Packard Model 403B Voltmeter. This voltage was recorded on the score data sheet. Thereafter, the voltage of the calibration tone was used to set the acoustic signal level for each test presentation.

2.4 Results

Test scores for the communication systems tested at PMSRC are given in Tables II, III, IV, V, VI, and VII. The type of signal processing used with tapes RCB-4, 5, 6, 7, 8, and 9 was not revealed to EAL so no evaluation was attempted. Table VIII shows the results of a simulation performed at EAL. The theoretical input filter response curve corresponding to SP-9 is shown in Figure 1. Figure 2 shows the acoustic signal spectrum plotted relative to the normalized noise spectrum at the loudspeaker output for the communication system with signal processing filter SP-9 used. The articulation index for this system was calculated to be .15 but because the curves showed a positive signal-to-noise ratio over a significant portion of the pass band it was decided to obtain MRT scores by re-constructing these spectra as closely as possible. The MRT test score of 68% word intelligibility (AI = .26) is higher than the value of 35% word intelligibility (AI = .15) predicted by the computer program probably because of slight errors in duplicating the required signal and noise spectra.*

The last two tapes analyzed, RCB-10 and 11, (Tables IX and X) were prepared by USBM to determine the effect of compression on the MRT scores. Although the standard deviations of the scores for RCB-11 are slightly greater in general than those for RCB-10, there appears to be no significant difference between compression and no compression as far as word scores are concerned. The major improvement that should be expected from the use of compression is a slight improvement in signal-to-noise ratio that results from a higher average signal energy in the transmit loop while staying within the peak current limit.

3. Low-Power Amplifier

The through-the-earth communication system involves a man-carried amplifier and loudspeaker which is powered by the miner's cap-lamp battery and is worn on his safety belt.

In anticipation of the use of the man-carried loudspeakers to reproduce the test words in our audiometric test suite, a 100 milliwatt (maximum) power amplifier (Figure 3) was built to reduce the possibility of accidental burnout of these small speakers. This amplifier is adequate to produce

*See page 3a.

Note: The AI quoted for SP-9 was obtained with the original computer program in which transmit power was calculated with the formula $P = \frac{E^2}{z}$. This example was included primarily to illustrate the differences that can arise between test word scores predicted from the AI and those actually measured.

Parameter for SP-9 are as follows:

Radius of XMIT Loop	2500 cm
Wire Diameter	0.2053 cm
Inductance	0.000328 henrys
Resistance	0.8184 ohms
Capacitance	10^6 farads

65 dB sound pressure level (conversational level) at one meter distance from the loudspeaker. The man-carried speakers will be used with tapes recorded in actual coal mine situations to provide an overall test of USBM experimental communication systems. Preliminary simulation tapes produced in the laboratory at PMSRC are being reproduced through this amplifier and a Philips 5060M8 wide-range speaker mounted in an infinite baffle.

4. Computer Simulation

4.1 Calculations to be performed

An Articulation Index Computer program was designed to simulate the proposed Bureau of Mines communication system. Based on the 20 band method for the calculation of Articulation Index (AI), this program proceeds through all calculations of speech and noise levels in the 20 separate bands whose mid frequencies are given in Table 8 of ANSI 3.5-1969. All variables listed with a subscript (I) are thus indexed for bands 1 through 20.

The program processes a speech spectrum (Figure 4) and Audio noise spectrum of 20 bands each through a signal processing filter, an Audio Power Amplifier, the "earth attenuation" and the receiver system (Fig. 5). Currents for speech and noise are determined as they appear at the transmit loop. From these currents, the total power for transmission is calculated. At this point, the program checks that the power is less than or equal to 500 watts and that the total peak current is equal to or less than the maximum allowable peak current. Any necessary adjustments are performed. The open circuit voltages induced in the receiving loop are then calculated for speech and noise in each of the 20 bands and attenuated by the earth filter response (Figure 6). (For other power limits, see Note 1, p. 26.)

After electromagnetic noise in the mine (Fig. 7) induced in the receiving loop is added into the noise array, the signals are then combined with the electronic noise of the receiver and amplified. Here, also, the speech and noise arrays are adjusted for the band pass of the receiver (Figure 8). The total power appearing at the loudspeaker is then adjusted to .01 watts simulating the action of the automatic gain control. The 20 band arrays of speech and noise are converted through the speaker response spectrum (Figure 9) to sound pressure levels ref. .0002 n/m^2 at .5 metre. The AI is then calculated as specified in the ANSI standard.

4.2 Description of Program

The initial input data, for the calculation of electrical parameters, i.e., the circumference of the transmit loop (CT), wire resistivity (RHO) and wire diameter (DIM) are entered. The radius of the transmit loop (RA), wire diameter in cms (D), inductance (L) and wire resistance (R) are immediately calculated by the computer* (eqs 1-4, Table XI). If the capacitance (C) is to be taken in account, it is entered in Farads directly into the program after the comment card 'calculate C'. When no resonating capacitor is used, the value $C=10^6$ is entered. This value is

* See Table VII Equations used in Computer Program

presently in the program. RA, D, L, R, and C are printed output. The impedance (Z(I)) is calculated using equation (5) and printed alongside the center frequencies for each of the bands.

For the spectral shape of the speech signal, data from the idealized speech spectrum for male voices (SPHSPM(I)) is taken from Table 8, column (a) of ANSI S3.5-1969. Signal processing (SIGPRO(I)), bandwidths (BNDWTH(I)), acoustic surface noise (ACSFNO(I)), circumference of the receiving loop (CR), earth filter attenuation (ATTNUA(I)), maximum peak signal voltage (PEAKE), and maximum peak signal current (PEAKI) are read into the computer. The Peak and rms values are related to equations 6a and 6b. The earth attenuation is based on a value of 300 metres depth and 0.1 MHOs/metre conductivity but other cases can be used by changing the appropriate data cards. The maximum peak signal current of 19.1 amps and maximum peak signal voltage of 76.4 volts used in the program are based on the Phase Linear amplification specifications. These are also easily changed to allow introduction of loop matching transformers by inserting new data cards.

The ACSFNO presently used in this program is chosen 20 dB below the level of the idealized speech spectrum in each of the 20 bands. This choice is based on the following assumptions:

1. Airborne acoustic surface noise will probably be composed mainly of speech from other persons nearby and, hence, will have the same spectrum characteristic.

2. Data supplied by makers of noise—cancelling microphones (Telex, RCA, Roanwell) indicate that about 20 dB signal to ambient noise can be realized when using these microphones.

3. Airborne acoustic ambient noise of this type will dominate all other system noises in the surface transmitting system.

To determine the signals to be transmitted, SPHSPM(I) and ACSFNO(I) are processed through addition of the signal processing (e.q. 7a, b). The two resulting 20 band arrays are normalized to the maximum value in the resulting speech spectrum, and are then converted to decimal attenuations. (e.q. 8a, b). The transmit currents for speech (XMITSP(I)) and noise (XMITNO(I)) are calculated using equations (9a) and (9b) respectively. After determining the total current from both 20 band arrays (TOTALI), the power through the resistor (POWRES) is calculated from equation (10).

Next, the program goes through a number of check points. If POWRES is greater than 500 watts, the currents are adjusted to bring the power down to 500 watts (e.q. 11, 12a, b). If POWRES is below 500 watts, the program continues to the next check point. Here, the program checks to insure that the total peak current (TOTALI * 1.414) does not exceed PEAKI. When the total current is excessive, the currents for speech and noise are adjusted down (e.q. 13, 14a, b) so that the total peak current equals PEAKI (e.q. 15). POWRES is then recalculated. If the total peak current is within the allowable limit, the program continues.

The open circuit voltage induced in the receiving loop antenna is expressed by equation 16 taken from the Collins Phase I (1975) report.

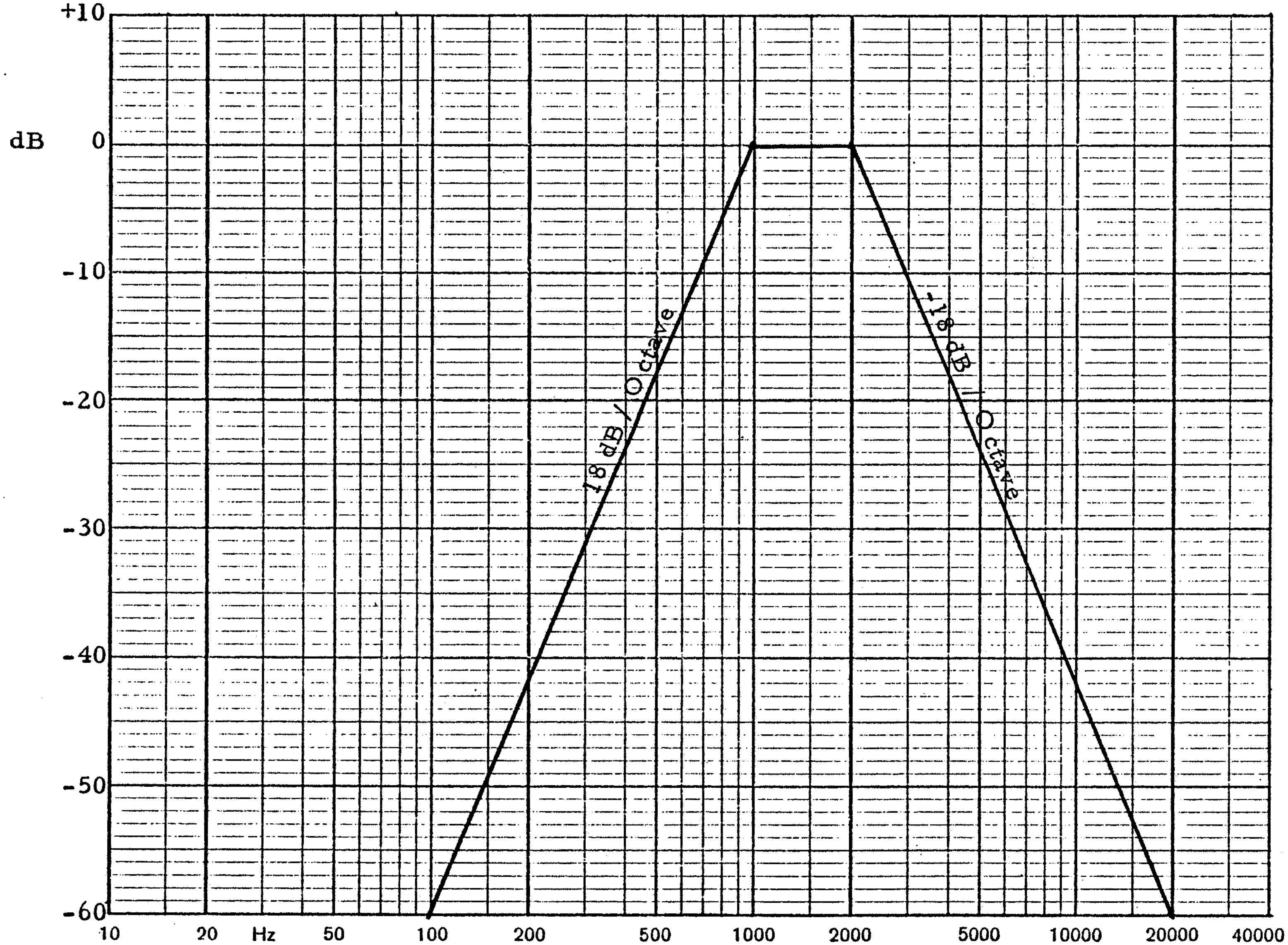

Fig. 1 SP 9 Filter response.

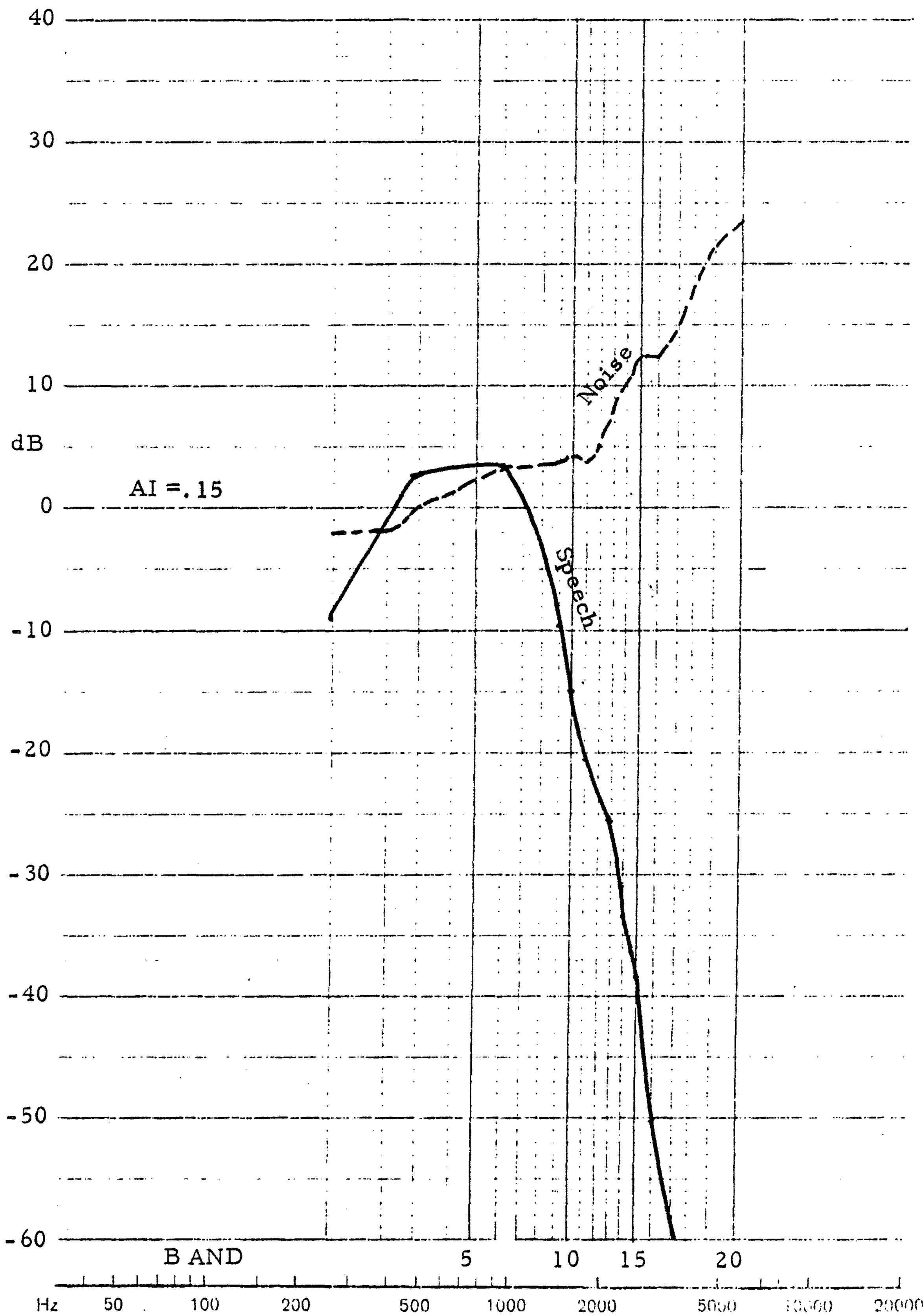

Fig. 2 Speech and noise spectra resulting from the use of SP 9 filter.

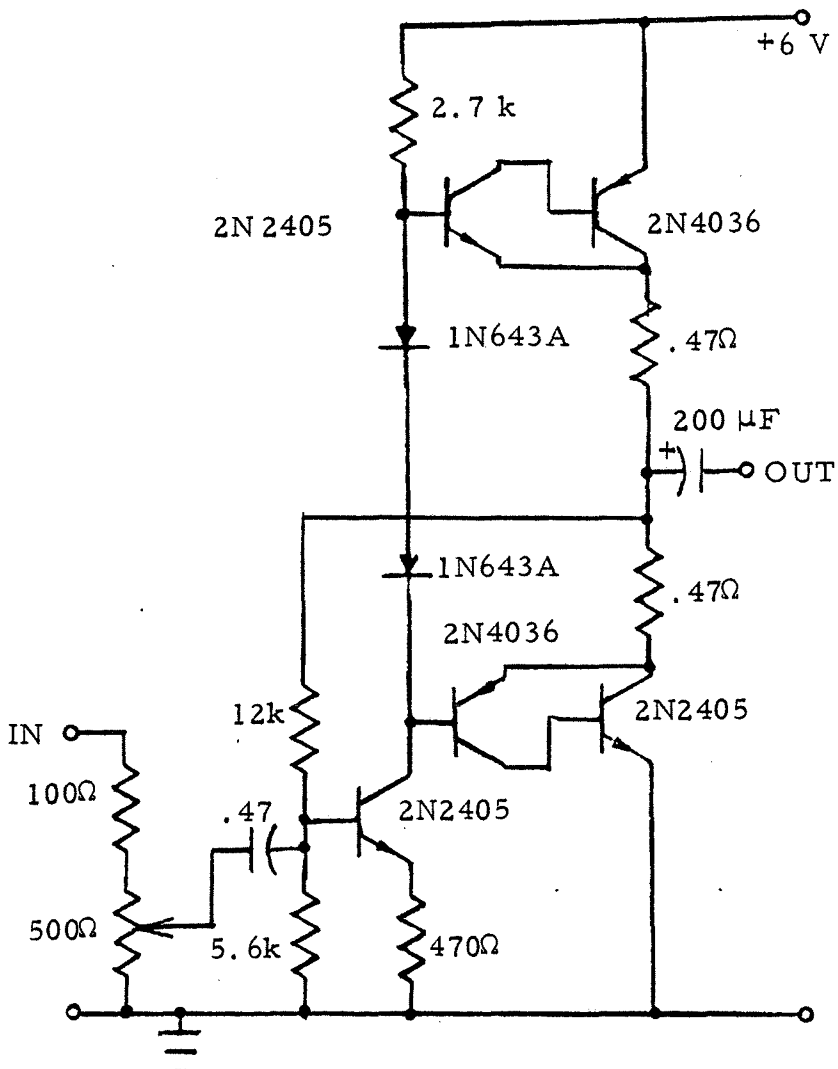

Fig. 3 Schematic diagram of 100 mW audio amplifier.

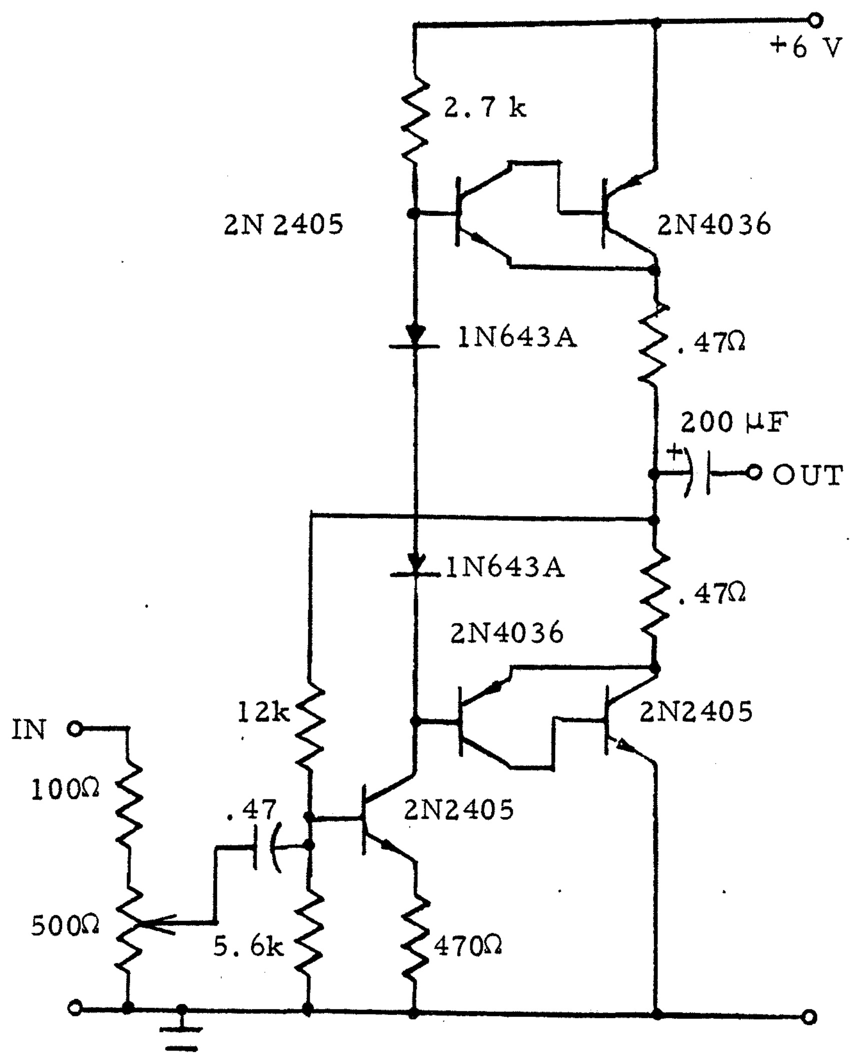

Fig. 3 Schematic diagram of 100 mW audio amplifier.

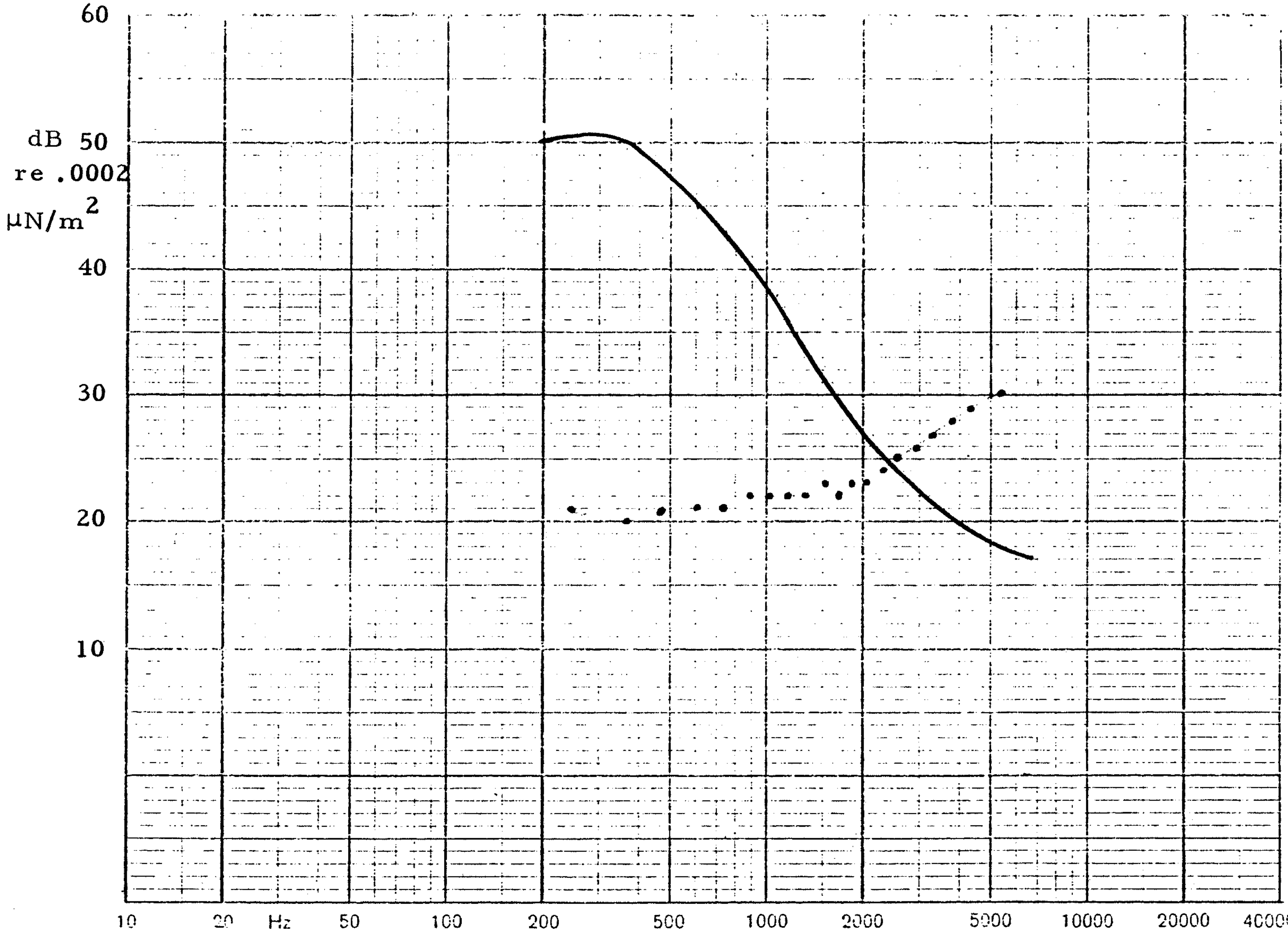

Fig. 4 Idealized speech spectrum level and bandwidth correction

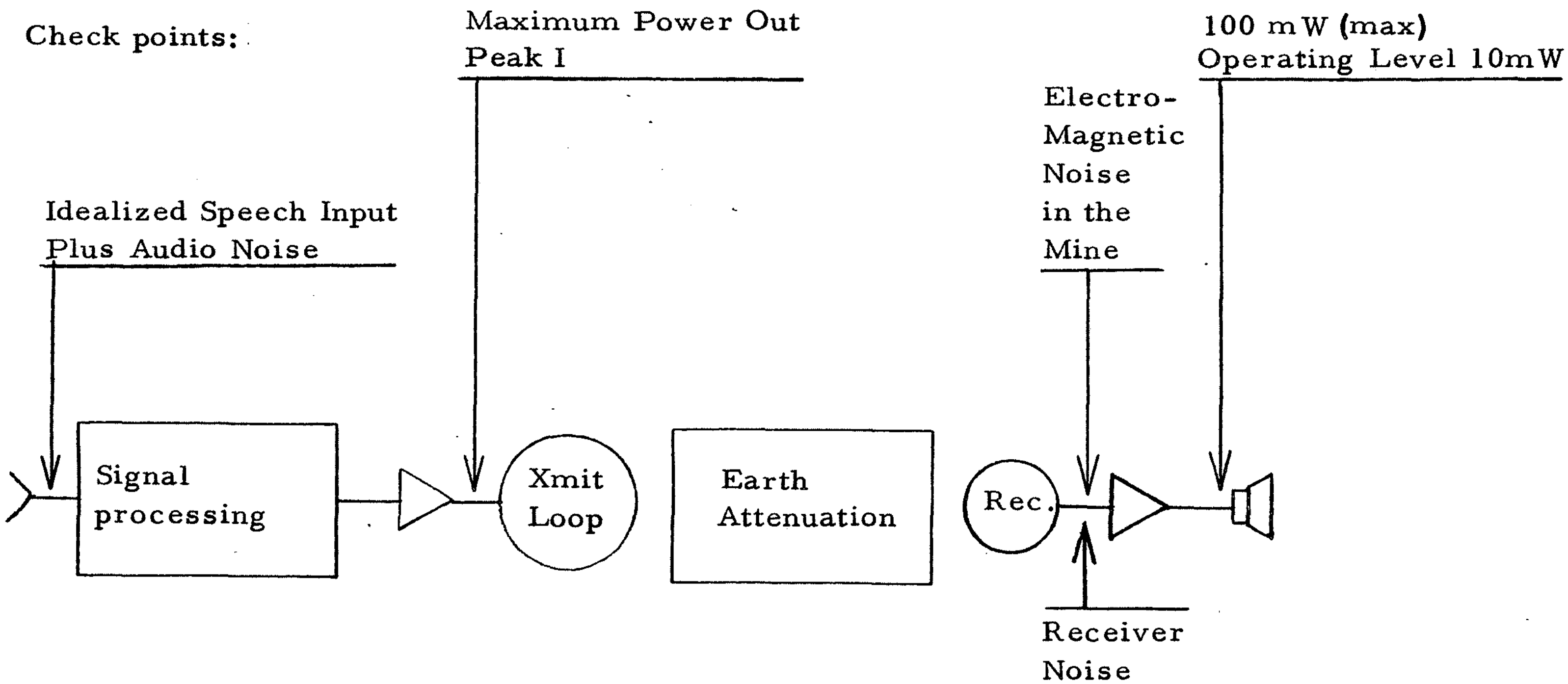

Fig. 5 Communication system block diagram.

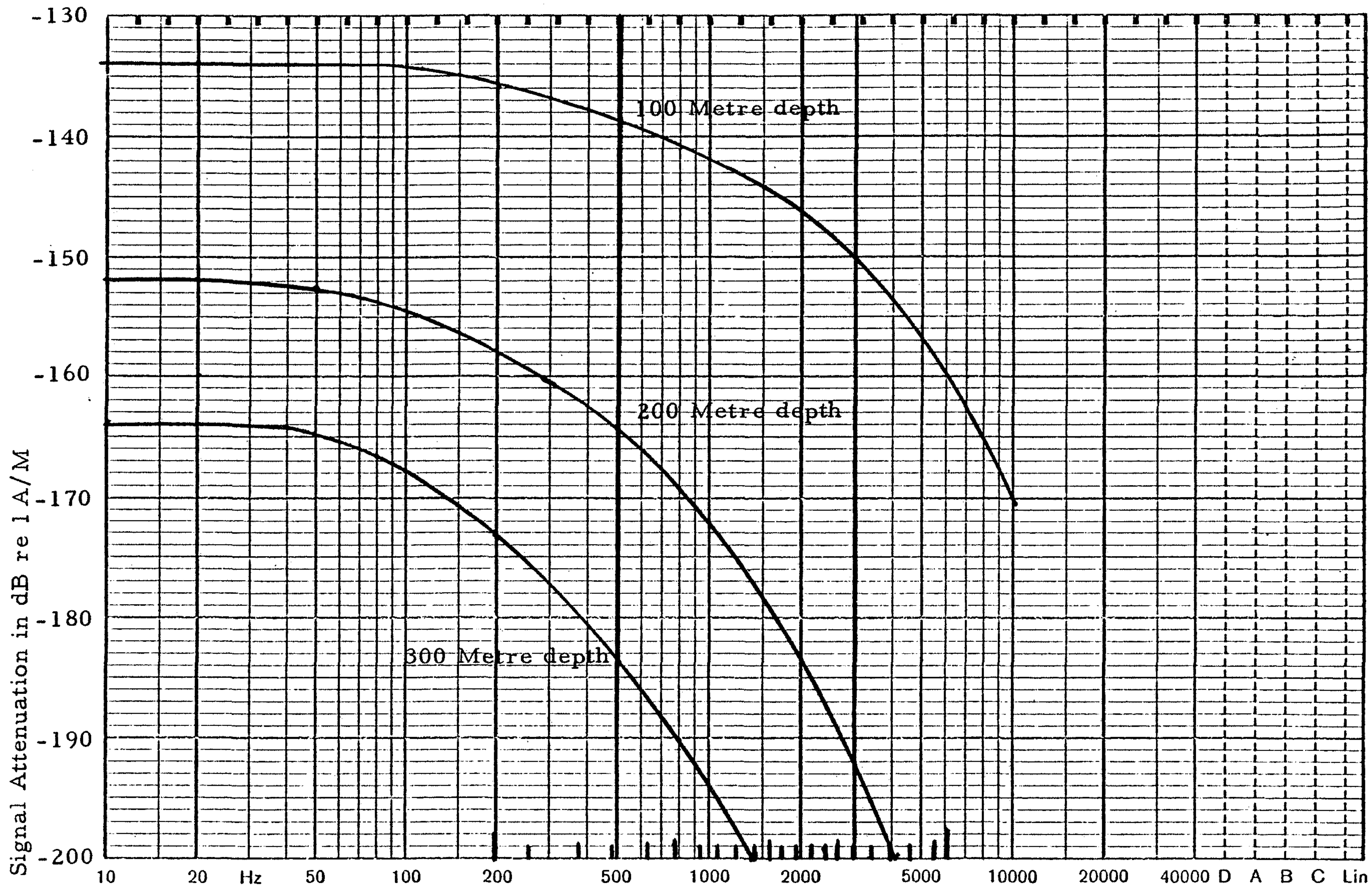

FIG. 6 VERTICAL SIGNAL COMPONENT ATTENUATION BY EARTH FOR COAXIAL LOOPS

CONDUCTIVITY = .100 MHOS /METRE MAGNETIC MOMENT = 1 AMP-METRE^2

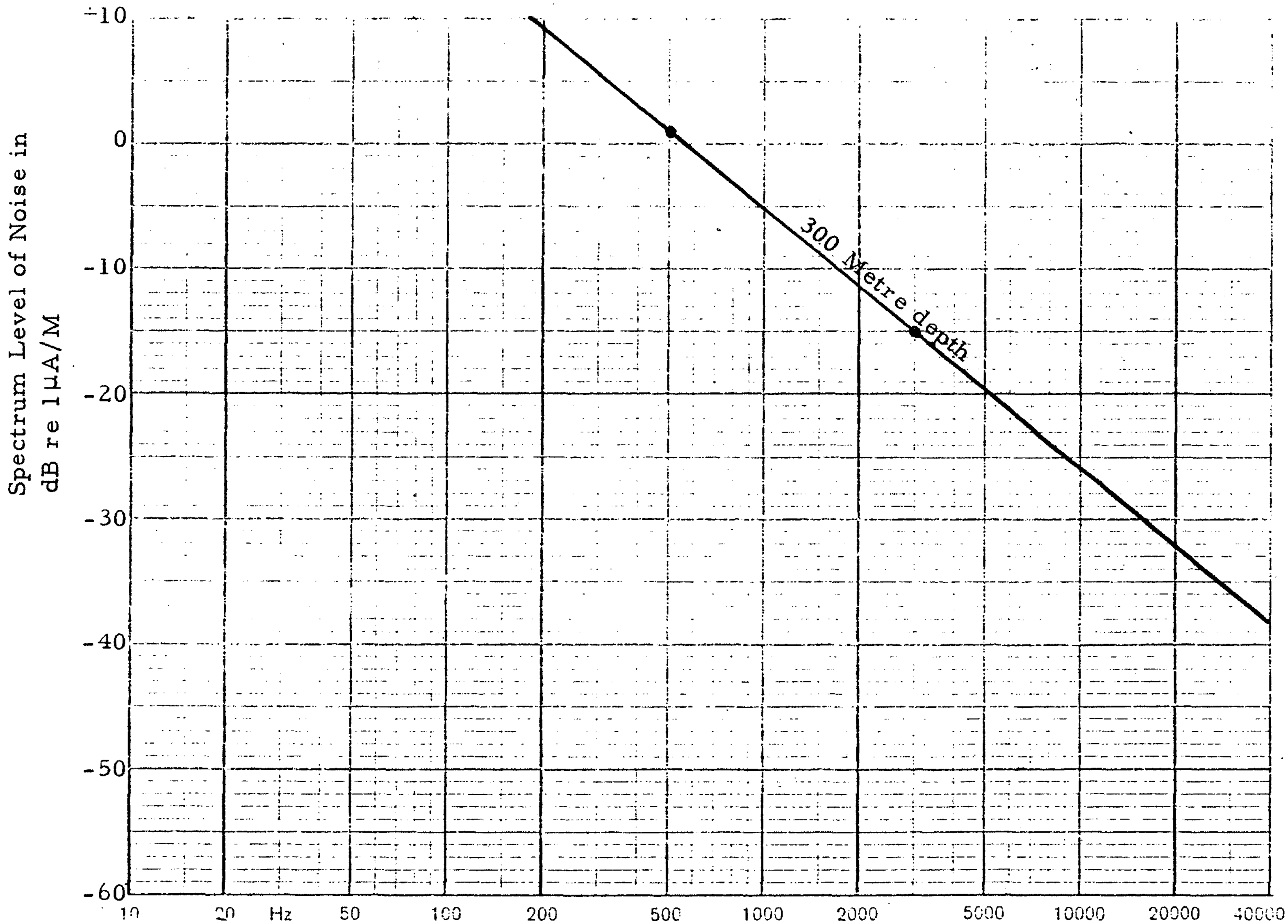

Fig 7 Electromagnetic noise in the mine at the receiver loop
From Collins Radio report (ref. b)

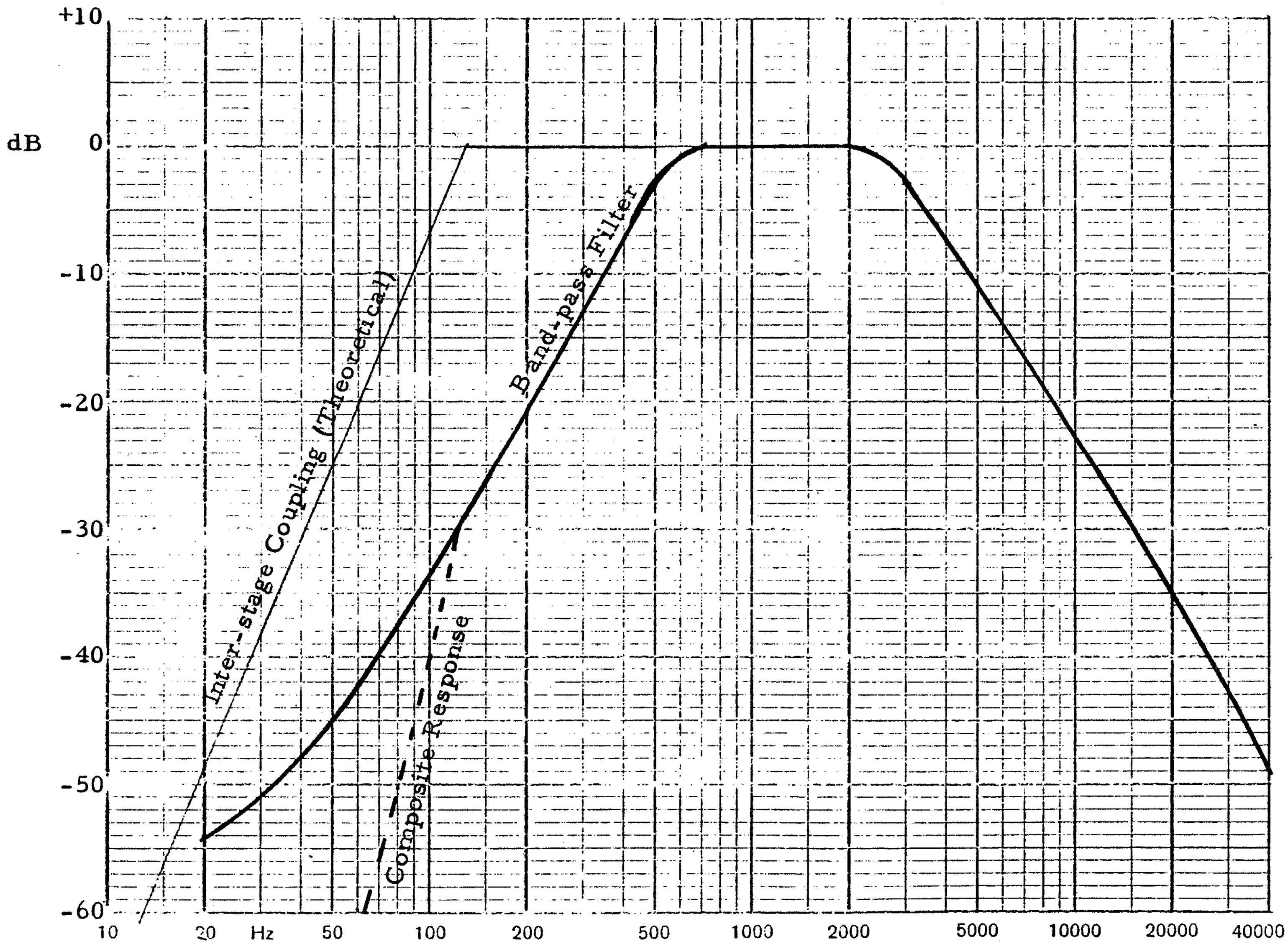

Fig 8 Relative Response of Down-Link Receiver Electronics

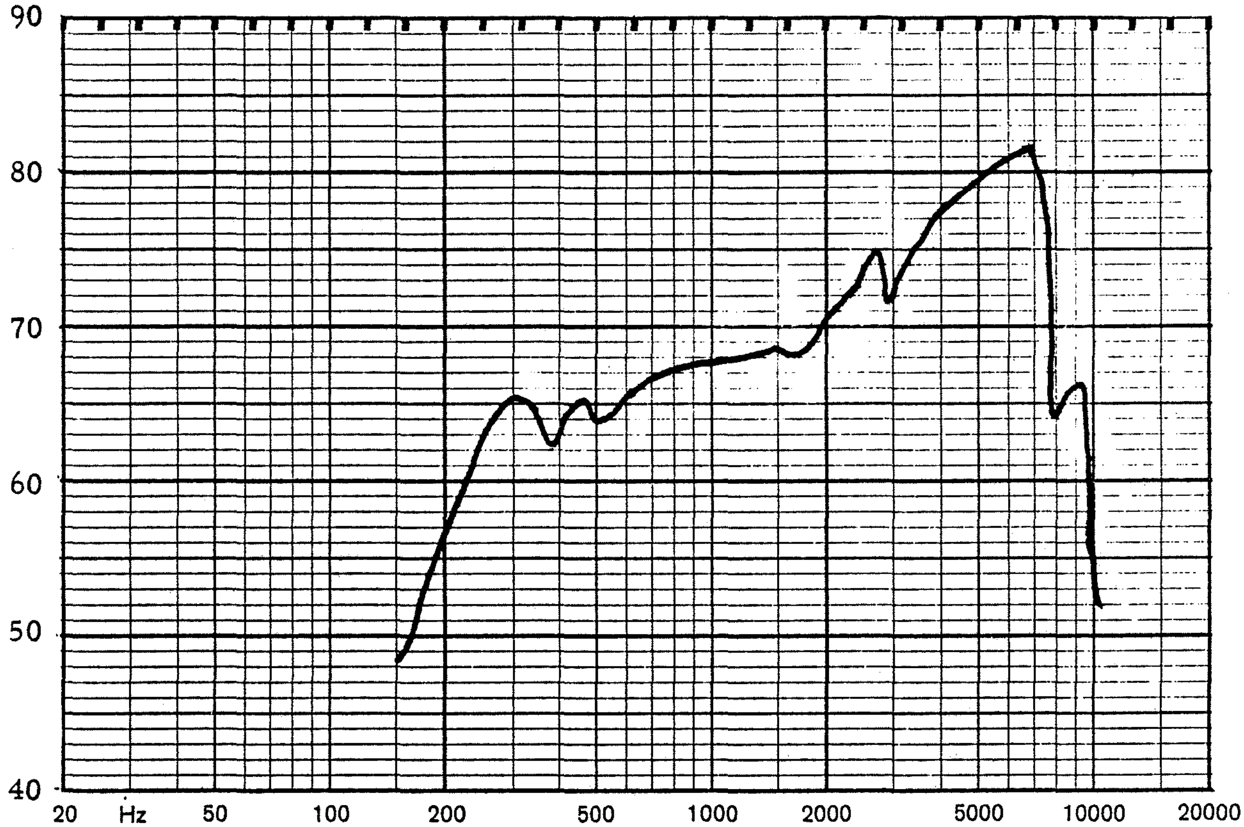

Fig. 9 Response of loudspeaker in dB re .0002 dynes/cm^2 for 0.5 Volts input at 0.5 metres on axis.

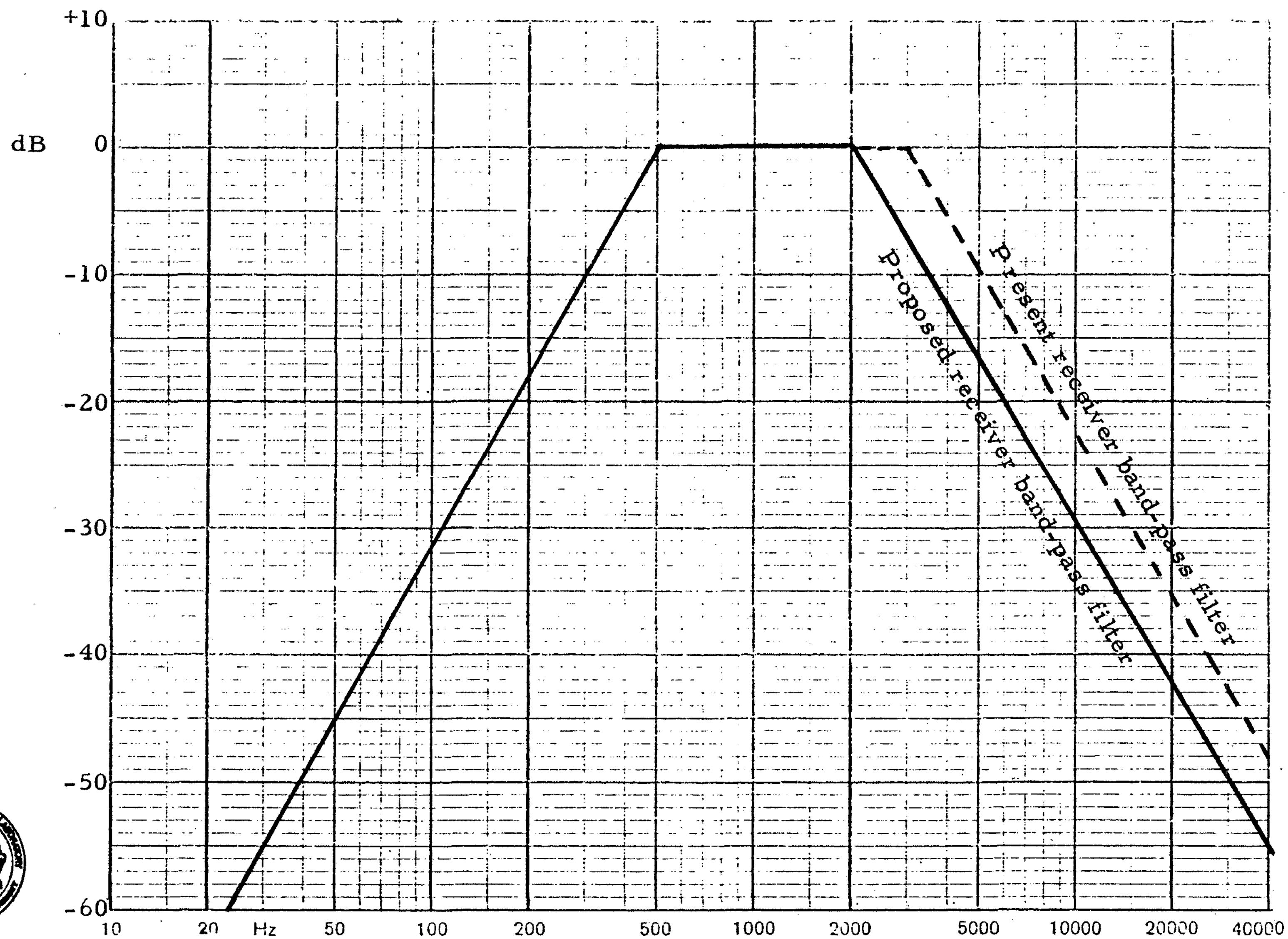

Fig.10 Proposed change in receiver bandpass to reduce noise.

Appendix A

Examples of Computer Simulation Print-out

The complete computer simulation program is reproduced on pages A-3 through A-8 and has been described in section 4 of this report. To illustrate the use of the program basic signal processing parameters were inserted as follows:

> PEAKE = 76.4 Volts and PEAKI = 19.1 Amperes corresponding to the maximum output of the Phase Linear audio power amplifier at clipping level;
>
> SP-7 represents an input filter having a pass band of 500 Hz to 3000 Hz corresponding to the pass band of the receiver system. Band edge skirts of 12 dB/octave were arbitrarily chosen as a practical consideration. The idealized shape of this filter is shown in Figure A-1.

(Note: The numbering SP-7 corresponds to an arbitrary numbering system assigned to signal processing parameters used at EAL to check out the computer program.)

The print-out for SP-7 on pages A-9 through A-13 is best understood by referring to the description of the program in section 4.2 of this report. The Normalized Speech and Noise data are useful in comparing the output signals with the input signal processing. The data in these columns has been normalized to band 3 of the noise spectrum since the noise spectrum at 300 Metre depth is generally stable and little affected by input signal processing. Plotting this data easily shows changes in the speech spectrum and signal-to-noise ratio brought about by changes in signal processing. The signal and noise plots for SP-7 are shown in Figure A-2. The articulation index (AI) of .07 indicate that SP-7 is unsatisfactory for communication to depths as great as 300 Metres.

The power calculations for the transmitting condition using SP-7 indicate that the capabilities of the power amplifier are not fully used. More efficient power transmission can be realized by inserting a matching transformer with turns ratio of 2.6:1 between the power amplifier and the transmitting loop antenna. The signal processing parameters, identified as SP-25 are as follows:

> PEAKE = 29.4 Volts and PEAKI = 49.6 Amperes;
>
> Input filter shape same as for SP-7.

The data for SP-25 are shown on pages A-14 through A-18. The articulation index for these conditions becomes .12 and is a significant improvement over that achieved with SP-7 although still not high enough for satisfactory communication. (See section 5 of this report.) The change in signal-to-noise ratio is easily seen in the signal and noise plot of Figure A-3.

Further increases in AI can be realized by narrowing the signal processing filter band to concentrate the transmitted power over a narrower frequency range that can be passed through the system. Experience with the computer program at EAL has indicated that signal energy at frequencies above 2000 Hz is severely attenuated by the earth filter so that this portion of the pass band can be eliminated from a practical viewpoint. It does not appear advisable to reduce this upper frequency limit further because of the accompanying degradation in intelligibility. However, some increase in AI can be realized by balancing trade-offs between signal processing band pass and matching transformer turns ratio. It does not seem possible to find simple signal processing that will raise the AI to the recommended value of .40 to provide satisfactory communication to a depth as great as 300 metres.

```
C     THE PROGRAM BEGINS HERE
      REAL L
C     SET DIMENSIONS FOR ALL ARRAYS TO BE USED
      DIMENSION SPHSPM(20),SIGPRO(20),BNDWTH(20),CTRFRQ(20),Z(20),ACSFNO
     1(20),ATTNUA(20),VOCSPH(20),VOCNOI(20),THATNO(20),ELMGNO(20),SBTNO1
     2(20),SPKSPC(20),ELECNO(20),SBTNO2(20),DIFF(20),ELNOPO(20),POWSPH(2
     30),POWNOI(20),VOSPH2(20),VONOI2(20),SPLSPH(20),SPLNOI(20),SPLSNZ(2
     40),SPLNNZ(20),ATTSP(20),ATTNO(20),XMITSP(20),XMITNO(20),RERERE(20)
C     READ IN DATA FOR CALCULATION OF IMPEDANCE Z(I): CIRCUM. OF XMIT LO
C    1OP(CT IN FEET),RESISTIVITY(RHO IN OHMS/1000 FT.),WIRE DIAMETER (DI
C    2M IN MILS), CENTER FREQUENCIES (CTRFRQ(I) IN HZ)
      READ(5,1)CT,RHO,DIM
    1 FORMAT(F10.2)
      READ(5,3) (CTRFRQ(I),I=1,20)
    3 FORMAT(10F5.2/10F5.2)
C     CALCULATE RA(LOOP RADIUS CMS)
      RA=100.*CT/20.6139
C     CALCULATE D(WIRE DIAMETER CMS)
      D=DIM*0.00254
C     CALCULATE L(INDUCTANCE HENRYS)
      L=.01257*RA*(ALOG(16.*RA/D)-1.75)*0.000001
C     CALCULATE R(RESISTANCE OHMS)
      R=CT*RHO/1000.
C     CALCULATE C(CAPACITY FARADS)
      C=1000000.
      WRITE(6,5)RA,D,L,R,C
    5 FORMAT('0',4X,'RADIUS OF XMIT LOOP',F10.2,' CM'/5X,'WIRE DIAMETER'
     1,F8.4,' CM'/5X,'INDUCTANCE',F10.6,' HENRYS'/5X,'RESISTANCE',F8.4,'
     2 OHMS'/5X,'CAPACITANCE  ',F12.4,' FARADS')
C     CALCULATE Z(I)
      DO 7I=1,20
      Z(I)=SQRT(R**2.+(6.284*CTRFRQ(I)*L-1./(6.284*CTRFRQ(I)*C))*(6.284*
     1 CTRFRQ(I)*L-1./(6.284*CTRFRQ(I)*C)))
    7 CONTINUE
      WRITE(6,9)
    9 FORMAT('0',5X,'IMPEDANCE--Z(I)  OHMS',10X,'CENTER FREQUENCY--CTRFR
     1Q(I)  HZ.')
      WRITE(6,11) (Z(I),CTRFRQ(I),I=1,20)
   11 FORMAT(10X,F10.6,25X,F10.2)
C     READ IN ADDITIONAL DATA FOR CALCULATION OF CURRENT OF SPEECH AND S
C    1URFACE NOISE IN EACH BAND AT XMIT LOOP
      READ(5,13) (SPHSPM(I),I=1,20)
   13 FORMAT(10F5.2/10F5.2)
      READ(5,15) (SIGPRO(I),I=1,20)
   15 FORMAT(10F5.2/10F5.2)
      READ(5,17) (BNDWTH(I),I=1,20)
   17 FORMAT(10F5.2/10F5.2)
      READ(5,19) (ACSFNO(I),I=1,20)
   19 FORMAT(10F5.2/10F5.2)
C     READ IN DATA FOR CR (CIRCUM. OF RECEIVING LOOP IN FEET), AND ATTNU
C    1A (ATTENUATION OF EARTH FILTER IN DB)
      READ(5,21)CR
   21 FORMAT(F10.2)
      READ(5,23) (ATTNUA(I),I=1,20)
   23 FORMAT(10F5.2/10F5.2)
C     READ IN MAXIMUM PEAK SIGNAL VOLTAGE (PEAKE) AND MAXIMUM PEAK SIGNA
C    1L CURRENT (PEAK I)
      READ(5,25)PEAKE,PEAKI
```

```
   25 FORMAT(F10.2)
      RMSE = .707*PEAKE
      RMSI = .707*PEAKI
C     PROCESS SPEECH AND NOISE
      DO27I=1,20
      ATTSP (I) = SPHSPM (I) + SIGPRO (I)
      ATTNO (I) = ACSFNO (I) + SIGPRO (I)
   27 CONTINUE
C     FIND MAXIMUM VALUES IN ARRAYS AND NORMALIZE
      XLARGE = ATTSP (1)
      DO29I=1,20
      IF(ATTSP(I).LE.XLARGE)GO TO 29
      XLARGE = ATTSP(I)
   29 CONTINUE
      DO 31 I=1,20
      ATTSP(I) = ATTSP(I) - XLARGE
      ATTNO (I) = ATTNO(I) - XLARGE
   31 CONTINUE
C     CONVERT TO ATTENUATIONS
      DO33I=1,20
      ATTSP(I) = 10.**(ATTSP(I)/20.)
      ATTNO(I) = 10.** (ATTNO(I)/20.)
   33 CONTINUE
      DO37I=1,20
      XMITSP(I) = RMSE * ATTSP(I)/Z(I)
      XMITNO(I) = RMSE*ATTNO(I)/Z(I)
   37 CONTINUE
      K=0
      J=0
   39 J=J+1
C     SUM CURRENTS FOR SPEECH
   41 TOTSPI=0.
      DO43I=1,20
      SUBTSI = XMITSP(I) + TOTSPI
      TOTSPI = SUBTSI
   43 CONTINUE
C     SUM CURRENTS FOR NOISE
      TOTNOI = 0.
      DO45I=1,20
      SUBTNI = XMITNO(I) + TOTNOI
      TOTNOI = SUBTNI
   45 CONTINUE
C     SUM THE SPEECH AND NOISE CURRENTS
      TOTALI=TOTSPI+TOTNOI
C     CALCULATE POWER THROUGH RESISTOR
      POWRES=TOTALI**2.*R
      IF(K.EQ.1)GO TO 59
      IF(J.EQ.2)GO TO 51
C     PERFORM CHECKS
      IF(POWRES.GT.500.)GO TO 47
      IF(POWRES.LE.500.)GO TO 51
C     STEP  47 FINDS THE CORRECTING FACTOR USED TO ADJUST THE CURRENTS T
C     10 PREVENT THE POWER FROM EXCEEDING 500 WATTS
   47 CORFC3=SQRT(500./POWRES)
      DO49I=1,20
      XMITSP (I) = XMITSP(I) *CORFC3
      XMITNO (I) = XMITNO(I) * CORFC3
   49 CONTINUE
C     GO BACK TO  39 TO SUM THE ADJUSTED CURRENTS AND RECALCULATE POWRES
      GO TO 39
```

```
   51 IF(TOTALI*1.414.LE.PEAKI)GO TO 59
      IF(TOTALI*1.414.GT.PEAKI)GO TO 53
   53 WRITE(6,55)TOTALI
   55 FORMAT('0',4X,'TOTAL RMS CURRENT EQUALS',F10.2,' AMPERES -- THIS E
     1XCEEDS PEAK SIGNAL CURRENT')
C     CORRECT FOR EXCESSIVE CURRENT
      CORFC4 = PEAKI/(TOTALI*1.414)
      DO57I=1,20
      XMITSP(I)=XMITSP(I)* CORFC4
      XMITNO(I) = XMITNO(I)* CORFC4
   57 CONTINUE
      K=1
C     GO BACK TO  41 TO SUM THE ADJUSTED CURRENTS AND RECALCULATE POWRES
      GO TO 41
   59 WRITE(6,61)TOTALI
   61 FORMAT('0',4X,'TOTAL RMS CURRENT EQUALS',F10.2,' AMPERES -- THIS D
     1OES NOT EXCEED PEAK SIGNAL CURRENT')
   63 WRITE(6,65)PEAKI
   65 FORMAT ('0',4X,'MAXIMUM PEAK SIGNAL CURRENT ENTERED IS',F10.2,' AM
     1PERES')
      WRITE(6,67)PEAKE
   67 FORMAT ('0',4X,'MAXIMUM PEAK SIGNAI VOLTAGE ENTERED IS',F10.2,' VO
     1LTS')
      WRITE(6,69)
   69 FORMAT('0',5X,'XMITSP(I)--RMS CURRENT FOR SPEECH SIGNAL',5X,'XMITN
     1O(I)--RMS CURRENT FOR NOISE SIGNAL    AMPERES')
      WRITE(6, 71) (XMITSP(I),XMITNO(I),I=1,20)
   71 FORMAT(14X,F10.5,35X,F10.5)
      WRITE(6,73)POWRES
   73 FORMAT('0',4X,'POWER THROUGH RESISTOR EQUALS',F10.2,' WATTS')
C     XMIT SIGNAL IN EACH BAND FOR SPEECH AND NOISE THROUGH LOOP-EARTH-
C    1FILTER-RECEIVER SYSTEM: RESULTS IN VOLT.OPEN CIRCUIT DB
      DO75I=1,20
      VOCSPH(I)=(-144.6756+20.*ALOG10(CTRFRQ(I))+40.*ALOG10(CR)+40.*ALOG
     110(CT)-42.6234+20.*ALOG10(XMITSP(I))+ATTNUA(I))
   75 CONTINUE
      DO77I=1,20
      VOCNOI(I)=(-144.6756+20.*ALOG10(CTRFRQ(I))+40.*ALOG10(CR)+40.*ALOG
     110(CT)-42.6234+20.*ALOG10(XMITNO(I))+ATTNUA(I))
   77 CONTINUE
C     PROCESS THERMAL-ATMOSPHERIC NOISE THROUGH RECEIVING LOOP
C     READ THATNO DB
      READ(5,83) (THATNO(I),I=1,20)
   83 FORMAT(10F5.2/10F5.2)
      DO85I=1,20
      ELMGNO(I)=(-144.6756+20.*ALOG10(CTRFRQ(I))+40.*ALOG10(CR)+THATNO
     1(I))
   85 CONTINUE
C     READ IN RELATIVE RECEIVER RESPONSE(RERERE(I))
      READ(5,90) (RERERE(I),I=1,20)
   90 FORMAT(10F5.2/10F5.2)
C     PROCESS VOCSPH(I),VOCNOI(I), AND ELMGNO(I) THROUGH RERERE(I)
      DO92I=1,20
      VOCSPH(I)=VOCSPH(I)+RERERE(I)
      VOCNOI(I)=VOCNOI(I)+RERERE(I)
      ELMGNO(I)=ELMGNO(I)+RERERE(I)
   92 CONTINUE
      WRITE(6,79)
   79 FORMAT('0',5X,'VOCSPH DB--OPEN CIRCUIT VOLT. FOR SPEECH',5X,'VOCNO
     1I DB--OPEN CIRCUIT VOLT. FOR NOISE   REF. 1 VOLT')
```

```
      WRITE(6, 81) (VOCSPH(I),VOCNOI(I),I=1,20)
   81 FORMAT(5X,E20.3,25X,E20.3)
      WRITE(6,87)
   87 FORMAT('0',5X,'ELMGNO DB -- ELECTROMAGNETIC NOISE   REF. 1 VOLT')
      WRITE(6,89) (ELMGNO(I),I=1,20)
   89 FORMAT(5X,E20.3)
C     ADD ELMGNO(I) TO VOCNOI(I) TO GET SUBTOTAL 1(SBTNO1) OF NOISE IN
C    1WATTS
      DO91I=1,20
      SBTNO1(I)=10.**(ELMGNO(I)/10.)+10.**(VOCNOI(I)/10.)
   91 CONTINUE
      WRITE(6,93)
   93 FORMAT('0',5X,'SBTNO1--ELMGNO+VOCNOI--POWER BEFORE GAIN   WATTS')
      WRITE(6,95) (SBTNO1(I),I=1,20)
   95 FORMAT(8X,E15.3)
C     CALCULATE ELECTRONIC NOISE POWER SPECTRUM
      DO97I=1,20
      ELNOPO(I)=6.987E-15*BNDWTH(I)
   97 CONTINUE
C     MODIFY ELNOPO(I) BY RERERE(I)
      DO98I=1,20
      ELNOPO(I)=ELNOPO(I)*10.**(RERERE(I)/10.)
   98 CONTINUE
      WRITE(6,99)
   99 FORMAT('0',5X,'ELNOPO--ELECTRONIC NOISE POWER--BEFORE GAIN   WATTS
     1')
      WRITE(6,101) (ELNOPO(I),I=1,20)
  101 FORMAT(8X,E15.3)
C     ADD ELNOPO(I) TO SBTNO1(I) TO GET SBTNO2(I)
      DO103I=1,20
      SBTNO2(I)=ELNOPO(I)+SBTNO1(I)
  103 CONTINUE
C     CHANGE VOCSPH(I) DB TO POWER
      DO105I=1,20
      POWSPH(I)=10.**(VOCSPH(I)/10.)
  105 CONTINUE
      WRITE(6,107)
  107 FORMAT('0',5X,'POWSPH--POWER OF SPEECH--BEFORE GAIN   WATTS')
      WRITE(6,109) (POWSPH(I),I=1,20)
  109 FORMAT(8X,E15.3)
C     ADD 93DB TO EACH BAND
      DO111I=1,20
      POWSPH(I)=POWSPH(I)*2.E9
      POWNOI(I)=SBTNO2(I)*2.E9
  111 CONTINUE
      WRITE(6,113)
  113 FORMAT('0',5X,'POWER FOR SPEECH SIGNAL',10X,'POWER FOR NOISE',5X,'
     1WATTS',5X,'AFTER 93 DB GAIN--BEFORE ADJUST TO .01 WATTS')
      WRITE(6,115) (POWSPH(I),POWNOI(I),I=1,20)
  115 FORMAT(8X,E15.3, 15X,E15.3)
C     GO THROUGH PROCESS OF ADJUSTING POWER TO .01WATTS
C     SUM POWER OF EACH BAND OF SPEECH SIGNAL TO GIVE TOPS2
      TOPS2=0.
      DO117I=1,20
      SUBTS2=POWSPH(I)+TOPS2
      TOPS2=SUBTS2
  117 CONTINUE
C     SUM POWER OF EACH BAND OF NOISE TO GIVE TOPN2
      TOPN2=0.
      DO119I=1,20
```

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      SUBTN2=POWNOI(I)+TOPN2
      TOPN2=SUBTN2
  119 CONTINUE
C     SUM THE TWO POWER LEVELS
      SUMPO2=TOPS2+TOPN2
C     FIND CORRECTING FACTORS FOR SPEECH AND NOISE POWER
      CORFC2=.01/SUMPO2
C     ADJUST POWSPH(I) AND POWNOI(I)
      DO121I=1,20
      POWSPH(I)=POWSPH(I)*CORFC2
      POWNOI(I)=POWNOI(I)*CORFC2
  121 CONTINUE
      WRITE(6,123)CORFC2
  123 FORMAT('0',5X,'CORFC2 EQUALS  ',F5.3)
C     TOTAL POWER IS NOW .01W
C     CONVERT TO VOLTS
      DO125I=1,20
      VOSPH2(I)=SQRT(POWSPH(I)*8.)
      VONOI2(I)=SQRT(POWNOI(I)*8.)
  125 CONTINUE
C     CONVERT TO DB
      DO127I=1,20
      VOSPH2(I)=20.*ALOG10(VOSPH2(I))
      VONOI2(I)=20.*ALOG10(VONOI2(I))
  127 CONTINUE
C     READ IN SPEAKER SPECTRUM AND PROCESS THROUGH SPEAKER
      READ(5,129)(SPKSPC(I),I=1,20)
  129 FORMAT(10F5.2/10F5.2)
      DO131I=1,20
      SPLSPH(I)=VOSPH2(I)+SPKSPC(I)
      SPLNOI(I)=VONOI2(I)+SPKSPC(I)
  131 CONTINUE
      DO133I=1,20
      SPLSNZ(I)=SPLSPH(I)-SPLNOI(3)
      SPLNNZ(I)=SPLNOI(I)-SPLNOI(3)
  133 CONTINUE
      WRITE(6,135)
  135 FORMAT('0',5X,'SPL OF SPEECH',10X,'SPL OF NOISE',2X,'BEFORE SPECTR
     1UM LEVEL CALCULATION',3X,'NORMALIZED SPEECH',6X,'NORMALIZED NOISE'
     2)
      WRITE(6,136)
  136 FORMAT(43X,'@.5 METERS  REF. .0002 N/SQ. METER')
      WRITE(6,137)(SPLSPH(I),SPLNOI(I),SPLSNZ(I),SPLNNZ(I),I=1,20)
  137 FORMAT(10X,F10.2,10X,F10.2,37X,F10.2,10X,F10.2)
C     CORRECT FOR BANDWITH TO GET SPECTRUM LEVEL
      DO139I=1,20
      SPLSPH(I)=SPLSPH(I)-10.*ALOG10(BNDWTH(I))
      SPLNOI(I)=SPLNOI(I)-10.*ALOG10(BNDWTH(I))
  139 CONTINUE
      WRITE(6,141)
  141 FORMAT('0',5X,'SPL OF SPEECH',10X,'SPL OF NOISE',10X,'SPECTRUM LEV
     1ELS')
      WRITE(6,143)(SPLSPH(I),SPLNOI(I),I=1,20)
  143 FORMAT(10X,F10.2,10X,F10.2)
C     CALCULATE ARTICULATION INDEX
C     ADD 12 DB PEAK CORRECTION
      DO145I=1,20
      SPLSPH(I)=SPLSPH(I)+12.
  145 CONTINUE
C     FIND DIFFERENCE BETWEEN SPEECH AND NOISE SPECTRUM LEVELS
```

```
      DO151I=1,20
      IF(SPLSPH(I)-SPLNOI(I))147,147,149
  147 DIFF(I)=0.
      GO TO 151
  149 DIFF(I)=SPLSPH(I)-SPLNOI(I)
      S=30.
      IF(DIFF(I).GE.S)DIFF(I)=30.
  151 CONTINUE
C     SUM ALL THE DIFF AND DIVIDE BY 600
      TOTAL=0.
      DO153I=1,20
      X=DIFF(I)+TOTAL
      TOTAL=X
  153 CONTINUE
      AI=TOTAL/600.
      WRITE(6,155)AI
  155 FORMAT('0',5X,'ARTICULATION INDEX IS',F10.2)
      WRITE(6,157)
  157 FORMAT('1',55X,'DATA SHEET')
      WRITE(6,159)CT,RHO,DIM,CR,PEAKE,PEAKI
  159 FORMAT('0',4X,'CT ENTERED IS',F10.2,' FT'/5X,'RHO ENTERED IS',F10.
     12,' OHMS/KFT'/5X,'DIM ENTERED IS',F10.2,' MILS'/5X,'CR ENTERED IS'
     2,F10.2,' FT'/5X,'PEAKE ENTERED IS',F10.2,' VOLTS'/5X,'PEAKI ENTERE
     3D IS',F10.2,' AMPS')
      WRITE(6,161)
  161 FORMAT('0',5X,'CTRFRQ',5X,'SPHSPM',5X,'SIGPRO',5X,'BNDWTH',5X,'ACS
     1FNO',5X,'ATTNUA',5X,'THATNO',5X,'SPKSPC',5X,'RERERE')
      WRITE(6,163)(CTRFRQ(I),SPHSPM(I),SIGPRO(I),BNDWTH(I),ACSFNO(I),ATT
     1NUA(I),THATNO(I),SPKSPC(I),RERERE(I),I=1,20)
  163 FORMAT('0',1X,F10.2,1X,F10.2,1X,F10.2,1X,F10.2,1X,F10.2,1X,F10.2,1
     1X,F10.2,1X,F10.2,1X,F10.2)
      STOP
      END

,COMPILE TIME=     0.45 SEC,EXECUTION TIME=     0.31 SEC, CORE USED-OBJECT=   12736 ,ARRAYS=    2400 ,TOTAL=   49152  BYTES
```

DATA SHEET

CT ENTERED IS 515.35 FT
RHO ENTERED IS 1.59 OHMS/KFT
DIM ENTERED IS 80.81 MILS
CF ENTERED IS 100.00 FT
PEAKE ENTERED IS 76.40 VOLTS
PEAKI ENTERED IS 19.10 AMPS

SIGNAL PROCESSING SP-7

CTRFRQ	SPHSPM	SIGPRO	BNDWTH	ACSFNO	ATTNUA	THATNO	SPKSPC	RERERE
270.00	50.00	-12.00	130.00	30.00	-176.00	-113.00	69.00	-16.00
380.00	50.00	-5.00	100.00	30.00	-180.00	-116.00	70.00	-8.00
490.00	48.50	0.00	130.00	28.50	-183.00	-118.00	71.00	-3.00
630.00	45.50	0.00	140.00	25.50	-187.00	-121.00	72.00	0.00
770.00	42.50	0.00	140.00	22.50	-190.00	-122.00	73.00	0.00
920.00	40.00	0.00	160.00	20.00	-193.00	-124.00	74.00	0.00
1070.00	37.50	0.00	150.00	17.50	-195.00	-125.00	74.00	0.00
1230.00	35.00	0.00	160.00	15.00	-197.00	-126.00	74.00	0.00
1400.00	33.00	0.00	170.00	13.00	-200.00	-128.00	74.00	0.00
1570.00	31.50	0.00	180.00	11.50	-203.00	-129.00	74.00	0.00
1740.00	30.50	0.00	170.00	10.50	-206.00	-130.00	74.00	0.00
1920.00	28.50	0.00	190.00	8.50	-209.00	-131.00	75.00	0.00
2130.00	27.00	0.00	220.00	7.00	-212.00	-132.00	77.00	0.00
2370.00	26.00	0.00	260.00	6.00	-216.00	-133.00	79.00	0.00
2660.00	24.50	0.00	320.00	4.50	-219.00	-134.00	80.00	-1.00
3000.00	23.00	0.00	380.00	3.00	-224.00	-135.00	79.00	-3.00
3400.00	21.50	-2.00	450.00	1.50	-229.00	-136.00	81.00	-5.00
3950.00	20.00	-6.00	600.00	0.00	-235.00	-137.00	83.00	-7.00
4650.00	19.00	-8.00	800.00	-1.00	-241.00	-138.00	85.00	-9.00
5600.00	18.00	-13.00	1050.00	-2.00	-251.00	-140.00	86.00	-13.00

ACCOUNT: P0846
DATE: 01/24/77 IDENT:
USER: BIENVENUE GORDON
DESTINATION: AA
OS-21.8 HASP-2.T4E 370/168

MAXIMUM TIME (SEC): 25 NET CPU (SEC): 1
ACTUAL TIME, INCLUDING 2.0 SEC SYSTEM TIME: 5 @ $.11/SEC = $ 0.55
LINES PRINTED: 600 CARDS PUNCHED: 0 @ $.35/100 = $ 0.00
MAXIMUM RECORDS: 2500 TOTAL RECORDS: 600 @ $.07/100 = $ 0.42
CARDS READ: 361 ***** TOTAL COST = $ 0.97 JOB NAME AU 1 22446

RADIUS OF XMIT LOOP 2500.01 CM
WIRE DIAMETER 0.2053 CM
INDUCTANCE 0.000328 HENRYS
RESISTANCE 0.8184 OHMS
CAPACITANCE 1000000.0000 FARADS

IMPEDANCE--Z(I) OHMS	CENTER FREQUENCY--CTRFRQ(I) HZ.
0.989447	270.00
1.132401	380.00
1.299355	490.00
1.534120	630.00
1.784662	770.00
2.064085	920.00
2.350912	1070.00
2.662324	1230.00
2.997452	1400.00
3.335667	1570.00
3.676118	1740.00
4.038401	1920.00
4.462823	2130.00
4.949597	2370.00
5.539569	2660.00
6.233041	3000.00
7.050614	3400.00
8.176848	3950.00
9.612477	4650.00
11.563280	5600.00

TOTAL RMS CURRENT EQUALS 24.72 AMPERES -- THIS EXCEEDS PEAK SIGNAL CURRENT

TOTAL RMS CURRENT EQUALS 13.51 AMPERES -- THIS DOES NOT EXCEED PEAK SIGNAL CURRENT

MAXIMUM PEAK SIGNAL CURRENT ENTERED IS 19.10 AMPERES

MAXIMUM PEAK SIGNAL VOLTAGE ENTERED IS 76.40 VOLTS

XMITSP(I)--RMS CURRENT FOR SPEECH SIGNAL	XMITNO(I)--RMS CURRENT FOR NOISE SIGNAL AMPERES
1.23267	0.12327
2.41123	0.24112
3.14421	0.31442
1.88530	0.18853
1.14732	0.11473
0.74389	0.07439
0.48978	0.04898
0.32432	0.03243
0.22882	0.02288
0.17300	0.01730
0.13991	0.01399
0.10116	0.01012
0.07702	0.00770
0.06190	0.00619
0.04653	0.00465
0.03480	0.00348
0.02056	0.00206
0.00941	0.00094
0.00567	0.00057
0.00236	0.00024

POWER THROUGH RESISTOR EQUALS 149.32 WATTS

VOCSPH DB--OPEN CIRCUIT VOLT. FOR SPEECH

-0.140E 03
-0.128E 03
-0.121E 03
-0.124E 03
-0.130E 03
-0.135E 03
-0.139E 03
-0.144E 03
-0.149E 03
-0.153E 03
-0.157E 03
-0.162E 03
-0.167E 03
-0.171E 03
-0.177E 03
-0.185E 03
-0.196E 03
-0.209E 03
-0.220E 03
-0.240E 03

VOCNOI DB--OPEN CIRCUIT VOLT. FOR NOISE   REF. 1 VOLT

-0.160E 03
-0.148E 03
-0.141E 03
-0.144E 03
-0.150E 03
-0.155E 03
-0.159E 03
-0.164E 03
-0.169E 03
-0.173E 03
-0.177E 03
-0.182E 03
-0.187E 03
-0.191E 03
-0.197E 03
-0.205E 03
-0.216E 03
-0.229E 03
-0.240E 03
-0.260E 03

ELMGNO DB -- ELECTROMAGNETIC NOISE   REF. 1 VOLT

-0.145E 03
-0.137E 03
-0.132E 03
-0.130E 03
-0.129E 03
-0.129E 03
-0.129E 03
-0.129E 03
-0.130E 03
-0.130E 03
-0.130E 03
-0.130E 03
-0.130E 03
-0.130E 03
-0.131E 03
-0.133E 03
-0.135E 03
-0.137E 03
-0.138E 03
-0.143E 03

SBTNO1--ELMGNO+VOCNOI--POWER BEFORE GAIN   WATTS

0.322E-14
0.213E-13
0.728E-13
0.111E-12
0.128E-12
0.115E-12
0.123E-12
0.130E-12
0.106E-12
0.106E-12
0.103E-12
0.998E-13
0.975E-13
0.959E-13
0.762E-13

0.486E-13
0.313E-13
0.212E-13
0.147E-13
0.536E-14

ELNOPO--ELECTRONIC NOISE POWER--BEFORE GAIN WATTS

0.228E-13
0.111E-12
0.455E-12
0.978E-12
0.978E-12
0.112E-11
0.105E-11
0.112E-11
0.119E-11
0.126E-11
0.119E-11
0.133E-11
0.154E-11
0.182E-11
0.178E-11
0.133E-11
0.994E-12
0.836E-12
0.704E-12
0.368E-12

POWSPH--POWER OF SPEECH--BEFORE GAIN WATTS

0.918E-14
0.175E-12
0.783E-12
0.370E-12
0.103E-12
0.308E-13
0.114E-13
0.417E-14
0.135E-14
0.486E-15
0.196E-15
0.624E-16
0.223E-16
0.710E-17
0.201E-17
0.286E-18
0.256E-19
0.115E-20
0.912E-22
0.915E-24

POWER FOR SPEECH SIGNAL	POWER FOR NOISE	WATTS	AFTER 93 DB GAIN--BEFORE ADJUST TO .01 WATTS
0.184E-04	0.521E-04		
0.350E-03	0.264E-03		
0.157E-02	0.106E-02		
0.740E-03	0.218E-02		
0.205E-03	0.221E-02		
0.617E-04	0.247E-02		
0.228E-04	0.234E-02		
0.834E-05	0.249E-02		
0.270E-05	0.259E-02		

0.971E-06	0.273E-02
0.391E-06	0.258E-02
0.125E-06	0.285E-02
0.446E-07	0.327E-02
0.142E-07	0.383E-02
0.403E-08	0.370E-02
0.571E-09	0.276E-02
0.511E-10	0.205E-02
0.229E-11	0.172E-02
0.182E-12	0.144E-02
0.183E-14	0.746E-03

CORFC2 EQUALS 0.216

SPL OF SPEECH	SPL OF NOISE	BEFORE SPECTRUM LEVEL CALCULATION @.5 METERS REF. .0002 N/SQ. METER	NORMALIZED SPEECH	NORMALIZED NOISE
24.01	28.54		-19.60	-15.07
37.81	36.59		-5.80	-7.02
45.32	43.61		1.71	0.00
43.06	47.76		-0.55	4.14
38.49	48.83		-5.12	5.21
34.28	50.30		-9.34	6.68
29.96	50.07		-13.65	6.46
25.59	50.35		-18.02	6.73
20.68	50.50		-22.93	6.89
16.25	50.73		-27.36	7.12
12.30	50.49		-31.31	6.88
8.34	51.93		-35.28	8.32
5.87	54.52		-37.74	10.91
2.90	57.20		-40.71	13.59
-1.58	58.06		-45.19	14.45
-11.06	55.78		-54.67	12.17
-19.54	56.49		-63.15	12.88
-31.02	57.72		-74.64	14.11
-40.01	58.95		-83.62	15.34
-59.00	57.10		-102.61	13.49

SPL OF SPEECH	SPL OF NOISE	SPECTRUM LEVELS
2.88	7.40	
17.81	16.59	
24.18	22.47	
21.60	26.30	
17.03	27.36	
12.24	28.25	
8.20	28.31	
3.55	28.30	
-1.62	28.20	
-6.30	28.18	
-10.01	28.19	
-14.45	29.14	
-17.55	31.10	
-21.25	33.05	
-26.63	33.01	
-36.85	29.98	
-46.07	29.96	
-58.81	29.94	
-69.04	29.92	
-89.21	26.89	

ARTICULATION INDEX IS 0.07

DATA SHEET

SIGNAL PROCESSING SP-25

CT ENTERED IS 515.35 FT
RHO ENTERED IS 1.59 OHMS/KFT
DIM ENTERED IS 80.81 MILS
CR ENTERED IS 100.00 FT
PEAKE ENTERED IS 29.40 VOLTS
PEAKI ENTERED IS 49.60 AMPS

CTRFRQ	SPHSPM	SIGPRO	BNDWTH	ACSFNC	ATTNUA	THATNO	SPKSPC	RERERE
270.00	50.00	-12.00	130.00	30.00	-176.00	-113.00	69.00	-16.00
380.00	50.00	-5.00	100.00	30.00	-180.00	-116.00	70.00	-8.00
490.00	48.50	0.00	130.00	28.50	-183.00	-118.00	71.00	-3.00
630.00	45.50	0.00	140.00	25.50	-187.00	-121.00	72.00	0.00
770.00	42.50	0.00	140.00	22.50	-190.00	-122.00	73.00	0.00
920.00	40.00	0.00	160.00	20.00	-193.00	-124.00	74.00	0.00
1070.00	37.50	0.00	150.00	17.50	-195.00	-125.00	74.00	0.00
1230.00	35.00	0.00	160.00	15.00	-197.00	-126.00	74.00	0.00
1400.00	33.00	0.00	170.00	13.00	-200.00	-128.00	74.00	0.00
1570.00	31.50	0.00	180.00	11.50	-203.00	-129.00	74.00	0.00
1740.00	30.50	0.00	170.00	10.50	-206.00	-130.00	74.00	0.00
1920.00	28.50	0.00	190.00	8.50	-209.00	-131.00	75.00	0.00
2130.00	27.00	0.00	220.00	7.00	-212.00	-132.00	77.00	0.00
2370.00	26.00	0.00	260.00	6.00	-216.00	-133.00	79.00	0.00
2660.00	24.50	0.00	320.00	4.50	-219.00	-134.00	80.00	-1.00
3000.00	23.00	0.00	380.00	3.00	-224.00	-135.00	79.00	-3.00
3400.00	21.50	-2.00	450.00	1.50	-229.00	-136.00	81.00	-5.00
3950.00	20.00	-6.00	600.00	0.00	-235.00	-137.00	83.00	-7.00
4650.00	19.00	-8.00	800.00	-1.00	-241.00	-138.00	85.00	-9.00
5600.00	18.00	-13.00	1050.00	-2.00	-251.00	-140.00	86.00	-13.00

ACCOUNT: P0846 MAXIMUM TIME (SEC): 25 NET CPU (SEC): 1
DATE: 01/26/77 IDENT: ACTUAL TIME, INCLUDING 2.0 SEC SYSTEM TIME: 5 @ $.11/SEC = $ 0.55
USER: BIENVENUE GORDON LINES PRINTED: 600 CARDS PUNCHED: 0 @ $.35/100 = $ 0.00
DESTINATION: AA MAXIMUM RECORDS: 2500 TOTAL RECORDS: 600 @ $.07/100 = $ 0.42
OS-21.8 HASP-2.T4E 370/168 CARDS READ: 362 ***** TOTAL COST = $ 0.97 JOB NAME AU 1 45743

RADIUS OF XMIT LOOP 2500.01 CM
WIRE DIAMETER 0.2053 CM
INDUCTANCE 0.000328 HENRYS
RESISTANCE 0.8184 OHMS
CAPACITANCE 1000000.0000 FARADS

IMPEDANCE--Z(I) OHMS	CENTER FREQUENCY--CTRFRQ(I) HZ.
0.989447	270.00
1.132401	380.00
1.299355	490.00
1.534120	630.00
1.784662	770.00
2.064085	920.00
2.350912	1070.00
2.662324	1230.00
2.997452	1400.00
3.335667	1570.00
3.676118	1740.00
4.038401	1920.00
4.462823	2130.00
4.949597	2370.00
5.539569	2660.00
6.233041	3000.00
7.050614	3400.00
8.176848	3950.00
9.612477	4650.00
11.563280	5600.00

TOTAL RMS CURRENT EQUALS 24.72 AMPERES -- THIS DOES NOT EXCEED PEAK SIGNAL CURRENT

MAXIMUM PEAK SIGNAL CURRENT ENTERED IS 49.60 AMPERES

MAXIMUM PEAK SIGNAL VOLTAGE ENTERED IS 29.40 VOLTS

XMITSP(I)--RMS CURRENT FOR SPEECH SIGNAL	XMITNO(I)--RMS CURRENT FOR NOISE SIGNAL AMPERES
2.25564	0.22556
4.41227	0.44123
5.75353	0.57535
3.44987	0.34499
2.09945	0.20995
1.36124	0.13612
0.89624	0.08962
0.59347	0.05935
0.41871	0.04187
0.31658	0.03166
0.25602	0.02560
0.18512	0.01851
0.14095	0.01409
0.11326	0.01133
0.08515	0.00852
0.06367	0.00637
0.03762	0.00376
0.01722	0.00172
0.01037	0.00104
0.00432	0.00043

POWER THROUGH RESISTOR EQUALS 500.00 WATTS

VOCSPH DB--OPEN CIRCUIT VOLT. FOR SPEECH	VOCNOI DB--OPEN CIRCUIT VOLT. FOR NOISE REF. 1 VOLT
-0.135E 03	-0.155E 03

-0.122E 03	-0.142E 03
-0.116E 03	-0.136E 03
-0.119E 03	-0.139E 03
-0.125E 03	-0.145E 03
-0.130E 03	-0.150E 03
-0.134E 03	-0.154E 03
-0.139E 03	-0.159E 03
-0.143E 03	-0.163E 03
-0.148E 03	-0.168E 03
-0.152E 03	-0.172E 03
-0.157E 03	-0.177E 03
-0.161E 03	-0.181E 03
-0.166E 03	-0.186E 03
-0.172E 03	-0.192E 03
-0.180E 03	-0.200E 03
-0.191E 03	-0.211E 03
-0.204E 03	-0.224E 03
-0.215E 03	-0.235E 03
-0.235E 03	-0.255E 03

ELMGNO DB -- ELECTROMAGNETIC NOISE REF. 1 VOLT

-0.145E 03
-0.137E 03
-0.132E 03
-0.130E 03
-0.129E 03
-0.129E 03
-0.129E 03
-0.129E 03
-0.130E 03
-0.130E 03
-0.130E 03
-0.130E 03
-0.130E 03
-0.130E 03
-0.131E 03
-0.133E 03
-0.135E 03
-0.137E 03
-0.138E 03
-0.143E 03

SBTNO1--ELMGNO+VOCNOI--POWER BEFORE GAIN WATTS

0.343E-14
0.254E-13
0.912E-13
0.120E-12
0.131E-12
0.116E-12
0.124E-12
0.130E-12
0.106E-12
0.106E-12
0.103E-12
0.998E-13
0.975E-13
0.959E-13
0.762E-13
0.486E-13
0.313E-13

0.212E-13
0.147E-13
0.536E-14

ELNOPO--ELECTRONIC NOISE POWER--BEFORE GAIN WATTS

0.228E-13
0.111E-12
0.455E-12
0.978E-12
0.978E-12
0.112E-11
0.105E-11
0.112E-11
0.119E-11
0.126E-11
0.119E-11
0.133E-11
0.154E-11
0.182E-11
0.178E-11
0.133E-11
0.994E-12
0.836E-12
0.704E-12
0.368E-12

POWSPH--POWER OF SPEECH--BEFORE GAIN WATTS

0.307E-13
0.585E-12
0.262E-11
0.124E-11
0.343E-12
0.103E-12
0.382E-13
0.140E-13
0.451E-14
0.163E-14
0.655E-15
0.209E-15
0.747E-16
0.238E-16
0.674E-17
0.956E-18
0.856E-19
0.384E-20
0.306E-21
0.306E-23

POWER FOR SPEECH SIGNAL	POWER FOR NOISE WATTS AFTER 93 DB GAIN--BEFORE ADJUST TO .01 WATTS
0.615E-04	0.525E-04
0.117E-02	0.272E-03
0.525E-02	0.109E-02
0.248E-02	0.220E-02
0.687E-03	0.222E-02
0.207E-03	0.247E-02
0.764E-04	0.234E-02
0.279E-04	0.250E-02
0.903E-05	0.259E-02
0.325E-05	0.273E-02
0.131E-05	0.258E-02

0.418E-06	0.285E-02
0.149E-06	0.327E-02
0.476E-07	0.383E-02
0.135E-07	C.370E-02
0.191E-08	0.276E-02
0.171E-09	0.205E-02
0.767E-11	0.172E-02
0.611E-12	C.144E-02
0.612E-14	0.746E-03

CORFC2 EQUALS 0.187

SPL OF SPEECH	SPL OF NOISE	BEFORE SPECTRUM LEVEL CALCULATION @.5 METERS REF. .0002 N/SQ. METER	NORMALIZED SPEECH	NORMALIZED NOISE
28.65	27.96		-14.50	-15.18
42.44	36.11		-0.70	-7.03
49.96	43.14		6.81	0.00
47.70	47.17		4.55	4.03
43.13	48.22		-0.02	5.07
38.91	49.68		-4.24	6.54
34.59	49.46		-8.55	6.31
30.22	49.73		-12.92	6.59
25.31	49.89		-17.83	6.74
20.88	50.12		-22.26	6.97
16.93	49.88		-26.21	6.73
12.97	51.31		-30.18	8.17
10.50	53.90		-32.64	10.76
7.53	56.59		-35.61	13.44
3.06	57.45		-40.09	14.30
-6.42	55.17		-49.57	12.02
-14.91	55.88		-58.05	12.73
-26.39	57.10		-69.54	13.96
-35.38	58.33		-78.52	15.19
-54.37	56.49		-97.51	13.34

SPL OF SPEECH	SPL OF NOISE	SPECTRUM LEVELS
7.51	6.82	
22.44	16.11	
28.82	22.01	
26.24	25.71	
21.66	26.76	
16.87	27.64	
12.83	27.70	
8.18	27.69	
3.01	27.58	
-1.67	27.56	
-5.37	27.57	
-9.82	28.53	
-12.92	30.48	
-16.62	32.44	
-22.00	32.39	
-32.22	29.37	
-41.44	29.35	
-54.17	29.32	
-64.41	29.30	
-84.58	26.27	

ARTICULATION INDEX IS 0.12

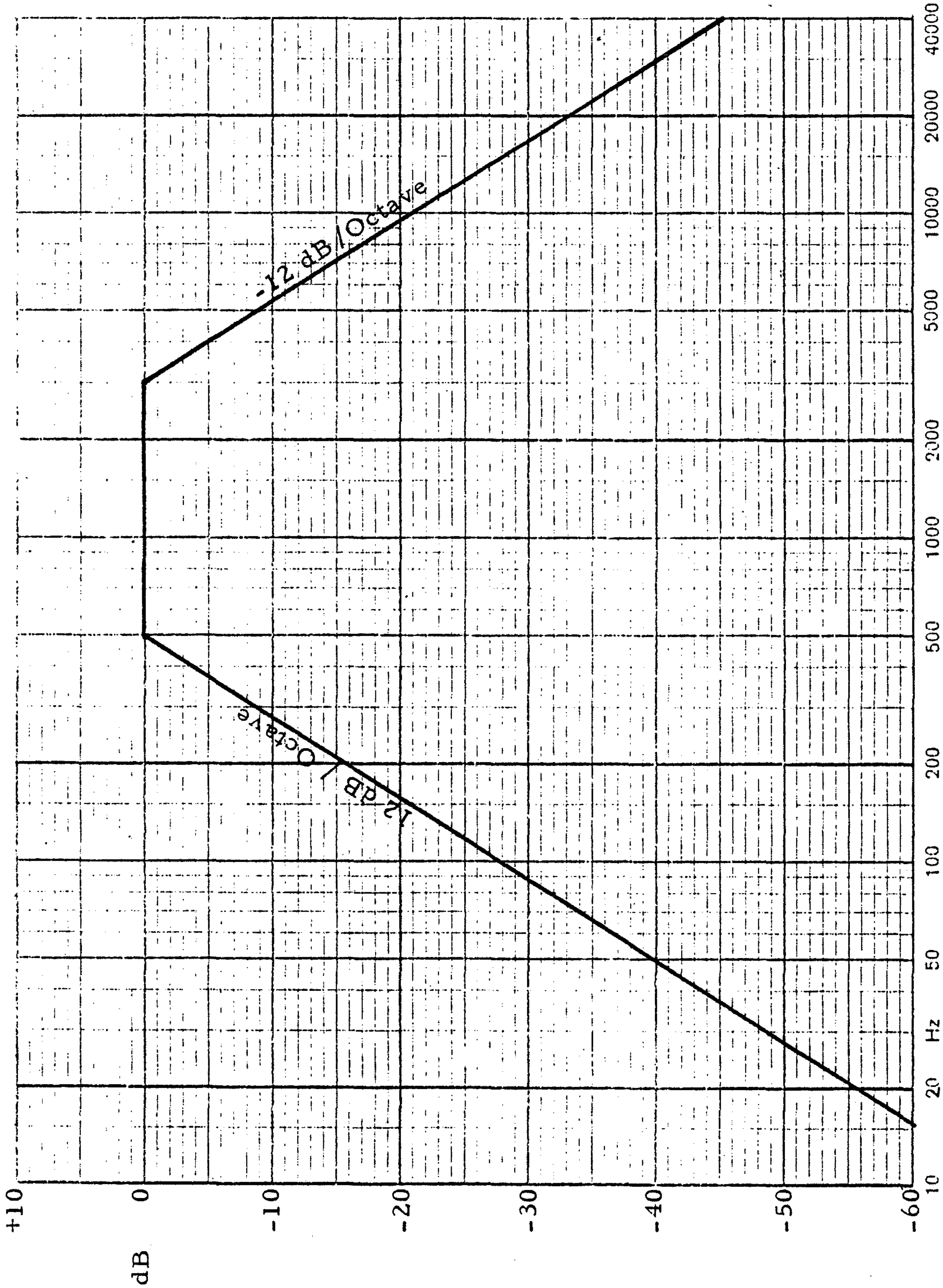

Fig. A-1 Basic filter shape SP 7.

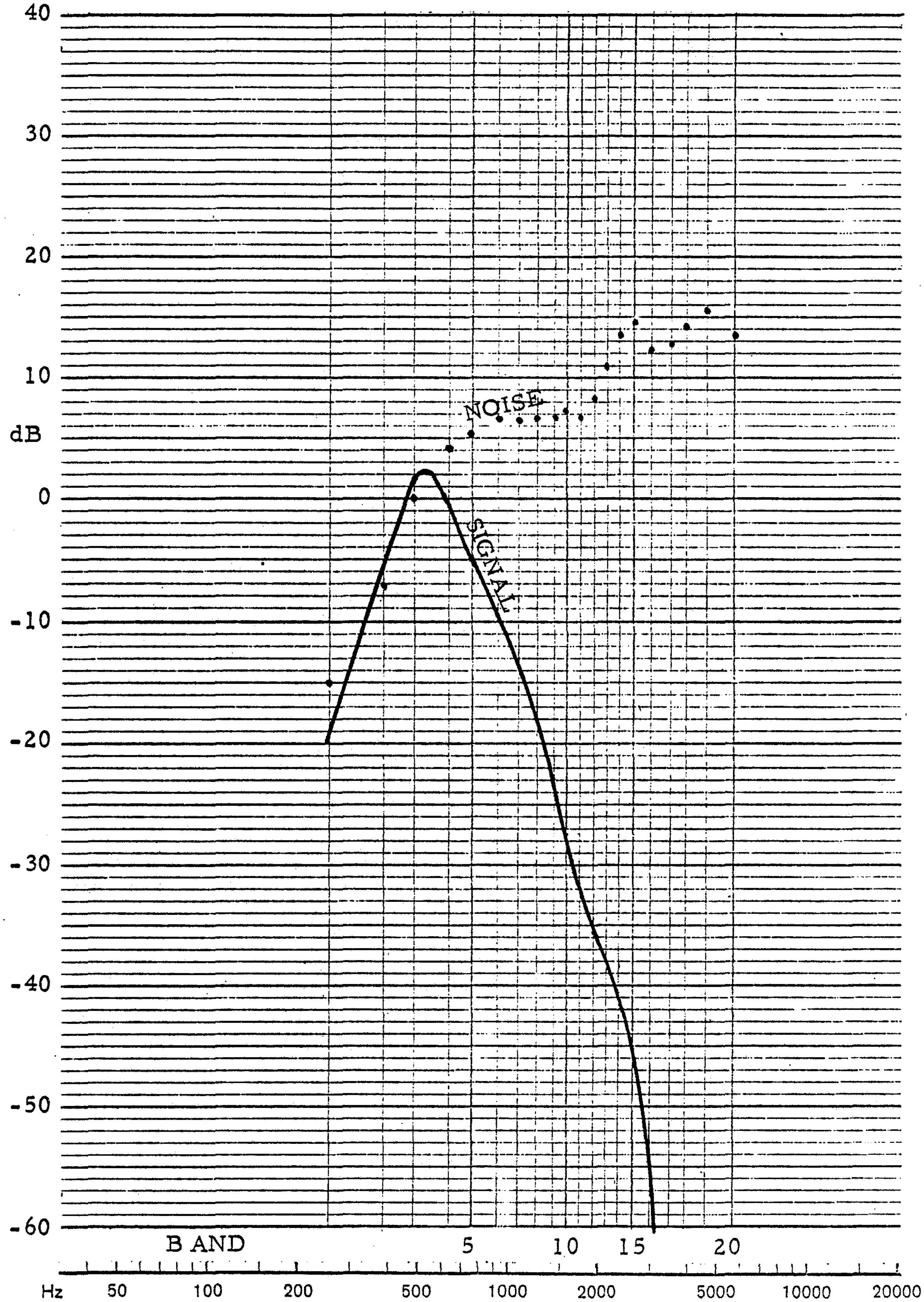

Fig. A2.Signal and noise plots for SP 7.

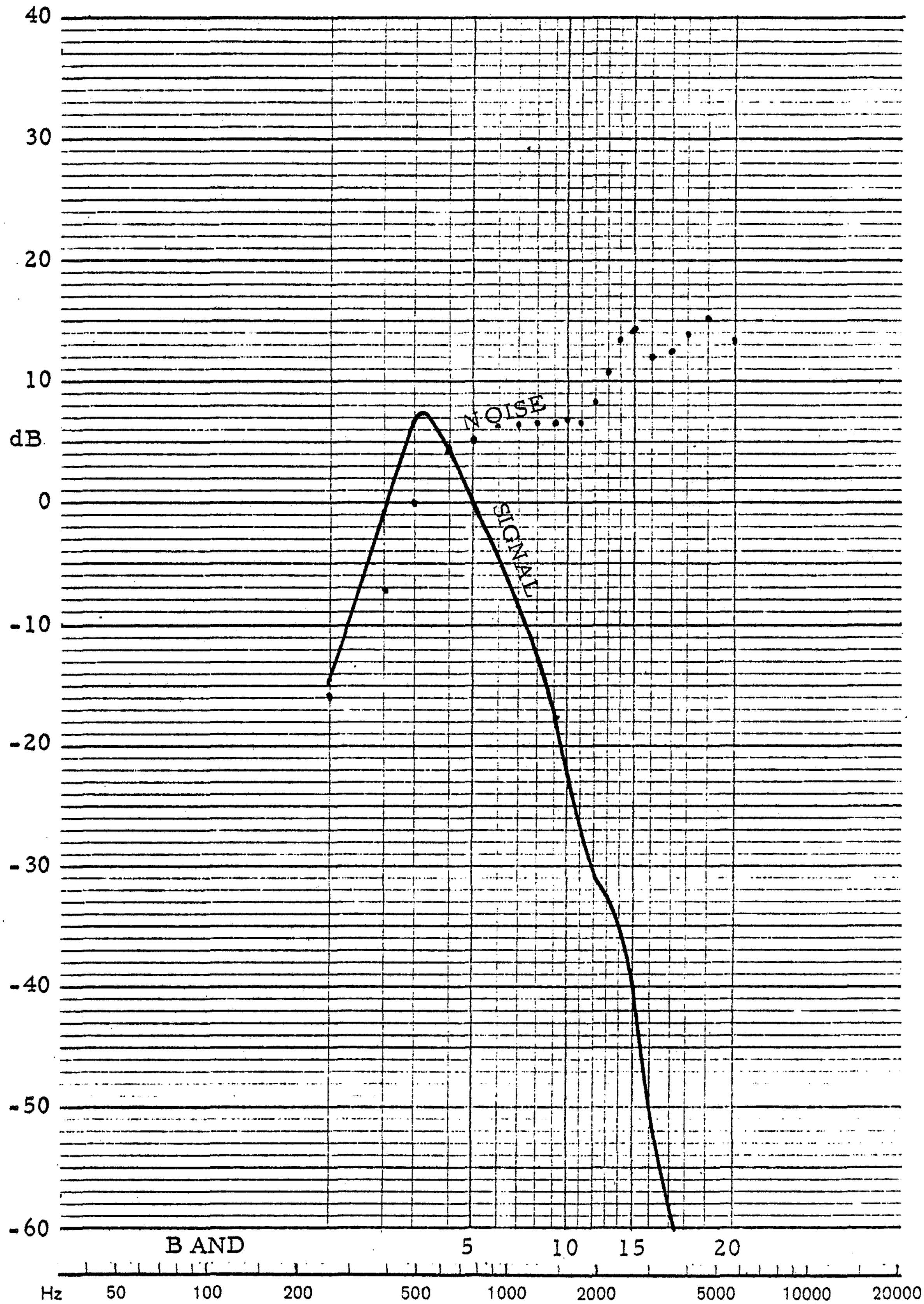

Fig.A3.Signal and noise plots for SP 25 .

Appendix B

TESTING SPEECH INTELLIGIBILITY OF COMMUNICATIONS SYSTEMS FOR USE IN COAL MINE ENVIRONMENTS

Introduction: In selecting a test procedure for evaluating the intelligibility of speech signals through any communications system, it is necessary to consider the many factors which contribute to the intelligibility of a given speech sound. Thus, this discussion will begin with a review of existing literature on the topic of speech intelligibility.

Whenever phenomena related to intelligibility are considered it is useful to keep in mind the entire communication situation. Shannon and Weaver (1949, Ch. 1) have characterized communication as a phenomenon made up of three basic components; the transmitter; the communication channel, and the receiver. Their model considers many parameters of communication and may, therefore, be further broken down into many factors affecting communication. For purposes of this paper, however, it will be sufficient to consider the three broad categories described above remaining cognizant that each category is made up of many component factors. The paper will present a discussion of the phenomenon of speech intelligibility and will be organized along the lines of the Shannon and Weaver model. Webster (1969) presents a discussion of speech intelligibility which covers source and channel parameters quite well and which has been used as a major source for the discussion of these areas. The paper will present an overall discussion of speech intelligibility considering source, transmission channel and receiver phenonema. The variables affecting speech intelligibility will be discussed in terms of an information units concept. Following this discussion, methods of testing speech intelligibility in specific contexts will be considered.

Finally, the information reviewed will be utilized in the development of a recommended procedure for evaluating speech intelligibility through communications systems used in a coal mine environment. Specific systems to be considered include active hearing protectors and emergency communications systems.

The Phenomenon of Speech Intelligibility: Webster (1969) states that speech intelligibility, in an everyday situation, is dependent upon the level and spectrum of the speech and upon the parameters of any interfering sound. Each of these areas will be discussed. The level and spectrum of the speech correspond to source parameters in the Shannon and Weaver (1949, Ch. 1) model of communication.

Some of the earliest research dealing with the effect of signal level upon speech discimination was done by Fletcher and Steinberg (1929). These workers noted that intelligibility increased as signal level increased up to a point. When levels were increased beyond this point, intelligibility of speech began to decrease with increases in the level of speech presented. This same phenomenon has been reported by many authors (Beranek, 1947; French and Steinberg, 1947; Pickett, 1956) and has come to be called the rollover phenomenon. The level at which this transition occurs has been reported at various levels ranging from 80 dB Lp (Pickett, 1956) to 95 dB Lp (Beranek, 1947).

A possible explanation of the rollover phenomenon may be derived by analogy with an electronic amplifier. When an amplifier is operated such that it functions at levels as or near its maximum power output a type of distortion called peak clipping will occur (for a more complete discussion of this phenomenon see Villchur, 1962, Ch. 9). Bekesy (1960, Ch. 5 & 11) has shown that the structures of the human ear, like any transducer, operate within certain gain and maximum response limitations. Under high level acoustic stimulation, the stapes alters its mode of oscillation resulting in some peak clipping. This effect has been demonstrated by Bekesy to occur at levels around 90 dB Lp (within the range where the rollover phenomenon occurs). Zwislocki and Feldman (1970) have further noted that reflexes are triggered by acoustic signals in the 90 dB Lp range which limit the movement of all middle ear structures. This action also results in peak clipping. Peak clipping in the hearing mechanism is not limited to middle ear structures. Bekesy (1960, Ch. 11 & 12) demonstrated that the maximum amplitude of vibration of the basilar membrane is limited by physical constraints upon the limit to which it may be stretched. This leads to the conclusion that at some level the basilar membrane will introduce distortion by peak clipping.

The existence of peak clipping in middle and inner ear structures is consistent with the rollover phenomenon. As signal levels become great they are peak clipped at several levels of the auditory transducer mechanism. This peak clipping removes certain components of incoming signals (such as speech) thereby degrading them and reducing their intelligibility.

Intelligibility of speech signals is also diminished when the level of speech is very low. (Fletcher & Steinberg, 1929; French & Steinberg, 1947; Pickett, 1956). The ability to discriminate speech decreases with decreased signal level until a level is reached at which the signal can no longer be heard. This phenomenon appears to be the result of two interacting factors; masking effects and threshold effects. Masking effects are a complex topic and this special case of masking will be considered later with other masking phenomena. Threshold effects are more clearly understood by analogy with electrical systems. Any electrical system requires a certain level of input signal in order to respond. Wever (1949, Ch. 6) has demonstrated that the ear (which converts an acoustic stimulus into a neuroelectric signal) like any transducer requires a certain signal level in order to respond. Any components of the input signal which do not reach this threshold level are not transduced and, so, are not included in the neuroelectric signal to the brain. That this phenomenon (clipping of low amplitude components; like peak clipping in reverse) does take place in the ear was demonstrated by Wever and Bray (1930) in their investigations of cochlear microphonic potentials. The signal degradation resulting from this effect could explain, at least in part, the poor intelligibility of low level speech.

The second speech parameter which Webster (1969) mentions is spectrum. The importance of the spectrum to the perception of speech was first noted around the turn of the century. Lord Rayleigh (1908) attributed poor intelligibility when listening over the telephone to the limited frequency spectrum transmitted by the telephones of his day. This conclusion was supported and characterized in greater detail by several workers. Maximum intelligibility is obtained with a transmission band from 250 Hz to 7000 Hz (French & Steinberg, 1947). Signal components at the higher and lower ends of this spectrum, however, may be removed without seriously affecting intelligibility. It has been demonstrated that the three octave bands in the region from 500 Hz to 2000 Hz are the most crucial ones for retaining intelligibility and that, within this region, the most important frequencies for discriminating speech are those centered at 1500 Hz (French & Steinberg, 1947; Egan & Wiener, 1946; Fletcher, 1953). In this case, then, a limited transmission band results in a signal degradation reducing the proportion of acoustic cues present in a signal, thereby causing decreased signal intelligibility.

Before leaving the area of source parameters, it is necessary to consider the interactive effect of level and spectrum upon intelligibility. Egan and Wiener (1946) noted that, for near threshold levels, it is necessary to increase the transmission bandwidth as level decreases in order to maintain a given level of intelligibility. This finding may be explained by reference to the Wiener-Shannon information theory (Wiener, 1948; Shannon, 1948). It is beyond the scope of a brief review such as this to adequately discuss information theory and accordingly, only concepts relevant to the current topic will be mentioned. Fano (1950) states that information theory has two premises: 1) that a communicative process is made up of a series of indications of a choice selected from a usually finite number of possible choices and 2) that the concept of probability plays a fundamental role in any communicative process. Miller (1953) discusses the concept of units of information and clarifies the notion of the part played by probability in a communicative process. When a communicator is transmitting a message, the receiver is trying to understand the message. In order to achieve this understanding, he makes successive predictions as to message content. These predictions are made on the basis of the information which is being received. Each bit of information reduces the number of possible interpretations of the message. In effect, the probability that the receiver will correctly predict the message is increased. Note that this probability is directly related to the amount of information received. The greater the amount of information, the higher the probability of correctly interpreting the message. According to the Wiener-Shannon theory, each element of a message can transmit information. In order for the element to be effective in increasing the probability of correct interpretation, however, the information transmitted must add to the amount of information the receiver gets (Miller, 1953). When the information conveyed by a particular element is identical with the information which has been conveyed by previous parts of the message, the amount of information is not increased. A message element which conveys no new information is referred to as redundant. Note that redundant information neither increases nor decreases the probability of correctly interpreting the message. S. S. Stevens (1950) points out that information theory predicts that a highly redundant system will be greatly resistant to distortion effects, since removal of information transmitting elements (i.e. signal distortion) does not reduce the total amount of information transmitted, but only some of the redundant information. (For more detailed coverage of information theory and its relationship to speech perception see Fano, 1950; Miller, 1950 and 1953; Stevens, 1950; Aborn & Rubinstein, 1952; Osgood & Sebeok, 1955; Miller & Isard, 1963; Pollack, 1964). Given speech would then be directly proportional to the amount of non-redundant information units present in it. When a low level signal is presented to a listener, the threshold effect acts to eliminate the lowest level acoustic information units, thus reducing the intelligibility of the speech by reducing its information content. The only way to retain the original intelligibility score in such a situation would be to find some non-level means of increasing the amount of information units present. This is exactly what increasing the transmission bandwidth would accomplish. The trading relationship between level and bandwidth may, therefore, be characterized as a compensation phenomenon where lost units of information usually transmitted on the level channel are replaced by other information units transmitted on the frequency channel. The listener is then utilizing the same amount of information units (though not the same units) in order to achieve a given condition of speech intelligibility.

It should be noted that the concept of information units may be used to characterize the other effects of source parameters upon intelligibility. Clearly, limiting the transmission bandwidth of a signal or peak clipping of a signal would result in a loss of acoustic information units which could then explain the consequent decrease in signal intelligibility. A consideration of this concept of intelligibility leads to a further conclusion. The same bandwidth-level trading relationship noted by Egan and Wiener (1946) should occur for very high level signals where the rollover phenomenon (Fletcher and Steinberg, 1929) is seen. As the signal level is increased and more level information units are peak clipped from the signal, it should be possible to maintain a uniform level of intelligibility by increasing the signal's frequency bandwidth (i.e. adding information units transmitted on the frequency channel). To the author's knowledge, there is no existing research on this topic, however, such an investigation certainly seems warranted.

Although certain parameters of the speech signal have long been recognized as having potential bearing upon intelligibility, they have not been adequately studied. It is generally known that female voices differ from male voices in both spectrum and level (Ladefoged, 1962). The work of Dunn and White (1940) provides some indication that female voices are lower in level than male voices and that spectral differences between males and females occur primarily in the lower frequencies of the voice spectrum. It should be noted, however, that this comparison is based upon an analysis of only six men and five women. Clearly, then, further research is needed before the relationship between male and female voices may be characterized. It is apparent, however, that there are differences between male and female voices. Recalling the information units paradigm for characterizing intelligibility, it may be seen that these differences could affect intelligibility. The units of information in a given speech signal are transmitted on level and/or spectrum channels. If the frequency and level characteristics of speech are altered one would expect that the information units necessary for intelligibility must be transmitted on different channel components of the acoustic signal. While the level and spectrum encoding of the information is probably similar, the specific values of information carrying components must vary. This would, of course, result in differences in the specific points at which interactions between level and spectrum would occur. It could also result in variations in the threshold and rollover phenomena. Current research dealing with level and spectrum effects (as reviewed above) is limited almost exclusively to the intelligibility of speech produced by male talkers. In order to more realistically characterize the phenomenon of speech intelligibility, research is needed which depicts the parameters of speech intelligibility for the voices of female talkers.

Another parameter of speech which may affect intelligibility is the clarity of the talker's articulation. That clarity of articulation affects speech intelligibility has long been recognized in the field of speech pathology (Van Riper and Irwin, 1958). This fact is actually the major motivation for the development of the field. In terms of the information units concept, this phenomenon may be characterized as a distortion effect. When speech articulation is poor, certain acoustical components of the speech signal will be distorted or may be altogether absent. Consequently, the information units normally conveyed by those acoustical aspects of the signal will be lost or at least distorted. This situation may, therefore, be expected to reduce the available information units to the listener thereby reducing intelligibility. The speech samples used in all existing research on speech intelligibility were produced by professional speakers. The speech of these professionals is highly trained to achieve maximum clarity and intelligibility. The average speaker does not articulate as clearly as a professional and, therefore, it may be expected that average speech will be less intelligible than that produced by a professional. Research is consequently needed which would allow the characterization of the intelligibility of speech produced by the average, non-professional talker.

Closely allied to talker intelligibility is the topic of communications systems intelligibility. Whenever a communications system is introduced between the source and the receiver of a signal, its response characteristics will affect speech intelligibility. The communications system will introduce peak clipping at some upper input level. Distortion may also be introduced when the signal input to the system is low in level. Also, any communications system will have some limitations on the range of frequencies passed on to the receiver. Spectrum limitations will, as explained earlier, result in a loss of acoustic cues. Clearly, then, the introduction of a communications system (such as an active hearing protector or an emergency communications system) between the source and receiver of a speech signal will introduce some distortion resulting in a loss of acoustic cues. The extent to which the lost cues are redundant (i.e. reproduce information that is present in other acoustic cues that are not lost) determines the intelligibility of speech received through the system. Thus, the goal to strive after in the design of any communications system is to pass a significant amount of non-redundant acoustic cues while maintaining the required fiscal economy. Unfortunately, economical systems tend to show limited response characteristics. Methods of empirically testing the intelligibility of communications systems have been developed and will be reviewed in a later section of this discussion. The significant point at this juncture is that communications systems can introduce distortion that will degrade signal intelligibility and must, therefore, be considered whenever a communications situation is evaluated.

The acoustical environment in which speech occurs is integrally related to all of the speech intelligibility factors described up to this point (Moncur & Dirks, 1967; Nabelek & Pickett, 1973; Ross & Giolas, 1971; Crum & Tillman, 1973). Factors in the acoustical environment (the transmission channel component of the Shannon and Wiener 1949 communication model) which may affect intelligibility will now be considered.

The final parameter affecting speech intelligibility which Webster (1969) cites is masking noise effects. For intelligibility purposes, Webster defines noise as "unwanted sound . . . an unwanted disturbance within a useful frequency range, the range being the one that carries the intelligibility of speech." Webster also defines those parameters of the speech-in-noise situation which determine the intelligibility of the speech; a) noise level, b) noise spectrum and c) speech level and spectrum. The action of these parameters is highly interactive and it is not realistically possible to consider the effect of these factors in isolation. It should be noted at this point that a communications system may also introduce noise into a communications situation.

One conventional means of considering the interaction of signal level with noise level is the use of the signal-to-noise ratio. It has been demonstrated that, for the speech-in-noise situation the absolute levels of speech or of noise are not the factors determining intelligibility. Rather, the determining factor is the level of speech relative to the level of noise; the signal-to-noise ratio (Pickett, 1956; Webster, 1965). In general, the higher the speech level is relative to the noise level, then the higher is the intelligibility of the speech. This phenomenon is also compatible with the information theory characterization of speech intelligibility. When the sound reaching the ear is a mixture of both speech and noise, it is possible that some components of the speech signal will reach the ear at a level equal to or lower than the level of the noise. Bekesy (1960) has demonstrated that the movement of auditory structures is directly proportional to the amplitude of sound energy incident upon the ear. The speech signal (in a noise background) is, therefore, manifested as a displacement of already displaced auditory structures. Clearly, in order to be detected, components of speech signal must result in a detectable increment in the movement of auditory structures. Jerger (1955) has shown that normal hearing listeners can detect increments in an auditory signal of the order of .75 dB or higher. Presumably, increments smaller than this are insufficient to trigger increments in the neuroelectric response of the inner ear transducers. This would predict, therefore, that any acoustic information unit in the speech signal whose level was 10 dB or more below the noise level would not be detected. The presence of noise, then, could result in the loss of acoustic information units and, thereby, cause a decrease in intelligibility. As the overall speech level is decreased relative to the noise (lower signal-to-noise ratio) increasingly larger amounts of acoustic information units will be lost and intelligibility will decrease.

It was noted earlier in this discussion that the phenomenon of threshold or low level hearing response is, at least in part, a special case of masking. Whenever a person is listening to a speech (or other type) signal he is listening to that signal in the background of ambient noise present in the listening environment. Whenever the signal is very low in level some components of the signal may be similar in level to the background noise. In some instances signal components may be lower in level than the ambient noise. For such circumstances the background noise will mask these signal components and acoustic cues will be lost, thus reducing intelligibility. This phenomenon combines with the low amplitude response limitations of the hearing mechanism to give the empirical observation of reduced speech intelligibility whenever signals are produced at very low levels.

Up to this point, the discussion of masking effects has been limited to a consideration of signal-to-noise ratio. The effect of signal-to-noise ratio is, however, dependent upon the spectrum of the noise and its relationship to the speech spectrum. It should be apparent from the foregoing discussion that, in order to produce masking, the noise energy must be concentrated in the frequency bands where the speech signal occurs (Klumpp and Webster, 1963). The effect of noise spectrum upon intelligibility is further modified by the phenomenon called spread of masking. Miller (1947) reported that, as noise level increases, lower frequency noises have increasingly greater disruptive effects upon speech intelligibility. Work by Bekesy (1960, Ch. 11) has shown that the basilar membrane is critically damped for those portions of its length which lie apical to its point of maximum displacement for a given input signal. It should be mentioned at this point that signals enter the inner ear at its basal end. The basilar membrane responds (by maximum displacement) to the highest frequencies at the basal end, and to the lower frequencies at the apical end. It may be seen, therefore, that a low frequency sound displaces the entire membrane to some extent while a high frequency sound displaces only the basal end. In effect sounds enter the inner ear at the basal end and travel apically to the point of maximum displacement. Beyond this point the basilar membrane is critically damped. Whitfield (1967, Ch. X) notes that, as level increases, the region of maximum displacement of the basilar membrane is broadened. Due to the critical damping effect, this broadening extends the region of maximum displacement basalwards (towards the higher frequency response areas). As the level of a low frequency noise increases, therefore, the basilar membrane is displaced by it into higher and higher frequencies. Wherever this displacement due to low frequency noise occurs, the masking phenomenon will result in a loss of information units and reduced intelligibility. The phenomenon of spread of masking, therefore, may also be explained in terms of the information theory concept of intelligibility.

Other aspects of the acoustical environment which may affect intelligibility are speaker-to-listener distance and reverberation time. The distance from the talker to the listener may be expected to affect the intelligibility of a speech signal. Beranek (1971, Ch. 9) points out that increasing the distance from an acoustical source to a receiver will reduce the level of a signal at the reciever (direct field effect). The effect of reducing signal level has already been discussed. We may expect, that increasing speaker to listener distance will reduce intelligibility, through threshold and masking effects. When the speech signal being considered is produced in a room this phenomenon is limited. Beranek (1971), Ch. 9) notes that, depending upon the absorbtion characteristics of the room, a source to receiver distance will be reached where the direct field effect no longer controls the signal level at the receiver. The level at the receiver is then controlled by the reverberation characteristics of the room (reverberant field effect). This would predict that intelligibility would decrease as the speaker-to-listener distance is increased until some asymptotic distance is reached (where reverberant field effect controls signal level). For this and greater distance intelligibility would no longer be affected by distance. This prediction was supported by the findings of Crum and Tillman (1973).

Room reverberation has an effect upon intelligibility beyond the one just noted. When the reverberation time in a room increases the intelligibility of speech in that environment decreases (Knudsen and Harris, 1950, Ch. 9). When the reverberation time exceeds about one second this decrease in intelligibility becomes significant. This phenomenon may also be discussed in terms of the concept of information units. When the reverberation time is long (longer than one second) a speech signal will persist at relatively high levels for long periods of time. Once a given signal has reached a listener and been perceived, it no longer serves a purpose to him. It becomes unwanted sound or, as Webster (1969) has defined it, noise. When the signal becomes noise, it then, has the same masking properties, as any noise, and the discussion of masking presented above then applies. Speech makes a particularly effective masking stimulus since all of its energy is concentrated in the speech frequencies. The task of discrminiating speech in a highly reverberant room is therefore, a special case of the task of discriminating speech in noise. Listening in long reverberation times results in poor intelligibility due to a loss of information units by masking effects.

It has been demonstrated that room reverberation time can interact with signal-to-noise ratio to increase the effects upon speech intelligibility (Crum and Tillman, 1973). This phenomenon is a natural outgrowth of the foregoing discussion. If listening in long (over one second) reverberation times is analagous to listening in noise, then, the combination of noise and long reverberation times is essentially the same as listening in louder noise. The reverberating, thus unwanted, speech signals will add to other environmental noises like any noise and will reduce intelligibility to the extent that they produce masking effects in the frequencies needed for speech intelligibility.

The phenomenon of reverberation time effects upon speech may not entirely be explained by the acoustical effects described above.

Since the masker generated by the reverberation phenomenon is speech, it results in a unique "non-acoustic" masking effect which must be reconciled to the concepts presented elsewhere in this paper. The phenomenon of changes in a masking effect due to the content of a masker was first reported by Pollack and Pickett (1958). These authors found that, for a uniform level of a masker, listeners showed decreasing speech discrimination scores for increases in the number of talkers on a speech babble masking tape. Jerger and Jerger (1967) also observed that masking by continuous speech produced a greater masking effect than could be accounted for in terms of the acoustic energy present in the masker. These authors put forward the idea that the masking effect was a psychological one due to the semantic content of the masker. This phenomenon has been labelled perceptual masking.

Studies have investigated the notion that semantic content accounts for perceptual masking. These researches (Dirks & Bower, 1969, and Brandt & Stewart, 1970) made use of a technique put forward by Meyer-Eppler (1950) who showed that a tape recording played backwards retains all of the original acoustic parameters of the signal but destroys semantic content. In both the Dirks and Bower work and the Brandt and Stewart study the forward and the backward speech maskers resulted in the same intelligibility score for the speech signal. It was concluded, therefore, that a speech masking stimulus exerts a greater masking effect than can be accounted for merely by the acoustic energy present in the signal but that this effect was uniform even when the semantic content of the masker was destroyed.

A review of investigations by Broadbent and Gregory (1964) and Kimura (1964) may prove useful in the understanding of this phenomenon. These researchers found indications that the cortical processing of speech stimuli is handled by the dominant hemisphere while the non-dominant hemisphere processes non-speech stimuli. In this conceptualization a speech type masker is processed in the same part of the brain as the signal while a non-speech masker is processed in another part of the brain from the speech signal of interest. This arrival at the cortex of both the signal and the masker still unseparated may account for the greater amount of masking resulting from the speech type masker. In terms of the information units notion of speech intelligibility the speech masker may be concieved of as contributing acoustically to signal degradation at the end organ as described above. The difference arising with a speech type masker would be that since it travels to the same cortex as the signal it acts to degrade the signal a second time at the cortical level and more information units are lost here.

A second area of non-acoustic masking effects has been reported in the literature and deals with the phase relationships of the stimuli at the two ears. Hirsh (1950) first demonstrated that, in a speech-in-noise situation, the intelligibility of the speech signal is better when the speech source is spatially the same.

Spieth, Curtiss and Webster (1954) and Shubert (1956) indicated that the critical aspect was the time of arrival (i.e. phasing) of the stimuli at the two ears. Leavitt and Rabiner (1967) working under earphones showed that phasing allows the listener to separate the speech from the noise and when he can do this the noise produces less reduction in intelligibility. In all of the above researches the masking effect relative to phasing was independent of level.

Information presented by Whitfield (1967) may shed some light on the mechanics of this phenomenon. He points out that the neural response of the superior olive to impulses from the cochlear nuclei is directly dependent upon the phasing of the auditory stimuli at the two ears (Ch. XI). Whitfield agrues for the notion that the superior olive provides the primary analysis of the phase relationships of the signals at the two ears. He also points out that two primary centrifugal pathways, the function of one of which is inhibitory, arise in the superior olive (Ch. VIII). In terms of information units, when the signal and masker are not in phase, they may be separated by the action of the superior olive or the masker may even be inhibited by the action of the centrifugal pathway mentioned by Whitfield. In either case the two signals may be separated before they reach the cortex limiting the masking to a peripheral effect and precluding the possibility of a second reduction in the number of information units at the cortical level.

In both cases of non-acoustic masking effects, therefore, the information units concept of intelligibility may be used. The phenomena require that the masker be separated from the signal at a precortical level. If the separation does not take place the masker will interfere with the final, cortical processing of the signal and further degrade it resulting in greater reduction in the intelligibility of the signal.

A final goal to consider is the parameters of the listener which may have an effect upon speech intelligibility. It is a generally known fact that hearing impaired listeners have poorer intelligibility than normal listeners (Davis and Silverman, 1960, Ch. IV). This phenomenon is quite compatible with the theory of speech intelligibility described in this paper. The general effect of hearing loss is to eliminate and to distort some of the acoustical components of signals arriving at the ears (Davis & Silverman, 1960, Ch. IV). This would, of course, result in the loss and/or distortion of whatever information units were transmitted by those acoustical components and, consequently, in a decrease in intelligibility.

It has been suggested that everyday speech intelligibility is dependent upon the level and spectrum of the speech and upon the masking effects of any background noise as well as the hearing abilities of the listener. An information theory concept has been described whereby all of these parameters may be characterized as affecting intelligibility by means of altering the amount of information units available to the listener. Such a concept provides a unified means of studying the varying phenomena associated with speech intelligibility.

Intelligibility Testing: Recognizing that intelligibility may vary significantly with changing parameters of the source, transmission medium, and receiver, interest has long centered upon the development of tools for the empirical evaluation of speech intelligibility in a given situation. Lord Rayleigh (1908) first reported the phenomenon of reduced auditory discrimination in the presence of "some interference of the input signal." Campbell (1910) recognized the problem encountered and proposed a 20 monosyllable list for use in testing discrimination. The list had been arbitrarily selected and was not representative of any particular communication situation.

Hudgins et.al. (1947) presented a sentence type discrimination test in which the listener was required to answer a predetermined set of sentences. This replicated a typical conversational situation but did not give information on the subject's ability to understand speech in any situation.

Egan (1948) introduced the PB-50 wordlists which were designed to be phonetically balanced to match English conversational speech. This is to say that all the phonemes of English were represented in each of these 50 word lists in the appropriate positions within words and in the same percentages relative to each other as are found in everyday English. Egan desired many lists and actually produced 20 PB lists of 50 words each.

In generating this many lists, however, it became necessary to generate some words which were not commonly used and, as a result, the lists were quite difficult. It was felt by some researchers that the use of such uncommon words resulted in discrimination scores which suggested worse intelligibility than subjects were actually experiencing. Hirsh et. al. (1952) reduced the PB lists to six, 50-word lists which included only the more common words. These materials were less difficult than the wordlists developed by Egan.

Several other discrimination tests have been developed. The concept of the multiple-choice discrimination test was suggested by Black in 1957. Fairbanks (1958) developed a type of multiple-choice test called the Rhyme Test. In this procedure a stimulus is presented to the subject and he is required to select the stimulus word from a group of possible answers all of which differ from the stimulus word by their initial consonant sound only. House et. al. (1965) modified the Fairbanks test and suggested the use of this modification for testing the efficiency of certain communications systems. The reasoning for this is that in many communications systems a limited set of statements are used to convey messages. In the modified rhyme test the listener selects his response from a limited number of possible answers and, in this way, the procedure replicates the real life situation in which the communications system will be used. As Webster (1969) has pointed out. the measurement of communication efficiency in noise is essentially the same procedure as the measurement of communications system effectiveness.

A further test procedure for assessing listeners' abilities to understand speech in everyday contexts has been presented by Jerger, Speaks and Trammel (1968). In their synthetic sentence intelligibility test, these authors present the listener with the task of identifying a grammatical sentence using real English words but having no semantic content such as: "Small boat with a picture has become." The sentences are presented in a background of competing speech; the voice of a person reading a story. This provides sentence material rather than single words and it presents test items in a context of background noise, thereby, overcoming the two of the primary objections to the wordlist-type tests. This procedure also has disadvantages, however. First, there are only ten possible choices for test sentences and listeners can, therefore, learn the test sentences. This factor is confounded by the fact that in any given "run" each sentence is only presented once so that the subject has 10 alternatives only for the first sentence. As the test continues, the number of possible alternatives is reduced. A second problem is that content words are rarely repeated in the various sentences and, in most cases, the identification of one content word would allow the listener to identify the entire sentence. A further criticism is that, since there is only one talker reading the story that serves as a competing signal, there are gaps in the masker due to the natural pauses in conversational speech. Often a part of the test sentences occur in one of these silent gaps and listeners can identify the sentence by understanding the one or two words that occurred during the pause. Finally, the task of repeating nonsense sentences introduces perceptual and cognitive difficulties and thus, test scores for nonsense material may be reduced for reasons other than the basic intelligibility of the speech materials used.

Having reviewed existing material dealing with both speech intelligibility and intelligibility testing, the task remaining is the development of a test paradigm for evaluating communications systems that will be used in a coal mine environment. It is first necessary, therefore, to characterize the unique aspects of the coal mine listening situation in terms that may be related to known aspects of speech intelligibility. The communications systems of interest will be used in one of two situations. The active hearing protectors will be used during working situations typically when noisy machinery is being used. In this situation the essential information to be conveyed by the system is work related either in the form of warning signals or comments pertaining to the work at hand. In this situation the available vocabulary will be quite limited by the scope of the possible messages that may occur. In the second situation listening will probably be done in quiet. Again, the number of different messages to be conveyed will be limited and, consequently, the vocabulary is small. Thus, the coal mine listening situations of interest involve limited numbers of possible alternative messages. As Williams and Hecker (1967) point out, these parameters dictate that the Modified Rhyme Test

(House et. al., 1965) would provide the most realistic appraisal of the effectiveness of the various communications systems that may be used in the coal mine setting.

Other advantages of the modified rhyme task have been reported. Northern, Hattler & Nilges (1970) reported a smaller standard deviation for this procedure than for other discrimination tasks with the same subjects, indicating that results obtained with this technique are quite consistent. A study reported by Nelson & Chaiklin (1970) demonstrated that discrimination tests results obtained by having the subject write down his response were more reliable than procedures where the subject's response is verbal. Finally, the various lists of the modified rhyme test showed no statistically significant, interlist differences (Beyer, Webster & Dague, 1969).

The presentation of the Modified Rhyme Test in quiet through the various communications systems of interest will give information on the optimum performance that can be expected from a given system. For a more detailed look at system performance, however, testing in background noise is desirable. Clearly, since the active hearing protectors will usually be operating in the presence of noise, testing these devices in background noise is a reasonable way to evaluate their effectiveness. By contrast, it has already been noted that miners will typically listen to the emergency communications system in relative quiet so the function of testing in noise is not immediately apparent. The usefulness of an intelligibility test in noise is that it provides information useful in the comparative evaluation of different systems. When several systems are compared in quiet (optimum conditions) two or more may appear to perform equally well. It is, therefore, desirable to obtain further comparative information on the two systems in order to select the one best suited to coal mine use. The usual means of providing such information is to test the intelligibility of the system using a difficult to discriminate signal. A speech signal in background noise is such a signal. By presenting the signal in noise some of the signal's acoustic cues are masked by the noise and the signal is made more difficult to discriminate. This method is commonly used in the evaluation of hearing aids and it is often found that two systems may perform equally well in quiet but quite differently in the presence of noise. A further consideration in this area is that, by virtue of the fact that it is used in emergency situations, the emergency communications system may not always be used in optimum (quiet) conditions. It is, thus, possible that both system types may be used in noise and it is, certainly desireable that the systems be tested in noise.

To make tests more relevant to the coal mine setting, the noise used should have a spectrum that replicates noise spectra encountered in coal mines In considering presentation levels for speech and noise, one should keep in mind the fact reported earlier in this discussion that it is not the absolute level of noise or of speech that controls intelligibility but, rather, the signal-to-noise ratio. Thus, if multiple intelligibility tests are performed in noise it is necessary to vary the signal-to-noise ratio if non-redundant data is to be derived.

Literature on the levels of noise commonly encountered in coal mines has been reviewed in previous reports by this laboratory (Michael et.al., 1972 and 1973) and so the selection of a reasonable noise level for testing is not difficult.

Selection of an appropriate presentation level for the speech stimulus is a more complex problem. Webster & Klumpp (1962) report that when noise reaches 50 dB Preferred Speech Interference Level (PSIL) talkers begin to increase their vocal intensity at a rate of 5 dB for each 10 dB increase in the noise above 50 dB SIL. It would seem from this that a simple calculation could be made to determine the appropriate speech level in a given amount of noise. The data is complicated, however, by data reported by Coles (1969) which states that if the speaker wears ear protectors (which will be the case in one coal mine situation) he will not use sufficient vocal level to overcome background noise. The reason for this is that while the ear protector attenuates the level of the noise, it does not attenuate the person's own speech since this is heard by bone conduction. The level of the speaker's voice then would still be raised at a rate of 5 dB for each 10 dB increase in noise above 50 dB SIL, but the noise level must be calculated as the ambient noise level minus the attenuation provided by the ear protector. An additional complicating factor pointed out by Webster (1969) is that a speaker's vocal level is also determined by the nature of any feedback information on how effectively he is communicating. This material would indicate that when ear protectors are first put in place the speaker would use a less than optimum vocal level. If he has feedback on the efficiency of his communication, however, the speaker will eventually increase his vocal level until some more efficient signal-to-noise ratio is reached. One final complication that must be considered in the selection of test levels for speech is that the expected increase in intelligibility scores with speech level reverses at some critical level that has been approximated to be 95 dB SIL (Beranek, 1947 and Pickett, 1956).

Based upon the factors outlined in the above paragraph a 95 dB SIL speech level appears to be an appropriate upper limit for high level noise backgrounds when a person is not wearing ear protectors and individually selected lower speech levels whould be chosen for tests in lower background noise levels or when ear protectors are worn. The lower speech level should be based on the mine noise spectra and on the attenuation characteristics of ear protectors when worn. The method used here would be similar to that presented by Webster and Klumpp (1962).